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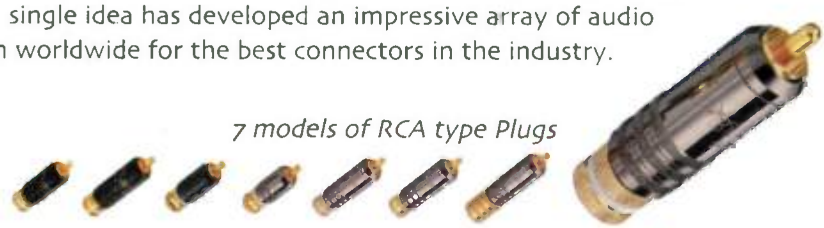

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Good News

ATLANTIC TECHNOLOGY'S SYSTEM 270 SPEAKERS AWARDED THX SELECT CERTIFICATION

Atlantic Technology® announces that its System 270-home theater speaker system has been awarded THX Select® certification. The six-piece system, now designated System 270 THX, has a redesigned center-channel speaker to comply with the home-theater performance standards set by LucasFilm's THX® Home Program. All other components remain the same. This added refinement brings THX Select performance to consumers at a more affordable price. Atlantic Technology, 343 Vanderbilt Ave., Norwood, MA 02062, (781) 762-6300, FAX (781) 762-6868, Website www.atlantictechnology.com.

Reader Service #141

NEW COLORFUL PYRAMID FOAM PANELS

NetWell Noise Control announces the release of brand new colors of their popular Pyramid Acoustic Foam Panels. These 2′ × 2′ hunter green and burgundy red panels are perfectly symmetrical, balanced and seamless in their installation. The NRC values on the 3″ thick Pyramids peek past 1.00, offering absorption rates of up to 80% of unwanted echo. Samples available upon request. NetWell Marketing, Inc., 6125 Blue Circle Drive, Minnetonka, MN 55343, (612) 939-9845, (800) 638-9355, FAX (612) 939-9836, Website www.asknetwell.com.

Reader Service #140

ZALYTRON CATALOG

Zalytron's new 1999 catalog contains over 100 pages of information on such product lines as Cabasse, Seas, Focal, Audax, and more. Zalytron Industries Corp., 469 Jericho Turnpike, Mineola, NY 11501, (516) 747-3515, FAX (516) 294-1943, E-mail zalytron@juno.com, Website www.zalytron.com.

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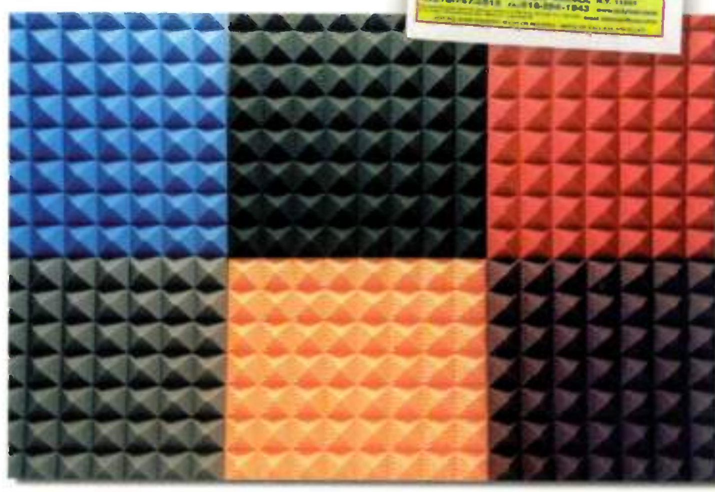

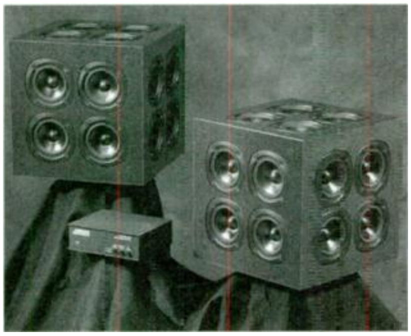

⊂ CUBE DESIGNS ⊃

Ambiance Acoustics has rolled out two new crossoverless loudspeakers—Super Cube and Hyper Cube. The new Super Cube adds five speaker drivers for a total of nine drivers. The new Hyper Cube takes the Super Cube's design approach several steps further, employing 16 drivers. Ambiance Acoustics, (858) 485-7514, E-mail rjsalvi@calcube.com, Website www.calcube.com.

Reader Service #137

■ LOUDSPEAKER & ROOM ACOUSTICS MEASUREMENT SOFTWARE

AcoustiSoft, Inc., has released version 5.0 of its popular ETF Audio & Acoustical Measurement Software. ETF 5 professional-quality Windows 95/98/2000-compatible program measures loudspeaker and room-acoustics parameters that affect the quality of critical listening environments, such as recording studios, editing suites, high-end audio listening rooms, commercial and home theaters, and more. The new program can be purchased directly from the program's web site for $199.95 (US). A demo program may also be downloaded from the web site. AcoustiSoft. Inc., Oshawa Center, PO Box 30621, Oshawa, Ontario L1J-8L8, (800) 301-1423, E-mail sales@etfacoustic.com, Website www.etfacoustic.com.

Reader Service #135

■ MULTIPURPOSE MINI-MONITOR

B&W Loudspeakers announces the B&W LM 1 Leisure Monitor, a compact two-way whose sonic refinement is reflected by its unusually contemporary, unique industrial design. The ultra-compact two-way, which stands less than a foot in height on a footprint just 5.5″ × 7.5″, is available in five high-tech color-schemes. The LM 1 employs a long-throw/high-output 5″ bass/midrange driver that exploits its computer-modeled, fourth-order-aligned vented enclosure to achieve bass extension to 65Hz. Despite its specified sensitivity of 91dB SPL at 1m/1W, it is rated for a full 100W of power handling, permitting it to fulfill the very wide dynamic demands of modern music and A–V sources. B&W Loudspeakers of America, 54 Concord St., North Reading, MA 01864-2699, (978) 664-2870, (800) 370-3740, FAX (978) 664-4109.

Reader Service #138

■ STEREO SOUND FOR SMALL SPACES

At CEDIA Expo '99, Niles Audio Corporation introduced an innovative new ceiling-mounted speaker. Designed to solve one of the hardest audio installation challenges custom installers confront, the CM650SI Stereo Input Speaker delivers stereo sound in areas where there is only enough space to install a single speaker. The CM650SI is built around a 6½″ dual voice-coil woofer and two ¾″ ultra-wide dispersion tweeters housed in a faceplate that delivers both left and right channels, making it possible to provide "stereo sound" through a single speaker. Niles Audio Corporation, PO Box 160818, Miami, FL 33116, (305) 238-4373, FAX (305) 238-0185, Website www.milesaucio.com.

Reader Service #139

About This Issue

Bill Fitzmaurice has generated much reader interest with his Snail series of folded-horn designs. His latest effort is called the Doppelganger Snail (p. 10), which features a pair of identical 10″ drivers in the same cabinet. Following the Snail tradition, this model delivers big sound from modestly sized drivers and cabinet.

Design two horns that are basically identical, with the only difference being the throat opening size and the design frequency. What would you expect the results to be? For the details, read **Louis C. McClure**'s experiment starting on p. 24 ("An Exponential Midrange Horn").

Chances are, you probably already have most of the equipment and parts you need to put together an impedometer. **Dick Pierce** explains this low-cost method for accurately measuring loudspeaker impedance ("The Impedometer," p. 28).

Which cabinet designs are most effective in converting electrical energy to sound energy in our speaker systems? To answer this question, **Mark Wheeler** enlisted the aid of a group of audiophiles to participate in a series of listening tests. Their findings are detailed in "Navigating Speaker Design: Listening to Walls" (p. 32).

Sometimes, designing a simple passive crossover can be anything but. What started out as a search for a simple solution to a particular problem turned into a study on the bass directionality problem. You'll be entertained and enlightened by this exchange between two audio veterans–**Charles Hansen** and **G.R. Koonce** ("Mating Subs and Satellites via Passive Crossovers," p. 38).

Also in this issue, speaker expert **Vance Dickason** tests a 5″ woofer from Peerless of Denmark ("Driver Report," p. 58), and **John Badalamenti** displays an attractive-looking center-channel loudspeaker for his surround-sound system ("Showcase," p. 61).

Speaker Builder

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JOHN STUART MILL

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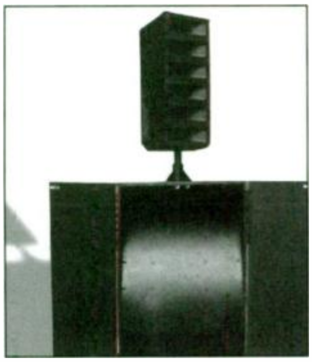
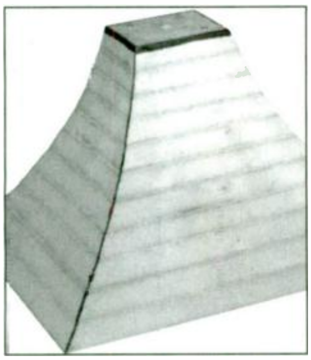

page 10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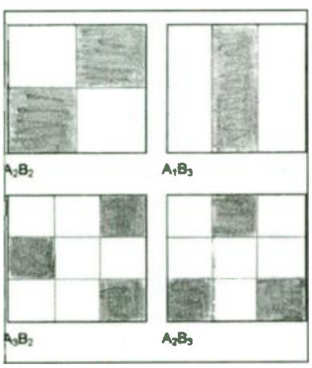

page 32


page 38

FEATURES

DEPARTMENTS

IN EVERY ISSUE

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Editorial

AMATEURING: A FRESH VISION

By Edward T. Dell, Jr.

For the Love of It, Amateuring and Its Rivals, by Wayne Booth, 1999, University of Chicago Press, 237 pp., $22 (Available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 603-924-9464, Fax 603-924-9467, E-mail custserv@audioXpress.com)

An elderly retired professor of English from Chicago's distinguished university has written a book that has redefined for me the meaning of the word amateur. Of course, he *is* an English professor after all, and words are his business. But his main theme is music, music made by amateurs.

After nearly 30 years of using and promoting the word and practice of amateurs, I find it both startling and delightful to have a new dimension added to a word I use a great deal.

Wayne Booth believes that amateur should also be a verb, and that any activity done for the passionate, unquestioning, irrational love of it should be called amateuring. (Microsoft Word has just redlined that neologism, telling me that MS's minions consider it an error.)

Dr. Booth is a fiercely devoted music lover, especially chamber music, and preferably music made with strings. At 31 he took up the cello. By any rational standard this was a foolish thing to do. Unless one signs on to play a stringed instrument before the age of six, one is doomed to amateurism eternally. No career in music will be possible, no stardom as a performer, no concert tours, and no fat fees for entertaining concert audiences around the globe.

Booth has been playing with chamber groups of all sizes for over 50 years. Appropriately, his book is titled *For the Love of It*. His subtitle is far more telling: "Amateuring and Its Rivals." He begins with a chapter of definings.

His proposed dictionary entry reads: "***Am-a-teur*** (am'a-choor, -toor, am'a-tur')[F.fr.L, *amator* lover, fr. *Amatus*, pp. Of *amare* to love]. *n.* 1. One who practices an art or science or sport for his own pleasure, rather than as a profession. 2. One who does something without professional skill or ease."

Booth's major theme is the angst at not playing well, but being unable to stop playing. His descriptions of the joys of the experience of playing with others are exquisite. Anyone who reads this magazine has had such moments just listening to music, live or reproduced. Words never do such exalted moments justice.

His book is far more than praises for making music, even though made badly. His thesis is that giving up doing some activity which you genuinely love to do is a defeat for our humanity. In short, Booth believes that anything humans do for the love of it is worth doing badly.

His book is a severe indictment of a society that worships competitiveness and winning above all else, a society that rewards the star and despises the "loser." Our landscape is littered with disasters of what happens to the bright and famous, of the corruption of steroids and money in sports, of bribed Olympic officials and injured ice skaters.

The book is not all diatribe. It is also a stunning exploration of the joys of music performed. He describes amateur groups always searching for the essence of what Beethoven or Brahms really had in mind in a masterpiece. He believes that those original visions of the composers may never have been fully realized—but that all the versions performed are a dizzying variety of approximations. Yet, he sees all of these, done for love, as valid music making.

The author's first two chapters are autobiographical. In Chapter 3 Booth looks at human motives in choosing what we do. It is a thoughtful discourse that can help any reader begin to understand why he or she has chosen an activity. He graphically examines the all too often choices following disappointment over our lack of skill or the perfection we dreamed of, of mindless gaming, watching television or the endless spectator activities offered all of us. The performance of the star, the skill of the winner, leave us too often feeling diminished.

Chapter 3 is an insightful examination of the relationship of loving an activity and doing it for other reasons, and an especially acute survey of what money often does to amateurs. Chapter 4 examines the balance between the power of failure and the strength of a loved activity. Chapter 5 takes up the question of teaching versus loving. How teachers evaluate the passion a student may have against the teacher's ideas of success. For those of us who are older, Booth's thoughts on how aging affects an amateur's relation to a passionate activity are reassuringly familiar and enlightening.

Booth takes time out to reflect wryly on his severe difficulties in writing about his amateuring, over against his prolific output as a professional writer about his chosen field, literature. The remainder of Booth's book is a sustained series of chapters on the intense and surprising pleasures—I could say ecstacies—of music itself. Especially illuminating is his wonderful meditation on how playing transforms listening.

For the Love of It is one of those rare books that has the power to permanently change the way you look at life. Although its main theme is music making, it is also a perceptive examination of personal values. It will certainly enhance your ideas about the emotional intricacies and motives of music making. I find myself searching through catalogs for chamber music. I hear the cello, especially, with deepened respect for a Casals or a Rostropovich. It is also a book that should enlighten and reinforce the love most of the readers of this periodical have for pursuing audio excellence. It certainly captures precisely the motives which have sustained us as a staff since the periodical came into being.

This author's latest project derives its name from its two identical horns occupying one cabinet. The design is no illusion; it delivers genuine performance for sound-reinforcement applications.

The Doppelganger* Snail

By Bill Fitzmaurice

Before I describe the next in the Snail series of folded horns, I would like to thank those of you who have written positive comments on my work with horn cabinets. I regret that I have been unable to respond to all your letters, but I would like to answer here two questions recently posed.

QUESTIONS ANSWERED

First, one reader, while appreciative of my work, questioned how I could apply for a patent on a horn cabinet, likening it to a tuba in design. Actually, perhaps credit for the horn style of sound-producing apparatus should be awarded to Joshua, circa 2,500 BC. The then residents of Jericho could have attested to the efficacy of horns where dB levels are concerned. But I was not trying to patent the concept of horn-loading.

My patent application referred to the vented driver chamber, which directs the vent output as well as the driver front wave into the horn throat. I researched patents going back 20 years, but was unable to find that anyone else had done that, or at least had patented it. As it turns out, I didn't search back far enough. One James F. Novak applied for and was granted a patent in September 1955. His concepts were virtually identical to those I independently developed 40 years later. For reasons unknown, his designs apparently never made it into mass production. If anyone has any information about James Novak and his cabinetry, please forward same to me c/o *SB*.

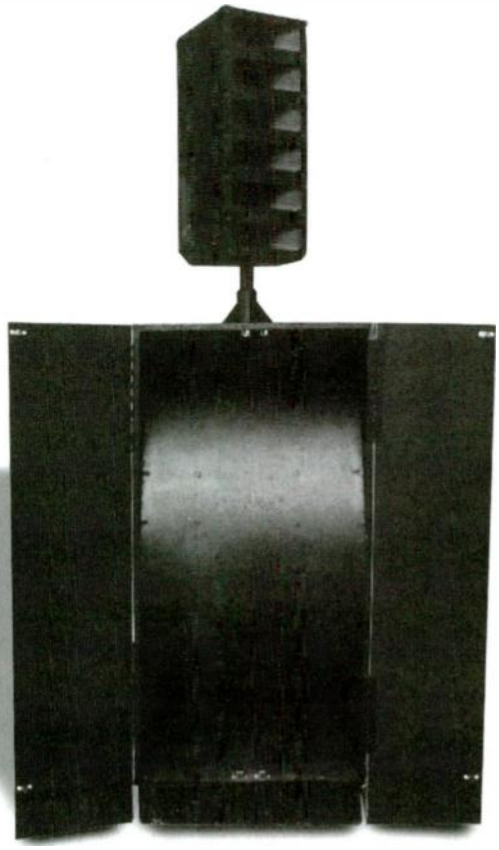

PHOTO 1: Doppelganger with pole-mounted high-frequency section.

Another writer wished to know if I'd design a Snail for an 18″ driver for subwoofer usage. The answer is, probably not. All my designs are for speakers that I myself have used on stage, and I have no need for cabinets with capabilities beyond that of Snail II (*SB* 3/98), which, as shown in that article, is quite capable of doing sub duty. Snails don't need big drivers for big sound (see the MiniSnail, *SB* 8/98). My intent with Snails is to get more power from less size, in both driver and cabinet, which brings me to the story of the Doppelganger Snail (*Photo 1*).

SHRINKING THE EQUIPMENT

Only four years ago my larger PA system had a sub, a woofer, and a mid/tweeter horn on each side of the stage, driven by three amps totaling 750W/8Ω, plus another 200W for my monitors. The ten-space rack required to carry all those amps and their associated paraphernalia weighed in at over 100 lb. Once I brought my Snail IIs on line, I was able to go two-way, allowing removal of one amp (and 25 lb) from the rack.

Then I came up with the MiniSnails, which have proven so versatile that I found myself using the big Snail II horn system only two or three times a year for the largest gigs. Since the MiniSnail gives me all the power I need from my 175W/channel Carver PM-175, I decided to find a way to replace the Snail II horn-biamped system with cabinets efficient enough so the Carver alone could drive them and still make them as loud as my biamp rig.

This would enable me not only to dispense with another amp and a crossover, but also to reduce my rack to six spaces, thus taking up less cargo space and lowering the rack weight to only 40 lb. At my age, losing weight is a priority; since I can't seem to accomplish that goal with my body, at least I can do it with my equipment.

You can gain more efficiency in two ways, one of which is simply to make the cabinet larger, allowing for a longer horn with a greater mouth area. Since my Snail II setup already strained my band's cargo capacity, not to mention my back, larger was not better. In fact, I wished to make it smaller and lighter. The other way to increase SPL/watt is

* *Doppelganger: German for "Double Walker." A ghostly counterpart of a living person. – Ed.*

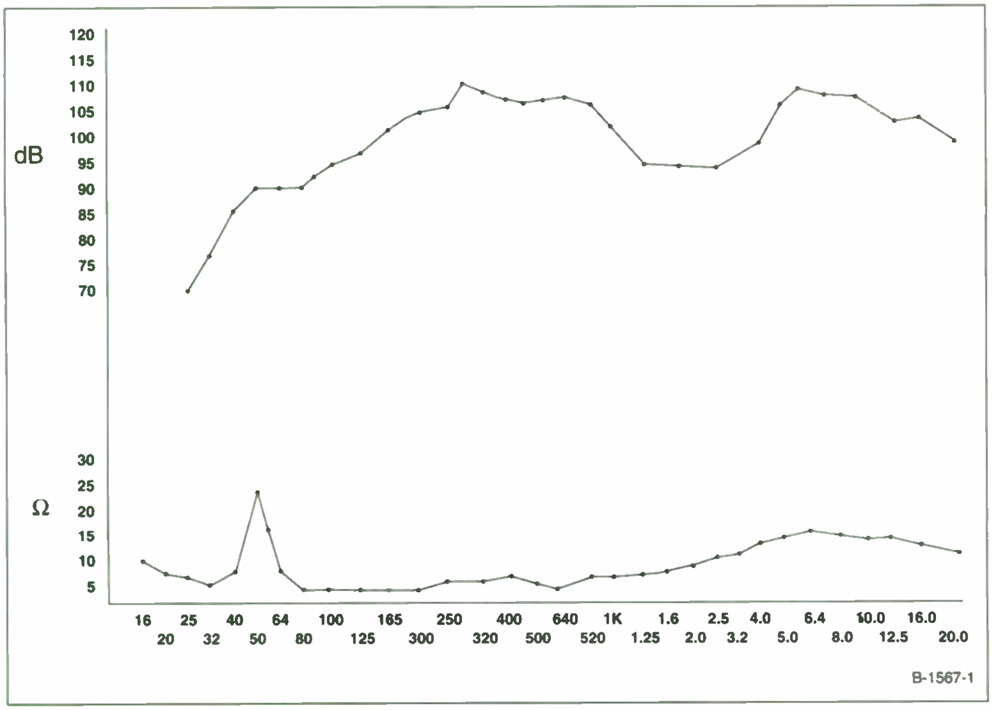

FIGURE 1: The Doppelganger's response graph (dB SPL/1m/2.83V input, f_s = 30Hz/3.8Ω, F_r [box resonance] = 48Hz/24Ω).

TABLE 1
VALUES FOR FIG. 1

FREQUENCY	DB	OHMS
16	NA	10
20	70	6
25	78	5
32	87	4
40	94	15
50	94	7
80	96	4
100	98	4
125	100	4
160	107	4
200	108	4
250	110	5
320	113	5
400	110	6
500	109	5
640	110	4
800	108	6
1k	100	6
1.25k	100	6
1.6k	98	6
2.0k	103	8
2.5k	108	8
3.2k	115	10
4.0k	118	12
5.0k	118	13
6.4k	116	15
8.0k	115	20
10.0k	110	19
12.5k	112	18
16.0k	112	17
20.0k	105	15

f_B—30Hz/3.8Ω
Fr—(Box resonance) 48Hz/24Ω

to reduce impedance. A 4Ω system would instantly give me another 3dB of sensitivity (more about this later). The amp manufacturers are wise to the impedance trick to gain more power; most of today's professional amps, including my Carver, will run happily into 4Ω, and many are rated stable with two-ohm loads.

Since 4Ω professional drivers aren't common, I decided to use two 8Ω drivers in parallel, and the desire to reduce cabinet size and weight led me to decide on 10″ drivers. I knew a smaller cabinet using 10s instead of 15s would likely lead to some loss of response below 100Hz as compared to the Snail II, but I could accept that since at larger gigs I use my Siamese Snail (*SB* 2/99) for the bass, and it is so powerful that reinforcing requirements in the PA are minimal, if required at all.

SIMILAR RELATIVES

The resulting Doppelganger is similar to the Siamese Snail, with two identical horns residing in one cabinet. Each driver's front wave feeds its own rapid-flair throat for maximum efficiency in the midrange. The drivers' rear waves share a common rear chamber that is vented into two slow-flare throats for maximum bass sensitivity. While each front wave exits its own mouth, whose small size accentuates the midrange by minimizing phase-cancellation effects, both drivers' rear waves combine to exit both mouths, doubling the mouth area available to the bass-rich vent outputs.

Notice that the rear-chamber vents are not ducted as in previous Snails. Minimal (approximately 10in^2) throat cross-sections lower the f_B of the chamber so effectively—from 75Hz without the horn to 30Hz with it—that ducts are unnecessary.

Another departure from former Snails is that the horn taper is not quite hyperbolic. When plotting out the horn, I noticed that the resulting shape was very close to circular. With only a slight alteration to the taper, I was able to make the flare of the horn plates a perfect semicircle. This might cause a slight loss of response, but two benefits outweigh any possible negatives. First, laying out circular parts is vastly simpler than laying out a hyperbolic shape. Even more important, since a circular shape resists acoustic pressures with a minimum amount of material, it is possible to use ¼″ plywood for the horn plate.

As in the Mid-Ranger, this cabinet utilizes café-style doors for protection from the elements while traveling and as waveguides when "open for business." They give about a 2dB boost in SPL from 150–300Hz, although from 1kHz–1.5kHz, they cause diffraction losses, also about 2dB. With relative power and excursion requirements being as they are, I'll gladly trade off 2dB at 1kHz for an additional 2dB at 150Hz.

DRIVER CHOICES

Choosing drivers for this project came down to two economical possibilities. First was the Carvin PS-10 (an OEM from Eminence), with specs of f_S = 57Hz, Q_{TS} = 0.5, V_{AS} = 2.5ft^3, an SPL of 95dB, a respectable 200W rating, and an attractive price of $49.95. The other nominee was the Eminence 10″ cast (available from MCM, Cat. #55-965), with f_S = 55Hz, Q_{TS} = .25, V_{AS} = 1.85ft^3, an SPL of 98dB, and a 150W rating. At about $70, it was still reasonably priced.

The higher Q_{TS} and V_{AS} indicated that the Carvin would have better bass, but the Eminence cast had the 3dB higher SPL in its favor. Also, everything I'd read about horns said that a low-Q_{TS} driver would work better than one with higher Q_{TS}. Unable to decide on one model of driver, I finally did the only logical thing by ordering both.

As in every Snail before it, the Doppelganger gave me the opportunity to dispel another myth about horn cabinetry. Upon testing the prototype with pairs of each model of driver installed, I found that there was no advantage to using the lower-Q_{TS} drivers. The response curves of the different drivers were virtually identical, with the only significant difference being a 2dB advantage across the bandwidth to the Eminence cast. So in the final version, my pair of Doppelgangers contain one of each driver.

The response graph (*Fig. 1*) reflects the mixed-driver version; when using a pair of PS-10s, you can expect a 1dB decrease in sensitivity, while a pair of Eminence-cast 10s should give about 1dB more, with minimal variation in the response curve.

As long as I was changing my bins, I figured I should try something new for my high-frequency section as well. While my JBL 2426s work reasonably well, with 110dB/W SPL, they're power hungry (their 16Ω impedance requiring 4V to obtain 1W), and they roll off above 12kHz. I'd had good results with Motorola piezos, so I decided to try an array of KSN1176 drivers. To achieve my desired SPL level of at least 112dB/2.83V, I went with six drivers per side, along with a seventh mounted at a 90° angle to the array for side projection.

I measured *Fig. 1* outdoors on a deck with a 2.83V input, to give a direct comparison with other cabinets with regard to voltage rather than wattage input. The graph is plotted using ⅓-octave points (for which I thank Louis McClure for his 4/99 *SB* article, "Plotting ⅓-Octave Frequency-Response Curves"). If there is an inherent defect in using very small horns to gain high SPLs, it is in the roughness of response, which makes an equalizer, preferably ⅓-octave, an absolute necessity. In using a ⅓-octave response-graph system, the setup is greatly simplified, setting the equalization (EQ) sliders inversely to the chart, in this case, using 106dB as the zero dB baseline.

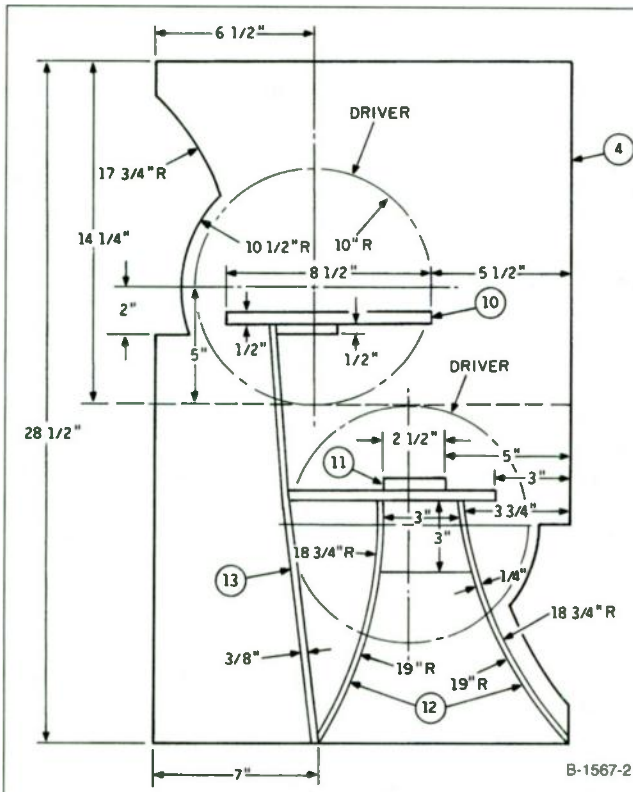

FIGURE 2: Baffle view; for clarity, not all parts or details are shown.

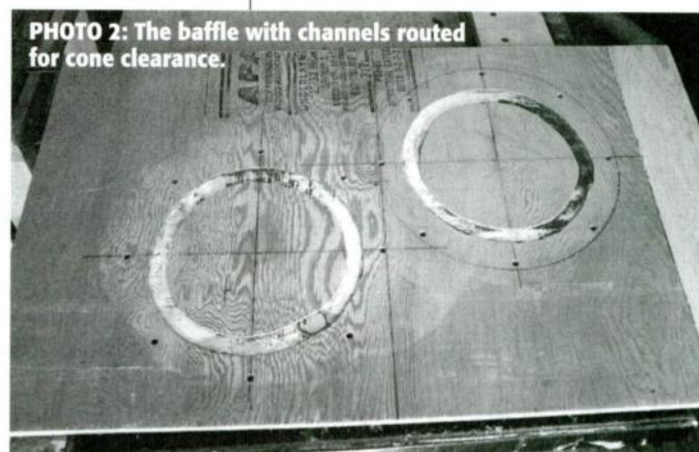

PHOTO 2: The baffle with channels routed for cone clearance.

Since 160Hz lies at 106dB, its slider is set to flat. 100Hz lies at 98dB, so its slider is boosted by 8dB, while the 4kHz slider is set at −12dB, and so on. This gives a neutral-response starting point for setting the EQ, with additional tweaking as required for room-acoustics correction and feedback control.

If you analyze the chart, 94dB sensitivity from 40–60Hz is quite respectable, especially from a pair of drivers with these specs. It results from the box frequency of 30Hz and the loading of the vent outputs by the slow-tapering throats. At 100Hz, horn loading of the front waves begins, getting the SPL up to over 100dB; from 250–800Hz, you see a greater than 110dB output, with woofer rolloff starting at 1kHz.

TABLE 2

Part descriptions Note: Numbers for the items refer to circled numbers on *Figs. 2–4*. Not all parts are shown in figures; see text and photos for full parts descriptions. Except as noted in the text, the recommended material for all parts is ½" plywood, 5-ply or more.

1. Sides
2. Top, bottom
3. Back
4. Baffle
5. Large horn plate
6. Large horn plate brace
7. Access door flange
8. Access door brace
9. Horn reflectors
10. Small horn reflector plates
11. Port horn plate brace
12. Small horn plate
13. Port horn plate
14. Small horn plate brace

ENDING THE ARGUMENT

On the continuing controversy over impedance load/number of drivers versus SPL, I have read countless explanations as to why using two drivers gives a 6dB boost in SPL for constant-voltage input. In theory, these permutations make some sense, but my gut always told me they didn't ring true. To put an end to the controversy, at least in my mind, I first measured this cabinet with only one speaker wired in, and then with two. With the same 2.83V input, the rise in SPL with the 4Ω load versus 8Ω, two drivers versus one, was 3dB.

PIEZO DRIVERS

Prior to designing the high frequency section, I tested the piezo drivers for polar response. The KSN1176 has the widest horizontal dispersion when it is mounted horizontally. I also wired six drivers in parallel and arranged them in every conceivable positional array to determine the best mounting configuration for horizontal dispersion. In every case

Swans M1 kit

High sonic resolution
Loudspeaker
with room friendly performance

"...explicit, easy to listen to, effortless, seamless and stunning."
Ernie Fisher
Swans M1 Speaker Systems Review
INNER EAR REPORT
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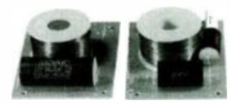
SWANS
SPEAKER SYSTEMS
The step beyond the limits


RT1C Tweeter


F5 Bass-midrange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Great news from Swans!

New beautifully cabinets for Swans M1 kits are available in three finishes: piano black, solid walnut and rosewood veneer.
Totally irresistible!

The Swans M1 speaker system is a two-way bass-reflex design. The front baffle is very narrow with rounded edges to reduce cabinet diffraction for better clarity and imaging. The internal panel and corner reinforcement substantially reduce unwanted cabinet vibrations. A flared port is mounted on the rear baffle for smooth transition from the port to cabinet boundaries. This provides linear bass performance and absence of port noise. The heavy-duty gold plated binding posts are mounted directly on the rear panel to enable easy cable connection.

The 5-inch paper/Kevlar cone woofer has a rubber surround, cast aluminum frame and a magnetically shielded motor system. This driver utilizes a central phase plug to avoid air compression, improving frequency response and dispersion. These key features greatly contribute to the M1's clear transparent sound and effortless dynamic performance.

The tweeter is a high-tech planar isodynamic design that employs Neodymium magnets and extremely light Kapton®film, with flat aluminum conductors.

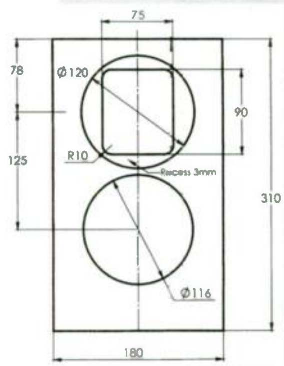
This unit provides an immediate and precise response to any transients in original signal, and gives the M1 an exceptional ability to reveal the true dynamics of instruments with a complex high frequency spectrum.

The crossover is a second order Linkwitz-Riley type resulting in an in-phase connection of the drive units. The crossover frequency between the two drivers is 3.3 kHz and only high quality polypropylene capacitors are used. Each filter has it's own dedicated board mounted on a special rubber interface to reduce vibrations and microphonic phenomenon. The filter boards are spaced inside the loudspeaker with the inductors positioned at right angles to minimize the interaction.

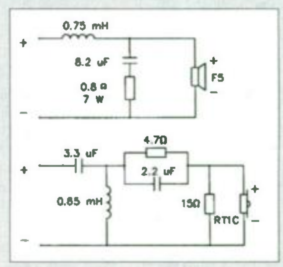
Swans M1 kit includes:
- 2x F5 paper/Kevlar bass-midrange drivers,
- 2x RT1C isodynamic tweeters with sealing gaskets,
- 2x dedicated tweeter crossovers,
- 2x dedicated bass-midrange crossovers,
- two flared ports,
- two pairs of heavy duty gold plated terminals.

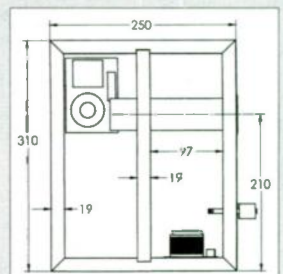
The drawings of the cabinet shown here represent general dimensions required for optimum bass performance. Rounded corners are advisable as they improve imaging and clarity. Actual finish and appearance is a matter of personal taste.

Price: $410 delivered without cabinets.
$660 delivered with cabinets.
Warranty 3 years. We accept VISA.

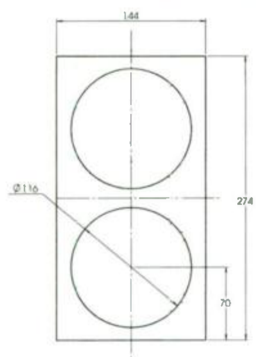


Front view

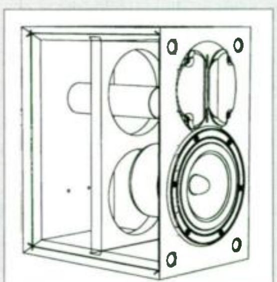




Right view of the cabinet with accessories(right side panel removed)



Internal panel



SPECIFICATIONS

Frequency response	60Hz-35kHz,±2dB
(1m,half space)	55Hz-40kHz,-3dB
Sensitivity,1W/1m	86 dB
(100Hz-8kHz averaged)	
Nominal impedance	8 ohms
(7.2 ohms minimum at 250 Hz)	
Power handling	50W nominal, 90W music
Dimensions,HxWxD	310x180x250 mm

Amplifier requirements:
30W recommended minimum.

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where the drivers were mounted side by side, there were huge peaks and valleys in the polar response as phase aberrations either boosted or cancelled output at various frequencies.

The final vertically arranged array of horizontally mounted drivers gave the best polar response and widest horizontal dispersion (120° average) of any of the combinations I tried. I used a seventh driver mounted at 90° to the array to cover the "dead spot" close to the side of the PA. This driver is mounted vertically to narrow its dispersion so it will not conflict with the main array.

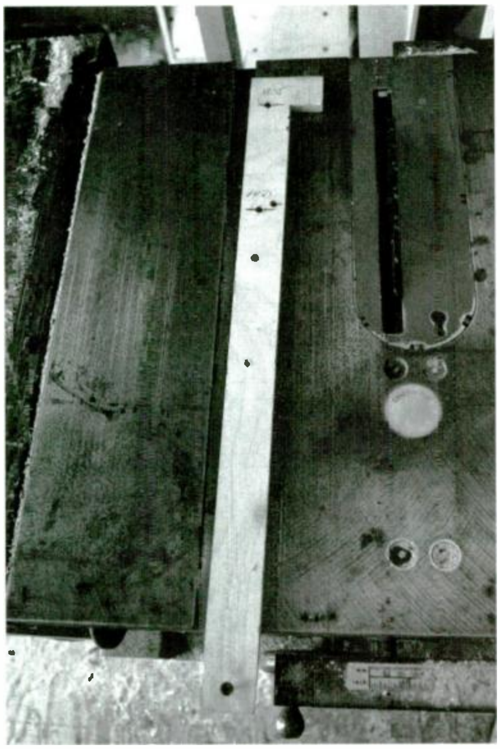


PHOTO 3: The "high-tech" compass.

The KSN1176 is advertised as having response down to 1.8kHz, but as you can see, it's more like 2.5kHz before these piezos start to rock. They really can't match a good dynamic driver like the JBL2426 in the midrange. But from 3.5kHz to over 16kHz, this array screams with a minimum SPL of 110dB, and peaks of 118dB.

Now consider that a JBL 2426 and its horn lens cost me about $325 eight years ago; seven KSN 1176s will set you back only about fifty bucks. Also consider that with this kind of efficiency, this horn array needs practically no power to drive it, making unnecessary a second amp for biamping, and an electronic crossover as well.

As to possible impedance problems with so many drivers, check out the impedance portion of *Fig. 1*. Truly high-impedance devices, even seven piezos in parallel, never go less than 10Ω within the audible bandwidth. There is still the risk of their going to a capacitive load at very high frequencies, so to preclude any problem, I series-wired them through a 2Ω resistor.

CONSTRUCTION

I built the prototype mainly from ½″ plywood (the exceptions are noted in the text and parts list). As always, you may opt for heavier materials to ease joinery, although the Snail's self-bracing design makes it unnecessary as far as strength goes. All dimensions are approximate, since actual material thicknesses are seldom if ever as specified. You may alter dimensions as required to accommodate your materials, making sure to measure the actual sizes required once you've drawn the part locations on the sides and baffle.

All joints are butted, fastened with drywall screws and construction adhesive. Use plenty of adhesive, since the joints must be airtight. I used screws of 1″ to 1⅝″—longer ones where necessary for holding power, shorter ones where required to prevent penetrating the panels. The nominal size for most joints is 1¼″. I drilled and countersank all screw holes.

Start by cutting out the baffle (*Fig. 2*), marking the driver locations on its rear. Drill holes for the driver-mounting T-nuts, and rout a ⅛″-deep channel to prevent the cone from slapping against the baffle during long excursions (*Photo 2*); alternatively, you can use a ⅛″ or ¼″ spacer.

The next step is to make a compass (*Photo 3*). This is simply a piece of wood about 20″ long and 2″ wide. Insert a drywall screw close to one end. Then drill ¼″ holes through the board at distances equal to the radii of the arcs you will draw with the drywall screw as a center. To use the compass, press the screw point hard into the center for the arc you wish to draw, and use a pencil point inserted through the appropriate hole to draw the arc (*Photo 4*).

On the baffle front, draw the locations of all the straight parts first; the duct-throat plates have such a shallow arc that you may approximate their positions with straight lines. Use the compass to draw the 19″-radius arc of the horn plates. To do this, clamp the baffle to another piece of plywood that will serve as a drawing extension (*Photo 4*). Draw an arc from the junction of the small horn plate and the horn reflector plate (12 and 10, respectively, in *Fig. 2*); draw another from the point where the horn plate ends at the baffle edge. Now place the compass point where these two arcs intersect and draw an arc between the small horn reflector plates (parts #10) and the baffle edge. This is the small horn plate.

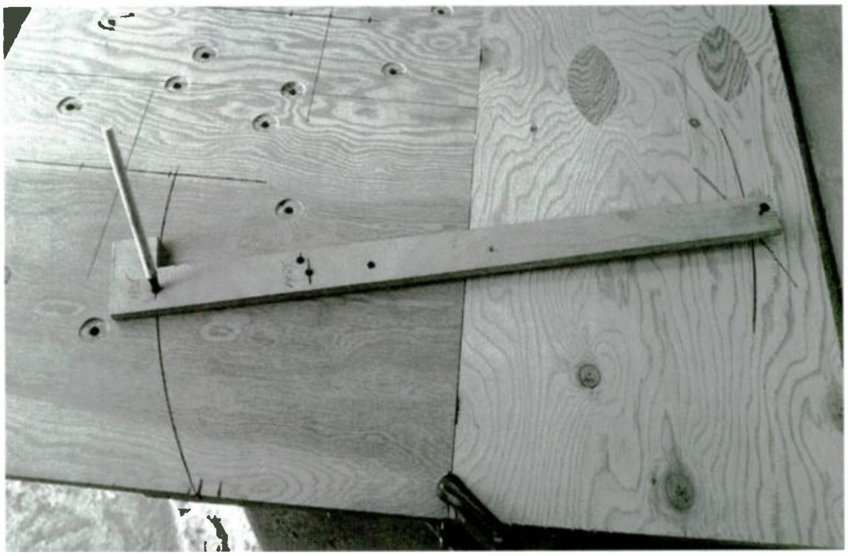


PHOTO 4: Tracing arcs on the baffle with the compass, using an extension board to locate the center.

Repeat the process for the 18¾″ small horn braces, and repeat again all steps for the second horn. Note that the small horns are located off-center; they are not symmetrically placed on the baffle. Also, to minimize phase cancellations in the throat, the throat hole in the baffle is not centered over the driver.

BAFFLE HOLES

Cut holes in the baffle for the driver

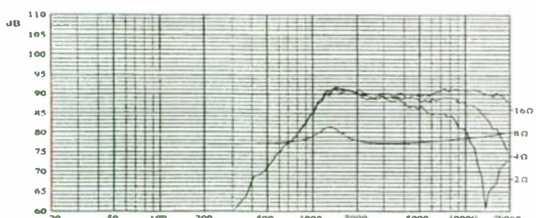
NEW Peerless Shielded Speakers at Introductory Prices

810435 1" Dome Tweeter

The 810435 is a 1" textile dome tweeter with a **neodymium** magnet. The tweeter is 2-1/16" square and 9/16" deep. The voice coil is ferrofluid cooled. The small size lends itself to use in small A/V systems, MTM's and speaker arrays. We recommend a minimum crossover frequency of 3000Hz.

Price Each ~~$18.70~~ $13.00

Znom	8.0	ohm	Cms	0.05	mm/N
Zmin	5.9Ω@3.1KHz		Sd	6.2	cm^2
Re	5.5	ohm	BL	2.0	N/A
Le	0.1	mH	Vas	-	ltrs
fs (free air)	-	Hz	Xmax	-	mm
fs (baffled)	1334.0	Hz	Sensitivity		
Qms (baffled)	2.70		2.83V / 1m	90.1	dB
Qes (baffled)	3.66		Longterm Max		
Qts (baffled)	1.55		System Power	80	W
Mms (baffled)	0.30	g	Magnet weight	0.01	kg

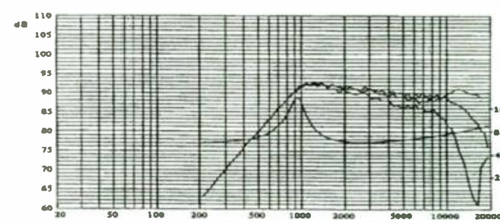


812978 1" Dome Tweeter

The 812978 is a textile dome tweeter with the **"Wide Angle"** face plate and shielded magnet. The dished out face plate controls directivity through out the frequency range of the tweeter and not just the high end frequencies. This economical high quality tweeter is a good choice for any A/V application.

Price Each ~~$24.45~~ $17.00

Znom	8.0	ohm	Cms	0.08	mm/N
Zmin	5.8Ω@259Hz		Sd	6.2	cm^2
Re	5.4	ohm	BL	2.8	N/A
Le	0.1	mH	Vas	-	ltrs
fs (free air)	-	Hz	Xmax	-	mm
fs (baffled)	940	Hz	Sensitivity		
Qms (baffled)	5.50		2.83V / 1m	88.6	dB
Qes (baffled)	1.36		Longterm Max		
Qts (baffled)	1.09		System Power	100	W
Mms (baffled)	0.34	g	Magnet weight	0.1+.07 kg	

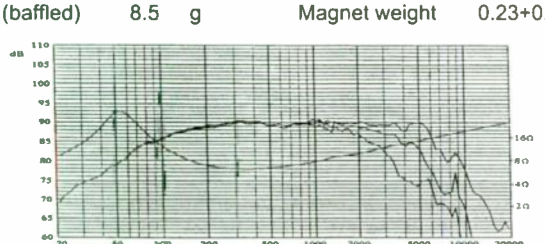


830377 5" Shielded Woofer

The 830377 features the unique **Nomex** fiber cone. The cones are made by a non-pressed air dried method, retaining the original structure of the fibers, resulting in a stiff non-resonant cone. The die cast basket, **Nomex** cone and shielded magnet system culminates in speaker with exceptional clarity and detail for any A/V or home speaker system.

Price Each ~~$40.45~~ $28.00

Znom	8.0	ohm	Cms	1.01	mm/N
Zmin	6.3Ω@345Hz		Sd	91	cm^2
Re	5.7	ohm	BL	6.4	N/A
Le	1.1	mH	Vas	11.6	ltrs
fs (free air)	55.8	Hz	Xmax	2	mm
fs (baffled)	54.4	Hz	Sensitivity		
Qms (baffled)	2.29		2.83V / 1m	89.3	dB
Qes (baffled)	0.40		Longterm Max		
Qts (baffled)	0.34		System Power	100	W
Mms (baffled)	8.5	g	Magnet weight	0.23+0.1 kg	



830378 6.5" Shielded WOOFER

The 830378 features the unique **Nomex** fiber cone. The cones are made by a non-pressed air dried method, retaining the original structure of the fibers, resulting in a stiff non-resonant cone. The die cast basket, **Nomex** cone and shielded magnet system culminates in speaker with exceptional clarity and detail for any A/V or home speaker system.

Price Each ~~$41.80~~ $29.00

Znom	8.0	ohm	Cms	0.54	mm/N
Zmin	6.1Ω@259Hz		Sd	143	cm^2
Re	5.7	ohm	BL	6.2	N/A
Le	1.1	mH	Vas	15.1	ltrs
fs (free air)	59.9	Hz	Xmax	4	mm
fs (baffled)	57.4	Hz	Sensitivity		
Qms (baffled)	3.44		2.83V / 1m	88.6	dB
Qes (baffled)	0.76		Longterm Max		
Qts (baffled)	0.62		System Power	100	W
Mms (baffled)	14.4	g	Magnet weight	0..23+0.16 kg	

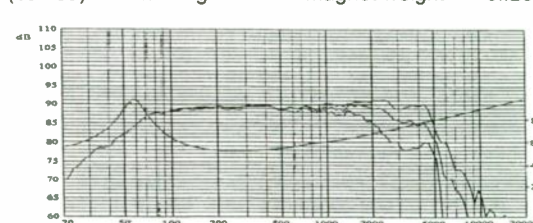


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PHOTO 5: The baffle, showing the throat and port holes.

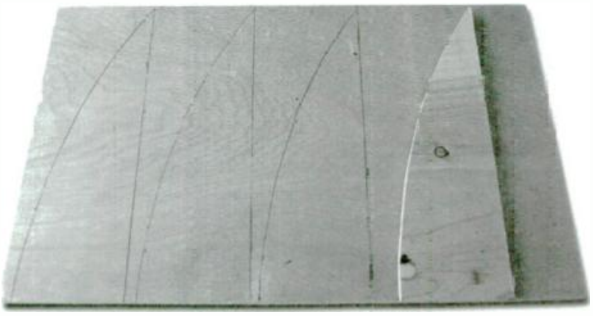


PHOTO 6: Tracing the small horn braces.

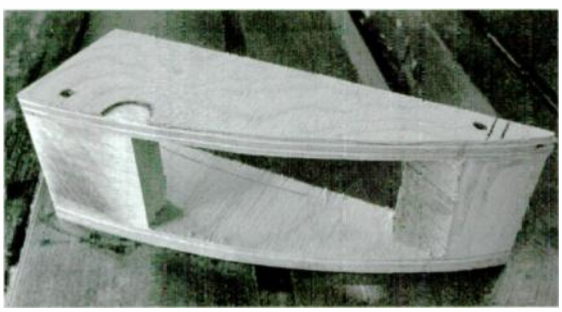


PHOTO 7: Small horn braces with bracing blocks attached and sanded to contour.

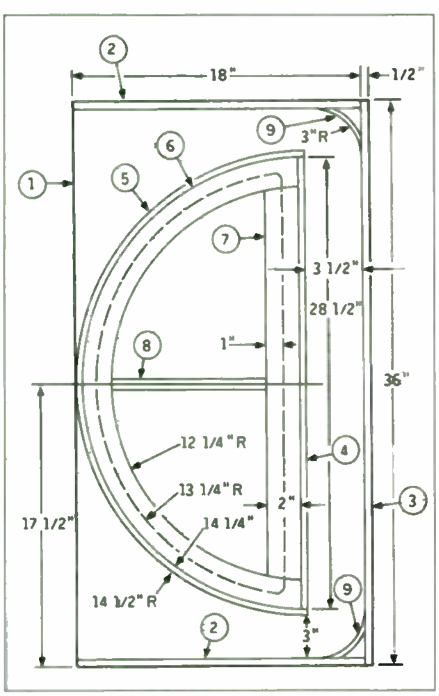


FIGURE 3: Cabinet side view; for clarity, baffle attachments are not shown.

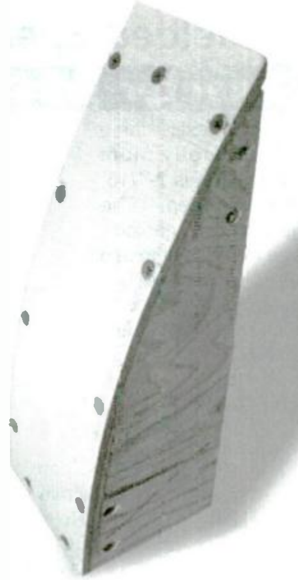


PHOTO 8: Completed small-horn assembly.

throats and ports as shown on the baffle diagram. Chamfer the throat-hole edges. Drill recesses in the baffle for the T-nut heads so that they will not protrude above the baffle surface; install the T-nuts. Also cut the ports, which occupy the area between the drivers, the cabinet side, and the small horns (*Photo 5*).

In a fashion similar to laying out the baffle, fashion the small horn-brace pattern (*Fig. 4*), cutting it out, preferably on a bandsaw, and then using it to trace three more braces—seven for two cabinets (*Photo 6*).

The parts that mount on edge to the baffle are cut to a consistent 3½″ width; for the sake of precision, it's a good idea to rip enough stock to make all these parts at once, without changing the rip-fence position on your table saw. From either ¾″ stock or plywood, cut out the small horn-brace blocks, measuring 2½″ square, which are first installed and then sanded so they taper to match the braces (*Photo 7*).

To these assemblies, attach the small horn plates. These are made from ⅛″ plywood; flex the sheet prior to cutting to determine the easier flexing axis. Cut the plates a good inch longer than the measured distance required, overhanging the excess at both ends. After the adhesive has set, saw the assemblies and sand them to match the outlines drawn on the

PHOTO 9: Reflector plates attached to the baffle; these may be made in two pieces as on the prototype, or in one piece as shown on the baffle diagram (see text).

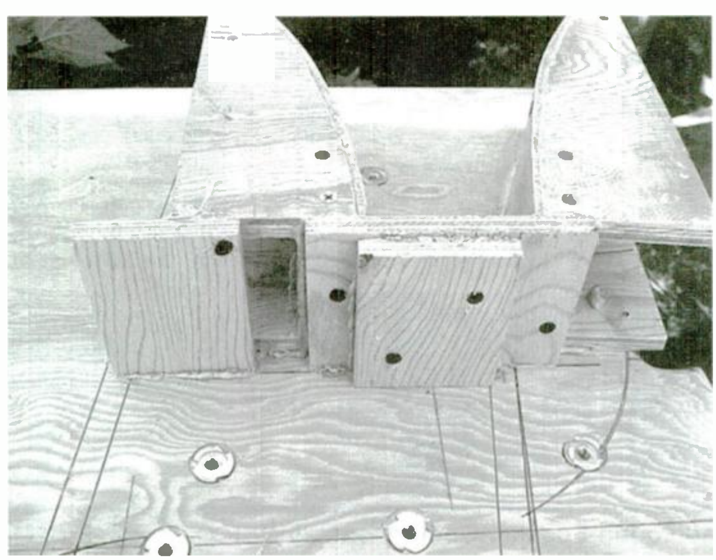
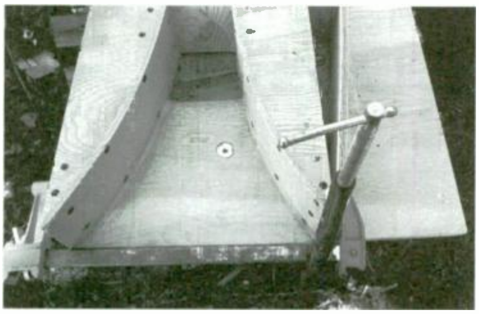


PHOTO 10: The port horn brace, attached to the rear of the reflector brace.



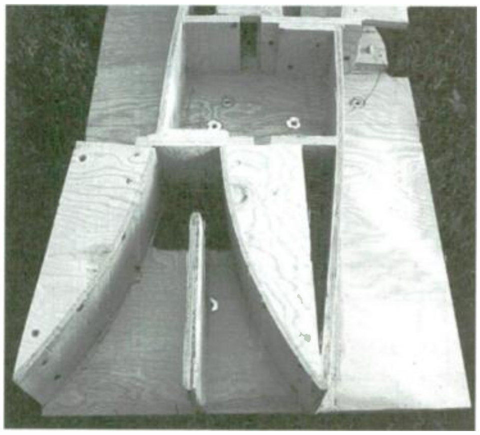


PHOTO 12: The completed baffle assembly.

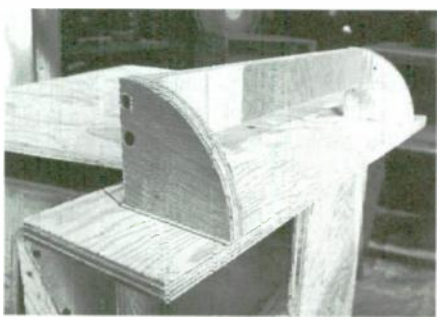


PHOTO 13: PVC cutting jig.

PHOTO 14: PVC clamped to the cutting jig.

PHOTO 11: Using two clamps to bend a port horn plate into place while it is screwed to the small-horn assembly.

baffle (*Photo 8*), remembering that the asymmetric design of the horn will result in two differently shaped assemblies.

Attach the throat reflector plates to the baffle (*Photo 9*), followed by the braces for the port horn plates and the small horn assemblies (*Photo 10*). (In the photos, the reflector plates are in two pieces because I was using up some small scraps. You may instead make the reflector plates of one larger piece of wood, as shown in *Fig. 2*.) Once in place, the small horn braces that overhang the ports can be trimmed away with a saber saw.

You should also cut the port horn plates, made from ⅜″ plywood, along the easier-bending axis. You bend them as you screw and glue them to the baffle. First attach them to the small horn assemblies (*Photo 11*), and then to their remaining braces, using plenty of adhesive on the joint with the baffle to ensure an airtight seal; they will easily bend to the proper shape as they are screwed in place.

Finally, attach the small horn dividers to the baffle. Make these just long enough to extend from the mouth hole to the edge of the baffle, with both ends chamfered. They are also asymmetrically located, placed about ½″ off center in the throat, so that phase cancellations caused by reflections within the throat will occur at different frequencies in either half of the throat, minimizing the cancellation effects (*Photo 12*).

THROAT REFLECTORS

The curved throat reflectors are made of 6″ PVC that you quarter lengthwise and then cut at right angles. PVC is perfectly safe to work with if you follow commonsense precautions. First, always screw or clamp it to a board or jig when cutting it

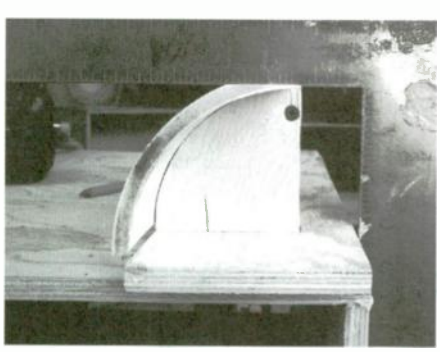


PHOTO 15: Squaring the PVC edge prior to making the second cut.

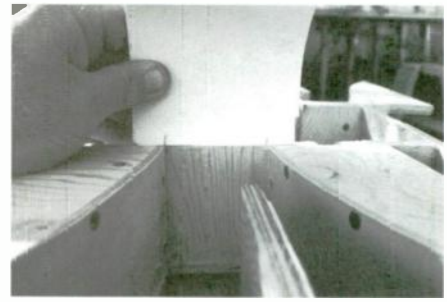


PHOTO 16: Marking quartered PVC at one end.

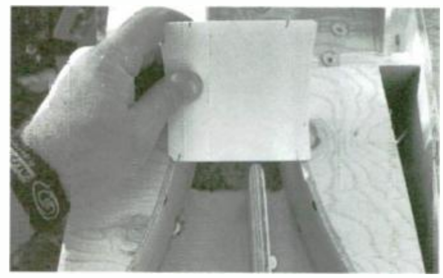


PHOTO 17: Marking the other end of the throat reflector.

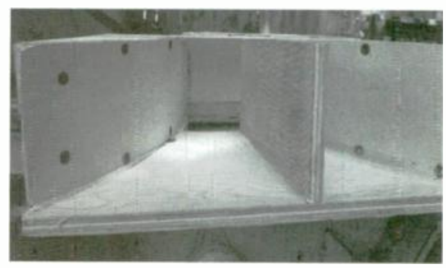


PHOTO 18: The completed throat reflector in place.

with a table saw. PVC will close tight on a blade if you attempt to cut it lengthwise in one pass, so set the blade height so it does not quite cut all the way through, and complete the job with a utility knife.

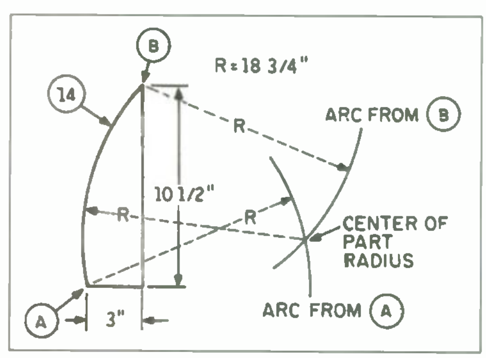


FIGURE 4: Small horn brace (part 14). Note how an 18¾″-radius arc is drawn from points A and B; the intersection of these arcs determines the point from which the arc of the brace is drawn. The same technique is used to lay out the parts on the baffle.

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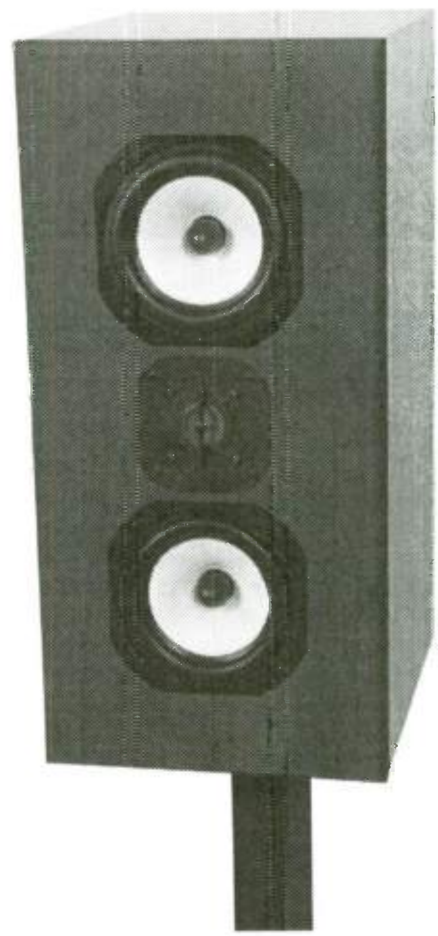

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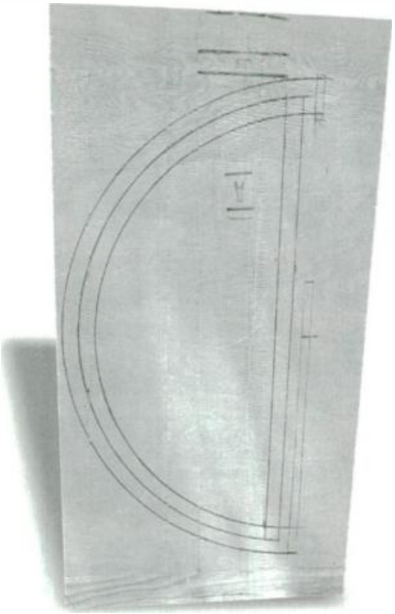

PHOTO 19: Three "half-moons" drawn on a cabinet side.

To get a good 90° edge on a quartered piece of PVC, use a jig such as pictured in *Photo 13*. The curvature of the rounded parts matches the inside curve of the PVC, which is either clamped (*Photo 14*) or screwed to the jig. When you feed the PVC through the saw, always handle the jig, never the PVC, to keep your hands away from the blade. After you've cut

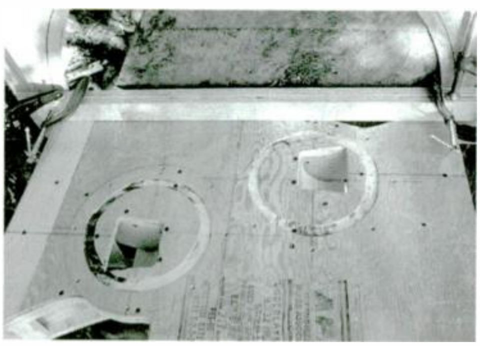

PHOTO 20: Using a guideboard and clamps to hold the baffle and side together while being screwed and glued.

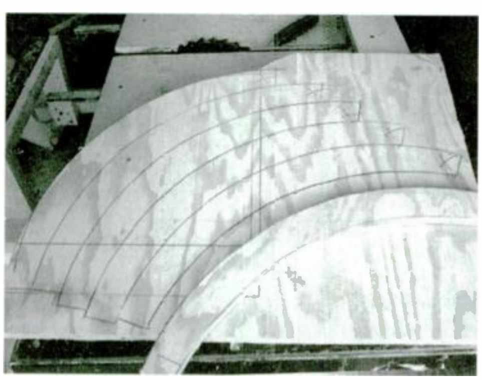

PHOTO 21: Tracing the large horn braces.

one edge, use a square to ensure an accurate 90° cut on the second edge (*Photo 15*). Finally, you achieve a further measure of safety, as well as smoother cuts, if you use an abrasive blade instead of a toothed blade when you are cutting the PVC.

With pieces of quartered PVC about 5″ long, trial-fit the reflectors and trim them to size on a bandsaw, marking on each piece the required width at either end (*Photos 16* and *17*). A precise fit is not required; when the cut piece is within about a ⅛″ tolerance, glue it in place with hot-melt glue, using the glue to fill the voids between the piece, the brace, and the horn (*Photo 18*).

Once you've completed the baffle, move on to the sides (*Fig. 3*), drawing the parts locations and using the compass again to draw the 14½″-radius arc for the large horn plate and the 14¼″ arc for the brace. On one side, also draw two more arcs to the inside of the horn, one with a radius of 13¼″, and the other with a radius of 12¼″, as well as two straight lines parallel to the baffle, one of them one inch and the other two inches inside the baffle (*Photo 19*). The middle line of the three "half-moons" (the dotted line in *Fig. 3*) is the cut-line for the driver-access door; cut it out (from one side only!) with a jig-saw, starting with a plunge cut.

Attach the baffle to the cabinet sides, ensuring an accurate joint by first clamping a straight guide board to the joint line and clamping the baffle to it while you drive in the screws (*Photo 20*).

LARGE BRACES

Next form the pattern for the large horn braces, with radii of 12¼″ and 14¼″. These are best made from ⅝″ or ¾″ plywood for ease of driving screws into them edgewise, although you may use ½″ if you're careful. Cut the pattern about 2″ longer than a perfect quarter circle, so that when two braces are screwed together to form a semicircle, you can overlap the ends.

When tracing the pattern to obtain a total of eight braces per cabinet, the outer edge of one brace can form the inner edge of the next to simplify cutting (*Photo 21*). Trim four braces lengthwise to fit in place on the cabinet sides, and install them. Overlap and joint the other two pairs, placing them over the braces already in the cabinet to make sure they are the correct overall length, and attach them to the baffle evenly spaced in the cabinet.

A simple brace of ¾″ plywood or

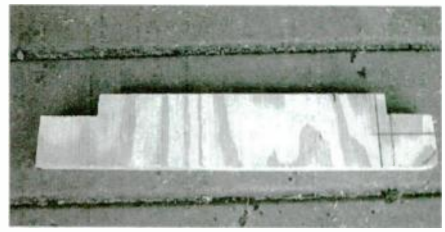

PHOTO 22: The fitted brace that spans the access door opening.

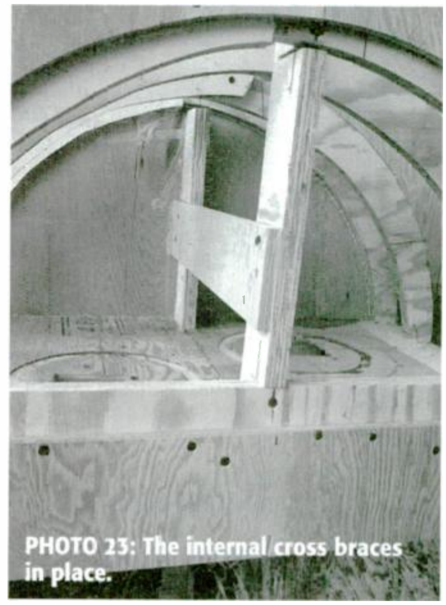

PHOTO 23: The internal cross braces in place.

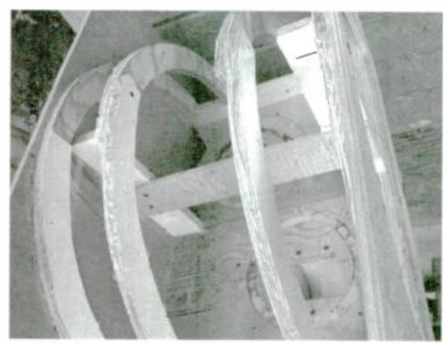

PHOTO 24: The completed large horn-brace assembly; note how the interior braces are lapped.

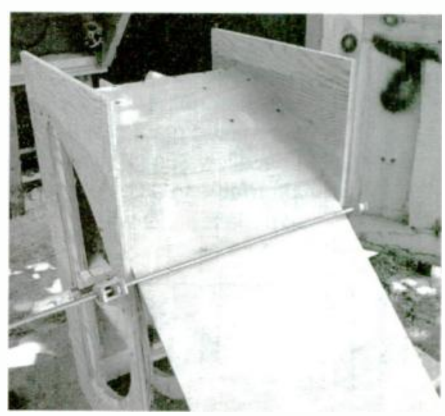

PHOTO 25: Holding the large horn panel in place during assembly.

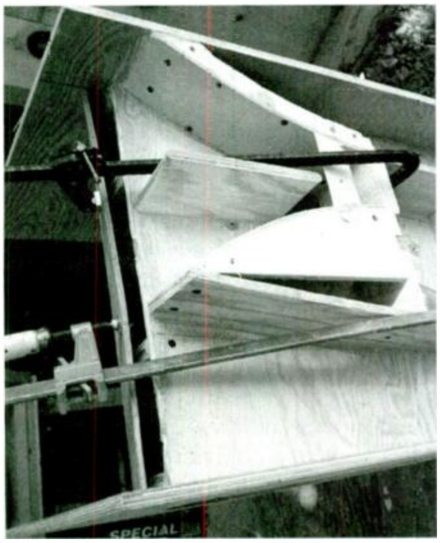

PHOTO 26: Pulling the large horn panel in place with clamps.

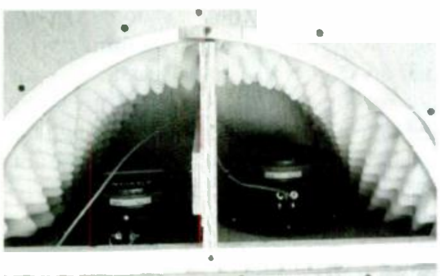

PHOTO 27: The driver chamber.

stock about 2″ wide runs from the baffle to the horn brace on the solid side, while a shaped brace (*Photo 22*) connects the baffle and access-door flanges on the other. Another piece of ½″ plywood also about 2″ wide connects these two braces (*Photos 23* and *24*), both to stiffen the cabinet and to hold the sides apart at the correct distance. Make sure that the brace at the door is flush with the flanges so that the door will lie flat against it when in place.

The large horn plate is next, ¼″ plywood cut 45⅜″ long and slightly narrower than the cabinet interior width for ease of assembly (or you can use luan underlayment, which is often referred to as ¼″, but may be as thin as 5.5mm). Flex it before cutting to determine the better axis on which to cut. Glue and screw it in place with screws about every 4″. It attaches first to one end of the baffle, is held down by a long clamp as the screws are driven (*Photo 25*), and is finally attached to the other end of the baffle, again using long clamps to pull it into place (*Photo 26*). Sand the joints between the horn plate and the baffle so they are flush, with the edges slightly rounded.

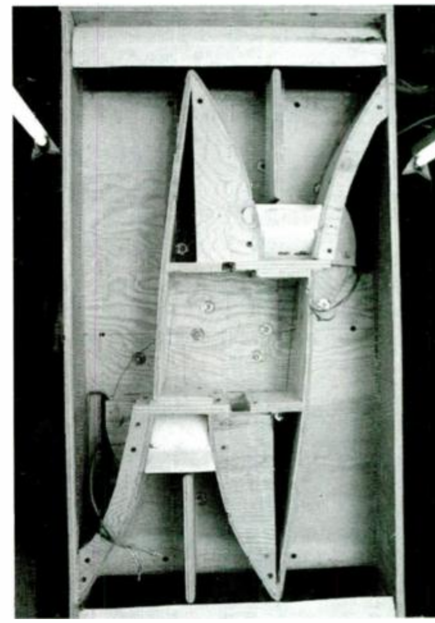

PHOTO 28: View of the baffle just prior to attaching the cabinet back.

DAMPING AND WIRING

Attach the top and bottom and the cor-

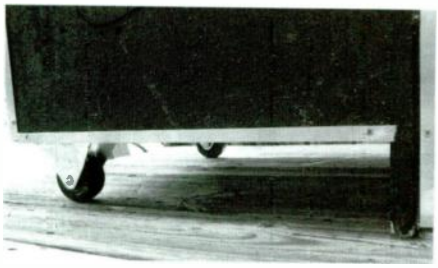

PHOTO 29: Optional casters on the cabinet rear, and flange on the cabinet front, with aluminum corner edging all around.

PHOTO 30: Cafe doors closed for transport.

PHOTO 31: Kitchen-cabinet latches hold the cafe doors closed.

PHOTO 32: The high-frequency box.

PHOTO 33: Holes in the high-frequency box for side-fire tweeter, pole-mounting socket, and handle.

ner reflectors—again quartered and chamfered 6″ PVC glued with hot-melt. Line the rear chamber behind the horn plate with acoustic foam, and install the drivers (*Photo 27*). Attach about 18″ of 16-gauge speaker wire to each driver, leading the free end of each through the ports (*Photo 28*). Use another piece of wire to connect the two wires to each other, making sure to maintain polarities.

When you install your choice of jacks on the back (when to do this depends on how you plan to finish the cabinet and the attachment method of the jacks), align them just over the port openings; in use, the lower jack is for the signal input, while the upper one serves as a tap for the signal to the high-frequency section.

If you intend to finish the cabinet with either carpet or laminate, apply the finish to the driver-access cover and back prior to assembly. For a painted finish, assemble the parts first and then paint. If you attach handles using nuts and bolts, they also should go on before you attach the access cover. Rim the access flange with neoprene weatherstripping and screw the cover in place with screws no more than 4″ apart for an airtight seal. Make sure you also screw the cover to the brace.

A cabinet this large really needs casters, and you can get away with buying only two per box by putting a plywood extension on the cabinet's lower front (*Photo 29*). Since it is cut to the same height as that of the casters, this extension flange keeps the cabinet stable when in use, and also serves to acoustically couple the cabinet to the floor. Tipping the cabinet back allows you to roll it easily; a handle at the top rear of the box helps with this.

If you opt for the cafe doors (*Photo 30*), attach them to the cabinet with piano-style hinges, and use kitchen-cabinet catches (*Photo 31*) to secure them for transit. Finally, to mount the high-frequency driver, attach a 1⅜″ pole mounting socket to the cabinet top, either recessed-style or flush mount.

THE HF BOX

The high-frequency box is just that: a simple box (*Photo 32*); no special care is required in its construction. Make it just large enough to house the driver array. When built from ½″ plywood parts, the dimensions are: top and bottom, 6¼″ × 8½″; sides, 19¼″ × 8½″; and back, 6¼″ × 18¼″. If you use a socket-pole mount, drill a hole in the bottom for it. A side-

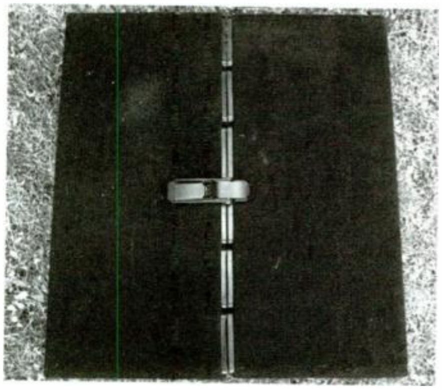

PHOTO 34: Two high-frequency boxes clamped together for transport.

PHOTO 35: Side-fire tweeter aligned vertically to minimize interaction with the main array.

fire tweeter requires another hole, while a hole on the cabinet back serves as a handle (*Photo 33*). A final hole is needed for a jack.

Finish the box as you wish, wire all the drivers in parallel, connecting them to the jack through a series-wired 2Ω 10W resistor, and screw them in place. If you wish, you may place two boxes face-to-face for travel, held together with clamps (*Photo 34*). The pair strapped together weighs less than 10 lbs! When set atop the bin, aim the front of the box towards the main body of the audience, and the side-fire (*Photo 35*) either inside the PA towards the "dead spot" directly in front of the stage, or to the outside if there is seating at the side of the stage. For maximum versatility, consider using side-fires on both sides of the box, with switches to engage them as desired.

Aluminum 1⅜" mounting poles are available commercially, but they aren't cheap; I made my own from 1¼" (inner diameter) copper pipe. You should carry two different lengths of poles. For gigs where the cabinets are elevated on a stage, a 6" or 8" mounting pole will be sufficient to get the high-frequency boxes above the heads of the audience, but when the bins must be placed on the floor, a 3′ pole may be required. Also note that placing the bins at the edge of a stage will rob response below 100Hz, so for maximum bass on a stage, pull them back from the edge a couple of feet.

PERFORMANCE

When I completed and tested this system, I found the average SPL/2.83V from 80Hz to 16kHz to be 2dB higher than my Snail II/Top Box combination. As expected, I lost a few dB below 80Hz, but I figured that with the Siamese Snail on my bass, I could live with that. The test, as is usually the case with new equipment, came not in a controlled environment, but at a live gig. And what a gig! We were playing outdoors at the Laconia NH Bike Week Tattoo Contest, with the roar of 75,000 Harleys as our customary accompaniment.

I run my sound from the stage, using LED meters at every gain stage to check on levels, so I must make an educated guess at how loud things are out front. At this gig in past years, with the Snail II/Top Boxes and 550W, I'd run it at a 0dB input to the power amps, but to be safe with the new setup, and despite having only 350W to power it with, I decided to start off at −3dB, prepared to crank it a bit more if management desired it louder. Well, sure enough, we'd only played one song before the promoter came over to my end of the stage and yelled in my ear to be heard over the roar of the bikes nearby:

"Hey, could you do us a favor? It's a little too loud. Can you turn everything down a bit? Especially the bass; some of the neighbors have complained."

Success.

SOURCES

MCM Electronics
650 Congress Park Drive
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How does changing the throat size of a horn affect the sound? This author investigates by building this midrange model.

Part 2

An Exponential Midrange Horn

By Louis C. McClure, Sr.

In Part 1 of this article (*SB* 7/99), I described a method of designing and constructing a midrange exponential horn for use with a 6½″ driver. The horn I had built used a large (30.25in²) throat, designed to equal the square of the diameter of the cone and surround of the driver. It resulted in a considerably larger throat than is customary for such a horn.

The conventional formula for the size of the throat is usually stated as $.8F_S \times Q_{ES} \times V_{AS}$, which, for the driver I used, would have resulted in a throat area of approximately 11.85in². Since the throat size I actually used in the horn was approximately 30.25in², this represents a considerable increase.

After building the large-throat horn, I was curious to see what a midrange exponential horn—designed according to the conventional formula—would sound like by comparison. I therefore decided to build another horn with a throat area of 11.85in² and a design frequency of 350Hz.

I could not use the small-throat horn and the same cutoff frequency (250Hz) because the resulting overall length of the horn (11.75″), when added to the 4″ minimum depth of the rear chamber, would not fit into the 12″-deep cavity of the bass horn I was building. Therefore the midrange horn could be no longer than 8″, and to make this possible, it was necessary to increase the design frequency to 350Hz. The mouths of the two horns are almost the same size (*Photos 9* and *10*).

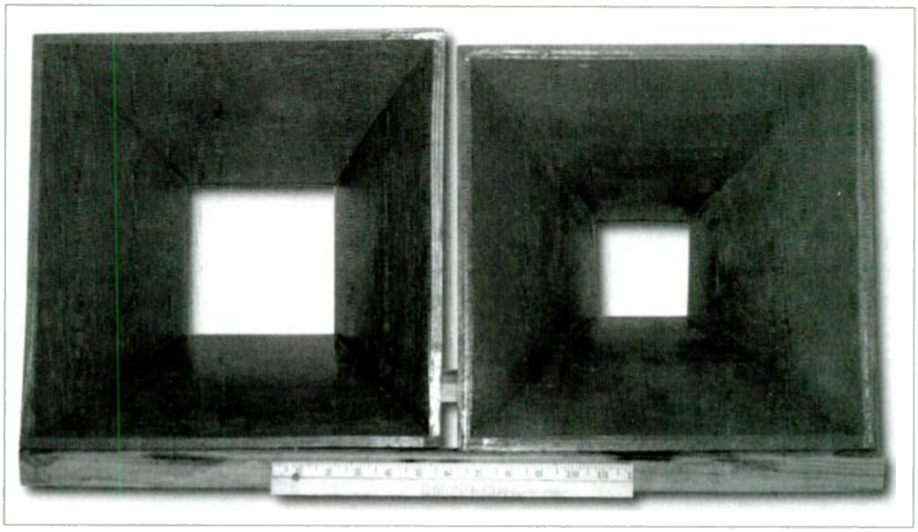


PHOTO 9: Large-throat horn (left) and small-throat horn (right) shown for comparison purposes.

THE DESIGN PROCEDURE

In making the calculations for the small-throat midrange horn (the method is described in greater detail in Part 1), you must first determine its design frequency. In this case, it is 350Hz. The rate of flare ("M" factor) is determined by the formula M = 4π × (design frequency)/13548. For this design, then, M = 12.566 × 350/13548 = .324640.

To calculate the cross section (S_x) of the horn at any given distance x from the throat, the formula is

$$S_x = \log_e(M \times x) \times S_i$$

where M = .324640, x = distance (in inches or fractions thereof) from the throat, and S_i = initial throat area.

The throat area is calculated with the formula

$$S_t = .8F_S \times Q_{ES} \times V_{AS}$$

Where S_t = area of throat at a distance x from throat, F_S = free-air resonance of driver, Q_{ES} = electrical Q of the driver (T/S parameter), and V_{AS} = equivalent volume of air to equal resistance of the cone and voice-coil mass, the stiffness of the suspension including the surround and the spider (T/S parameter).

For my horn, using the MCM Cat. No. 55-1585 driver, the T/S parameters were: F_S = 111Hz, Q_{ES} = .58, and V_{AS} =.23ft³. Thus, $S_t = .8 \times 111 \times .58 \times .23 = 11.85\text{in}^3$.

To determine the length of the horn along the center line, you must make each side of the mouth equal to ¼ wavelength at the design frequency. Continue to make the calculations at equal intervals (½″, ¾″, or 1″) until the mouth area equals or exceeds the minimum required cross-sectional area.

Since the design frequency is 350Hz, you must determine the cross-sectional area of the mouth to reproduce this frequency. Since each side of the mouth of a square horn must equal or exceed ¼ wavelength at the design frequency, you must continue the calculations of cross-sectional areas until you obtain at least the minimum required area.

Because the design frequency is 350Hz, the wavelength of 350Hz would be equal to 13548/350, or 38.71″. This means that for a square horn, the mouth must be at least one quarter of this value, or 9.68″, and 9.68″ squared equals 93.65in². In my case, this resulted in a horn length of approximately 6⅜″. However, I carried the expansion out until I reached a length of 8″, which corresponds to a minimum frequency of 268Hz for the mouth.

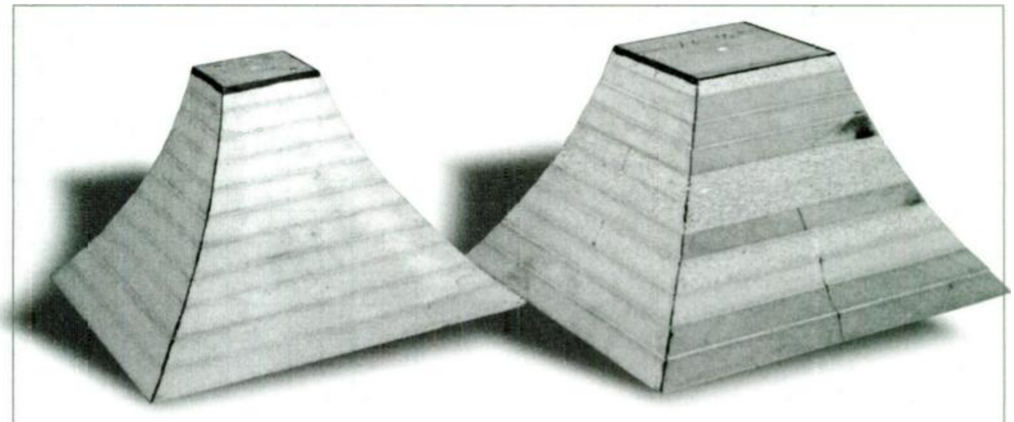


PHOTO 10: Comparison of the cores for the small-throat horn (left) and the large-throat horn (right).

You must continue calculating the cross-sectional areas until you obtain a result of at least 93.65in². You may, of course, continue to calculate greater lengths if you so desire. 93.65in² would be the theoretical minimum cross section for this horn. Extending the length would produce a larger mouth, allowing the horn to operate at a lower frequency.

In my case, I desired a length of 8″, so that when added to the 4″ depth of the back chamber, it would fit into the 12″-deep cavity in front of the bass horn.

You can construct the table of cross-sectional areas and the vertical side-elevation drawing in the same manner as described in Part 1. These two items will be very useful when you cut the pieces for the core assembly.

After constructing the table of cross-sectional areas (see *Table 1* in Part 1) and the side-elevation drawing (*Fig. 1* and *Photo 2* of Part 1), you are ready to proceed with the physical construction of the horn as described in Part 1. The construction techniques are identical.

TESTING, TESTING

In order to test the two horns, I built an adjustable rear chamber that you can increase in volume from approximately .1ft³ to .32ft³ (*Photo 11*). For the purposes of this test, I used .23ft³, the same as the stated V_{AS} of the driver.

The adjustable rear chamber, made of ½″ MDF, is approximately 22″H × 14″W × 4″D overall. You make the adjustment by moving a snug-fitting plunger into the chamber.

The driver is mounted near the bottom end of the chamber, with its face recessed into the chamber's face in order to create a flat surface. The seal consists of a strip of ⅛″ × ½″ weatherstrip foam placed on the rear flange of the horn around the throat.

To conduct the impedance and frequency-response checks, I attached a shelf to the front of a 6′ stepladder, approximately 4′ above the ground and facing upward at approximately an angle of 15–20° from the vertical. I placed the system in the yard to prevent unwanted reflections (*Photo 12*).

I installed the large-throat horn on the chamber, using elastic tie-down cords to secure it. I then ran a ⅓-octave impedance test on the unit, as well as a ⅓-octave frequency-response check (*Photo 13*).

Then I repeated the tests with the small-throat horn and plotted all the results on ¼″ quadrille-ruled paper (*Fig. 4*).

When I reviewed the results of the frequency-response checks, it occurred to me that had I used 1-octave instead of ⅓-octave intervals, the frequency response of the two horns would be almost identical. For instance, the response is virtually the same at 250Hz, 500Hz, 1kHz, 2kHz, and 4kHz. It is clear that the differences in the response of the two horns would not have shown up if I had used 1-octave intervals. The primary reason I prefer the use of ⅓-octave intervals is that it produces a higher resolution than either the ½-octave or 1-octave intervals.

I inserted an 80μF capacitor between the amplifier and the driver. Since the driver is a nominal 8Ω unit, this provided a 6dB rolloff at 250Hz, which should protect the driver from possible damage at very low frequencies.

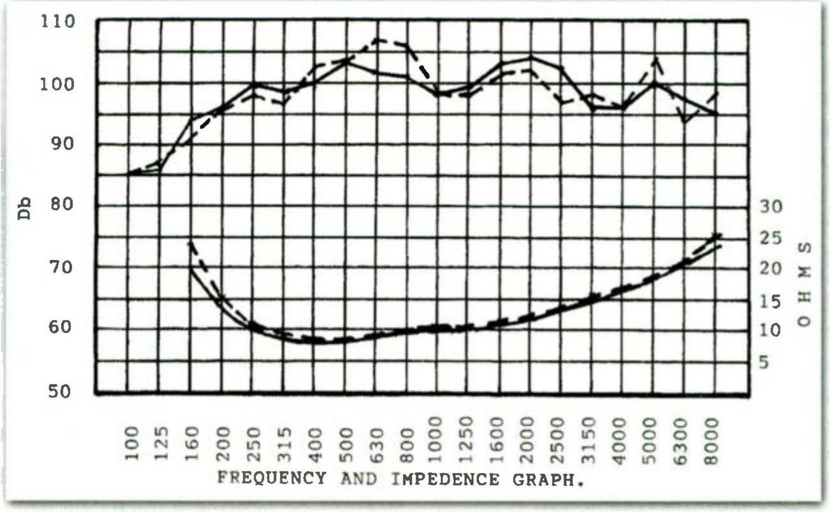


FIGURE 4: Frequency response and impedance graph. The solid line represents the large-throat horn, and the dashed line the small-throat horn. Horns were measured under identical conditions.

LISTENING TESTS

I removed the horn, and fed a music signal to the driver. The sound was very natural, clear, and uncolored. Then I reinstalled the large-throat horn. Again, the music sounded fine, but the output increased approximately 4–5dB. Otherwise, it was essentially identical to the sound produced by the driver without a horn attached.

After installing the small-throat horn, I repeated the same subjective listening test, using the same music, the same volume setting on the amplifier, and so on. I had expected the horns to sound similar, but was I mistaken. The small-throat horn sounded harsh, strident, and very much like a megaphone!

Since this was a purely subjective test, I invited several of my friends to make the comparison between the two horns. Some were music teachers or musicians, and others had considerable experience with quality audio systems. Some were younger, which meant that their hearing acuity was better than that of older persons. They all unanimously agreed that the large-throat horn sounded much better.

As a result of these subjective tests, I will use the large-throat horn in the large bass-horn system I am currently building. After examining the response curves, I plan to cross it over at 12dB per octave at 250Hz.

As I mentioned previously, the bass horn I am building has a mass rolloff at 390Hz. I had hoped to cross over to the midrange at 200Hz, but the space requirements precluded my making the

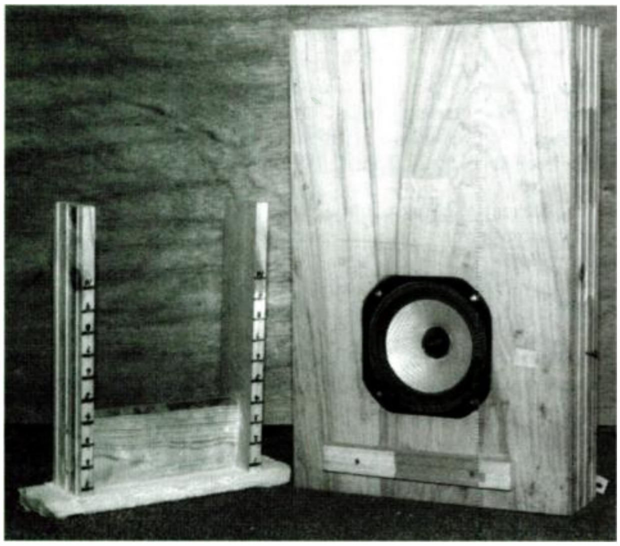


PHOTO 11: Adjustable-volume rear chamber.

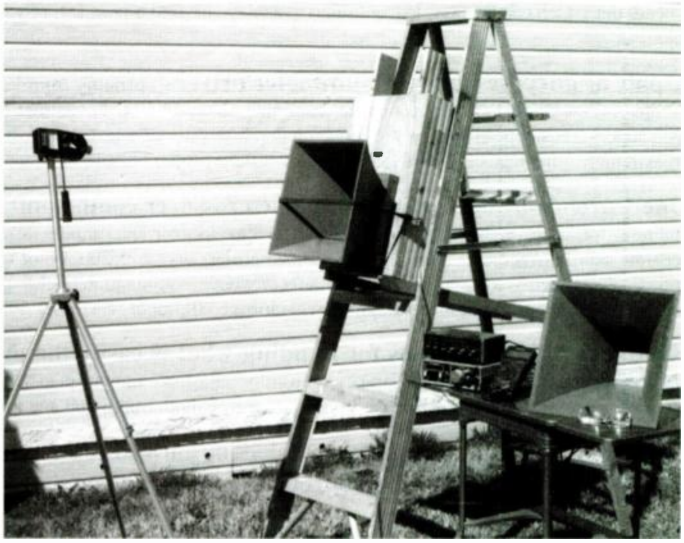


PHOTO 12: The improvised test setup using the rear chamber.

horn that long. However, I see no real problem with crossing over at 250Hz, the design frequency of the large-throat horn, and crossing over to the tweeter at 3.5kHz. However, this could change depending on further development of the system.

Throat area is an important factor in the design of a horn. Bass horns typically have a mass rolloff below 500Hz. However, most of the music is contained in the midrange, which is therefore very critical to the faithful reproduction of music, as well as being primarily responsible for the reproduction of the human voice.

Designing and constructing these midrange horns has resulted in raising more questions than providing answers. Could it be that there are other parameters not being addressed—parameters that would explain why two speakers with similar frequency response would not sound alike?

I have no qualms about others disagreeing with my own personal views. However, I would welcome theories as to why two horns would sound so different when they are designed and constructed the same way, with the only differences being their throat areas and design frequencies. Could it be that restricted throats are primarily applicable to bass horns rather than midrange ones? Or could the large-throat horn be considered a directional baffle instead of a horn, even though it is designed according to the accepted formulas for exponential horns?

Perhaps some of you would enjoy building a pair of horns such as those presented in this article, testing them for yourself, and sharing your experiences and results in *Speaker Builder*, thus possibly shedding some light on the whys and wherefores of the unexpected results I have described. In any case, I hope you have fun!

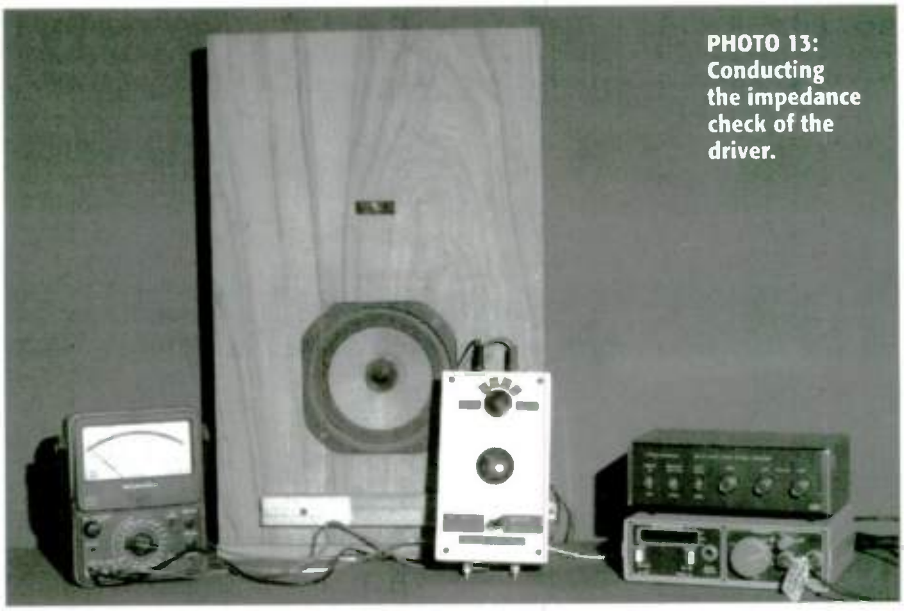


PHOTO 13: Conducting the impedance check of the driver.

Serious impedance measurement requires a hefty outlay of cash for the right equipment. Try this quick, inexpensive, but accurate method for occasional usage.

The Impedometer

By Dick Pierce

An oft-asked question is "What's a simple way to measure the impedance of a loudspeaker or driver?" The answer depends upon many things, such as what equipment you have at your disposal, how much work you wish to put into the enterprise, and so on. I'll present one method that gives reasonably accurate results requiring a bare minimum of test equipment.

WHAT IS IMPEDANCE?

Simply stated, it's the obstacle to current flow provided by an electrical circuit to the imposition of an AC electrical signal. It is like resistance in that sense, but different in that it is almost always (in these applications) frequency-dependent (its value is different at different frequencies), and it is "complex" (meaning that, mathematically, it is a vector quantity, consisting of a resistive and a reactive part).

The law governing the relationship between DC resistance, voltage, and current, known as Ohm's law, is:

$$E = I \times R$$

where E is the impressed voltage in volts across the resistance R in ohms, resulting in a current I in amperes flowing through that resistance. Simple high school algebra allows us to rearrange this basic equation. To solve for resistance, divide both sides by the current I and simplify:

$$\frac{E}{I} = \frac{I \times R}{I}; \frac{E}{I} = R$$

Similarly, to solve for current, divide both sides by R and simplify:

$$\frac{E}{R} = \frac{I \times R}{R}; \frac{E}{R} = I$$

AC impedance, voltage, and current follow the same basic rules:

$$E = I \times Z$$

where E is now the impressed voltage magnitude in volts impressed across the impedance magnitude Z in ohms, resulting in a current magnitude of I in amperes flowing through that impedance. And, as above, we can rearrange equations:

$$Z = \frac{E}{I}; I = \frac{E}{Z}$$

The AC impedance, as mentioned previously, is a complex value: it is vector sum of the resistive (or "real") and reactive (or "imaginary") components of the impedance. That vector sum is computed as:

$$Z_{Mag} = \sqrt{R^2 + X^2}$$

where R is the resistive portion and X is the reactive portion. (In this context, real and imaginary have very specific mathematical meanings: an imaginary number is not one that exists only in one's imagination; rather it is a number that has the square root of negative one as one of its factors.)

Because of the energy storage properties of the reactive portion, the instantaneous current flowing through the impedance is not in step or in phase with the instantaneous voltage across it. Rather, it precedes or follows the voltage by some amount dependent upon the ratio of reactance to the resistance, specifically:

$$\Phi = \tan^{-1} \frac{X}{R}$$

where Φ (the Greek letter *phi*) is the phase angle, usually expressed in degrees.

It should be noted that in the grand scheme of things, both the resistance R and the reactance X can take on any value—positive, negative, or zero. However, in the case of loudspeaker impedance, R will *never* be negative, and almost certainly never 0, while X can be positive (inductive), negative (capacitive), or 0.

Looking at the equation for the impedance phase angle, this means that the phase angle of the impedance will always be inside the range of –90° to +90°. (Indeed, it is quite unusual to find the impedance phase to be outside the range of ±70°.) The fact that the real or resistive portion of the impedance is always positive ensures that the impedance phase angle never exceeds these 90° limits. (For those with a more technical inclination, that means that the entire impedance is confined to the right of the imaginary axis in the complex s-plane.)

Basically, all you need to do is put a voltage across the unknown impedance, measure the current going through it, plug the numbers into the equation

$$Z = \frac{E}{I}$$

and out pops the impedance, Z.

In principle, this is absolutely correct, but in practice, it is more difficult. The main reason is the range of typical values for the impedance of most loudspeakers and drivers (from a few ohms to a few dozen ohms) combined with the sensitivity of most common measurement instruments.

Imagine putting 10V across an 8Ω loudspeaker. Ohm's law says that the current going through that speaker will be:

$$I=\frac{E}{R}$$

$$I=\frac{10\text{Volts}}{8\Omega}$$

$$I = 1.25 \text{ Amperes}$$

While 1.25A is a convenient current to measure (it's large enough to ensure reasonable accuracy with many common meters), it is a *lot* of current to put through the voice coil, and that poor speaker and the people near it will be subjected to a rather deafening level of sound. Additionally, it does pose some risk of damage to some drivers.

MEASUREMENT SCHEMES

A common assumption is that you need two meters: one to measure voltage placed across the impedance and one to measure current placed in series with the impedance. Then, by Ohm's law:

$$Z=\frac{E}{I}$$

However, this poses some problems. As mentioned previously, it requires a hefty amount of current to get enough of a reading to be dependable. Most commonly available meters that measure AC current well aren't very sensitive. There is also the issue of calculating for each and every frequency being measured.

Another method that seems to have escaped many people's attention is the "impedometer," which is nothing more than a calibrated constant-current source. When properly set up, no calculation is required, and it is reasonably accurate over a wide range of impedances. Another advantage is that it requires less equipment than other methods. I will discuss the impedometer method here.

TEST INSTRUMENTS

Very little is required for a properly working impedometer. The requirements are:

1. AC sine-wave generator. This can be either a function generator (usually an instrument with the capability of sine, square, and triangle waves, and often with pulse output as well) or a Wien-bridge or twin-T audio oscillator. The major requirements are stable AC output, stable frequency, reasonably low distortion (less than 1%), flat frequency response over the audio bandwidth, and reasonable voltage output (10V or more into 1kΩ is good).

A lot of new instruments are acceptable—function generators by B&K, Tenma, Leader, and others can be had, but often cost several hundred dollars new. Their performance is generally more than good enough, and they are versatile instruments for other purposes as well. Often they have frequency ranges far in excess of what's needed, such as 0.02Hz–2MHz, but that's okay.

On the other hand, you can often find used equipment that is very serviceable as well as inexpensive. I have seen excellent units for under $100 from the likes of Wavetek and Krohn-Hite. In working order, they have superb specifications and are ideal for this sort of use. Their distortion is not the lowest (because, like other function generators, they synthesize the square wave from the triangle output), but, for impedance and frequency-response measurements, they are superbly accurate for audio use.

One of the all-time best sine generators is the venerable HP 200 audio oscillator, which I have seen at swap meets and even at yard sales for as low as $5. They have good frequency response, good stability, and high output voltage (5V into 600Ω). There are several variants; the 200 AB and 200 CD are the most common and both are equally good.

Look for examples from General Radio (GR or GenRad), as well. You can purchase the GR1309—often for $50—and tune it to have very low distortion—under 0.05%—while the 1304 will do 20–20kHz without range sweeping and has high output voltage as well. Be prepared for a little tune-up work, such as cleaning and lubing dial shafts, or maybe replacing a tube and an electrolytic capacitor or two. Otherwise, these units last absolutely forever. I cannot recommend them too highly.

2. AC voltmeter. This can either be an analog or digital unit. Ideally, it should be capable of reading down to about 10mV full scale with reasonable accuracy. It must also have flat frequency response over the audio range.

Unfortunately, the sensitivity requirement eliminates most "passive" VOMs (volt-ohm-milliammeters), including the ubiquitous and venerable Simpson 260 (which is truly unfortunate, because the three different vintages I have all have excellent frequency response to well beyond 50kHz on the 2.5V AC scale; sigh).

Equally unfortunate is the fact that many hand-held DVMs (digital voltmeters) have poor high-frequency response, often showing significant errors as low as 500Hz.

Generally, most meters that advertise themselves as "true RMS" have adequate frequency response.

Again, turning to the used or surplus markets, there are treasures to be had. The Hewlett-Packard 400D has all the needed sensitivity (1mV full scale), wide frequency response (below 10Hz-1MHz), excellent linearity, and availability. Again, I have seen them for as low as $25 in serviceable condition, and even arrived 30 seconds too late one day as I saw 20 of them being crushed at a local landfill! Again, they may need new capacitors here and there and an occasional new tube, but little else is needed to keep them going. Most of the HP 400 series are equally useful. Look also for meters from GR, Ballantine, B&K, and others. Heath also made an AC millivoltmeter that is quite useful.

3. 1kΩ resistor. This need not be anything fancy. A noninductive carbon or metal film, ½W 5% resistor is really all you need. This will turn your oscillator into a current source.

4. 4, 8, or 10Ω precision noninductive resistor. This is used to calibrate the impedometer. It can be any value that's close to the impedance you expect to measure. Just make sure it's noninductive (film resistors work here) and that you know its resistance accurately (a 1% or better tolerance is ideal). You'll need only a small (½W is probably fine) resistor.

5. Frequency counter. Not essential, but considering that the frequency dial calibration on many oscillators and function generators can be considerably off, it's useful to have. You can pick up a digital frequency counter on both the new and used market for not a lot of money. Remember that the accuracy is directly proportional to the reciprocal of

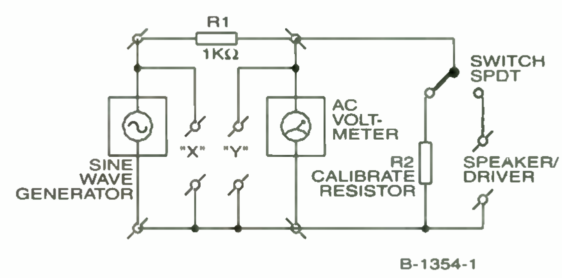


FIGURE 1: Connecting the impedometer.

the needed accuracy: if you require ⅒Hz accuracy, you must wait ten seconds to get there.

6. Oscilloscope. Not essential, but useful for several things, such as verifying that nothing is being distorted. Connected as an X-Y scope, it can help you unambiguously find the exact resonant frequency and also enable you to (via a rather laborious procedure) estimate the phase angle of the impedance. If you're going to get a scope, get one that has X-Y capability with no less than 10mV/cm on both axes. Scopes fitting the bill can be found for anywhere from $50 used to many tens of thousands of dollars. Look for an old HP 130, the best audio scope for the least money around. There are some big-ass Tektronix 500 series that are huge and cheap; also look for 400 series, and scopes by Philips and others.

7. Miscellaneous. If you're going to be doing this a lot, buy a metal box, some good five-way binding posts, and some high-quality switches to make your life easier. Use good-sized wire, because a ½Ω of parasitic resistance in your test harness is a ½Ω that *won't* be there when you connect your crossover.

MAKING AN IMPEDOMETER

The actual connection is very simple (*Fig. 1*). If you need more signal level, you can insert an amplifier between the oscillator and the 1kΩ resistor.

Calibration is simple: connect the calibration resistor to the output (via the switch, if you've constructed it that way, or just hook the resistor where the speaker would be connected). Adjust the output of the oscillator and the gain of the meter until you get a reading in some convenient units that is the same as the resistance of the calibrator. For example, if the oscillator was putting out 1V into the 1kΩ resistor, you'd probably find that the voltage across an 8Ω calibrator was almost exactly 8mV.

Fine, now you know that your impedometer has a calibration factor of 1mV/Ω. This is because you have calibrated your AC current source for 1mA output. Remember Ohm's law:

$$Z = \frac{E}{I}$$

I = 1mA, so Z (ohms) = E (millivolts)

You might prefer to adjust it for a higher level, such as 10mV/Ω. You see here the need for an oscillator with a nice high output voltage, because you might wish to measure the impedance at several different current levels. (I have a laboratory amplifier that's capable of more than 100V at 100mA into a 1kΩ load. This is very useful for measuring drivers at reasonably high current levels.) It's a good idea to check the calibration across the entire frequency range.

MEASURING IMPEDANCE

To actually measure the impedance, make sure your setup is calibrated, then disconnect your calibration resistor and connect your speaker. Dial the oscillator to the desired frequency and then read the impedance. It's that simple.

If you prefer to know the impedance across the whole frequency range, it's good to measure it at ⅓ octave intervals. This will be enough to plot a pretty accurate graph of the impedance curve. Here are the standard ⅓ octave frequencies:

20	200	2000
25	250	2500
31.5	315	3150
40	400	4000
50	500	5000
63	630	6300
80	800	8000
100	1000	10000
126	1260	12600
159	1590	15900

To find a resonance, look for a frequency where the impedance is at a maximum. In a typical loudspeaker system or a bass-reflex enclosure system, you'll find several such maxima. Record them all. Look for other "critical points" such as minima in impedance.

An oscilloscope can be useful here. Connect the X-axis to the oscillator output (shown as "X" in *Fig. 1*) and the Y-axis to the same place as the AC voltmeter ("Y"). Adjust the gain of the X-axis so that the trace takes up nearly the whole width of the screen. The Y-axis gain can be similarly adjusted, but you'll need to keep changing it as the impedance changes.

You'll notice that, over most of the range, the trace is an ellipse aligned along a line going from the lower left to the upper right (if it goes in the opposite direction push the switch on the scope labeled "phase invert"). The elliptical shape indicates that the impedance has both a reactive and a resistive component. In fact, you can measure the phase by measuring the relative "openness" of the ellipse, though I won't go into that here.

What *is* important is that at some frequencies, the ellipse closes up into a line. This indicates the impedance is purely resistive, and will occur at the exact center of a resonance, so it is a reliable way of finding the resonant frequency.

The trace can also tell you other things. If the ends of the ellipse are flattened or distorted, it's likely you've exceeded the output voltage capability of your oscillator or amplifier. Sorry, only one way to fix it: turn it down and recalibrate your impedometer. If the traces show a figure-eight shape, especially near and at resonance, you're likely looking at some nonlinearity in the driver itself. Finally, if your trace looks fuzzy or has lots of little wiggles on it, you have an electrical interference problem that you'll need to cure.

CONCLUSION

The impedometer method provides a simple, inexpensive, reliable, repeatable, and reasonably accurate way of measuring loudspeaker impedance, assuming you use reasonable instruments and take care to check and maintain calibration.

There are certainly more streamlined methods, including new computer-based applications that are fast, very detailed, and accurate. Not everyone can afford such a solution, not everyone has the time, and not everyone needs that level of sophistication. The impedometer method is useful for occasional measurements, and the equipment needed is quite useful for an entire array of audio measurements.

In the third part of this series, the author shares his findings on how effectively loudspeakers convert energy into music.

Part 3

Navigating Speaker Design: Listening to Walls

By Mark Wheeler

Once upon a time, a handful of people with nothing better to do were persuaded by this slightly obsessed speaker builder to participate in some listening experiments. It was autumn or winter, so they couldn't go rock climbing, and they offered to participate. After all, a succession of afternoons listening to their favorite music seemed like a good idea at the time.

Several of these listeners were used to regular doses of live music, both amplified and *au naturel*. A couple of them had serious hi-fi systems of their own, and others played musical instruments, sometimes even quite musically. These young men and women were all accustomed to my asking them to spot the differences between two apparently similar bits of hi-fi equipment, or two recordings, or even between two capacitors.

EXCESS ENERGY

At the time, I was occupied with the problem of unwanted energy in loudspeakers, a phenomenon that has been the staple diet of numerous book chapters and journal articles for many years. Ever since I was a teenager interested in music and hi-fi, I have been struggling through a major philosophical dispute with this whole idea of unwanted energy. I clearly recall being taught in school about the conservation of energy—that it is neither created nor destroyed, but merely transformed.

I've looked very carefully at my loudspeakers. I've walked all the way around them many times. I have picked them up, turned them over, and inspected them from every angle, and the only place where I can see energy entering them is via the input terminals. I am pretty sure about this, and am also confident that the rules of physics are not being broken. I have gone to great lengths to ensure that the expensive electrical energy delivered by my amplifier does not contain any "unwanted energy." Therefore, any phenomenon observed as energy not employed in the business of making music is not so much unwanted energy as diverted energy.

When it was delivered to the loudspeaker input terminals, this energy was the opposite of unwanted energy. This was once the electrical energy that contained the very nuances of a violin bow upon the strings, the thwack of stick on snare-drum skin, the wail of overblown tenor sax... you get the idea.

Some of that useful energy becomes diverted into a more destructive activity. Perhaps it should be called "delinquent energy," but it has no free will to choose to become destructive, so I'll call it "diverted energy." Somewhere between input terminals and acoustic output, this energy is diverted from its purpose of

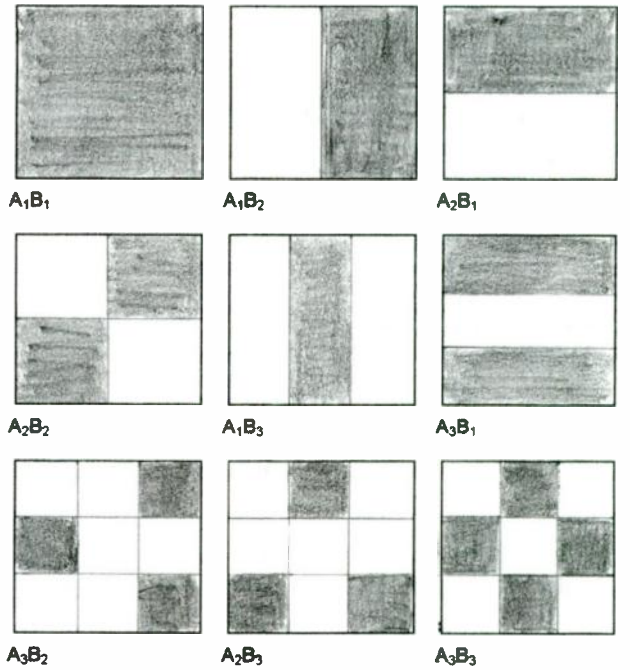


FIGURE 1: Simplified "clamped plate" panel resonance sequence to third order.

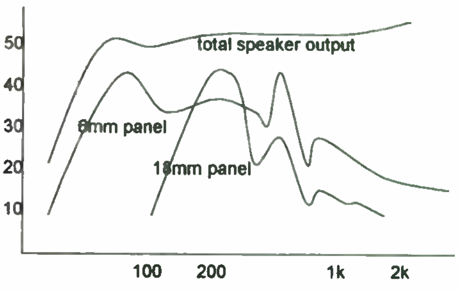


FIGURE 2: Sound output of loudspeaker with square plywood panels, smoothed and simplified for clarity.

transmitting music into the realms of distortion, coloration, and information loss.

A number of established manufacturers have attempted to ensure that as much as possible of this electrical energy is converted to sound energy without being diverted. One direction from which to approach this is to reduce the moving mass that the electrical energy must drive. These attempts have often been very successful, but they are also expensive to execute. They include various historic ultra-low-inertia drivers like the Ionofane, as well as ribbon, electrostatic, and some "plasma" tweeter designs.

REDUCING MASS

Moving-coil loudspeaker types get their share of the attention. Various approaches to design and construction have attempted to minimize the side effects of applying force to that moving mass. One obvious method is to reduce moving mass as far as possible by using exotic cone/dome materials. Another commonly applied method increases the mass to which the movement is referenced.

One notable commercial approach has in been KEF's models, which have included pairs of bandpass bass drivers designed to work as a closed mechanical system isolated from the supporting structure. The bass drivers share a common axis and are connected by a steel bar so that the momentum of each moving mass is balanced by the other within their operating bandwidth.

KEF has also included compliant midbass driver mountings on several models for years. More radical commercial attempts to ensure the path the energy takes from input terminals to music have included the Roksan Darius loudspeaker, with its exoskeletal stand and compliant tweeter suspended on springs from the front baffle.

The Naim SBL (separate-box loudspeaker) lives up to its name by featuring for each woofer and tweeter separate driver enclosures mounted directly to an external steel chassis/stand. The Naim SBL effectively couples the drive units to the metal stand while using compliant gaskets to isolate them from the box. This separates the driver frames from the big problem zone of the bass-cabinet panel areas.

The drive units are isolated from one another, but are accurately referenced to ground through the steel stand, so most of the electrical energy imparted to each drive unit is converted into music. This means that the energy has a very direct path, with few opportunities to stray into delinquency. The musical impact of this speaker is quite stunning, and was a real surprise when I first heard a pair several years ago.

All these attempts to make moving-coil loudspeakers behave effectively cost serious money, since construction techniques are complicated and thus labor-intensive by mid-market standards. The resulting products remain in a niche at the top of the market. The $3500 Naim SBL, for example, looks like a very small, paper-coned, 8″ bass-driver, two-way loudspeaker, so it remains a product with a limited market. However, labor costs do not concern those who build their own. To find out how these kinds of ideas actually affect the sound quality, I set up a series of listening tests.

THE LISTENING TESTS

The intention of these "listening panel" experiments was to find out which aspects of loudspeaker design and construction have which audible effects, and, further, which of those effects contribute to ensuring that a good proportion of the desirable musical energy will be expended in achieving its original purpose, rather than being sidetracked into antisocial coloration (*Fig. 1*). The measured responses of these panel tests are shown in *Fig. 2*.

The usual habit of speaker builders is to attempt to ameliorate the problems caused by this energy once it has already been diverted from its musical purpose. This often results in converting it into heat by the friction in lossy masses of damping materials. Instead, I intended to investigate which approaches to loudspeaker system design do the most effective job of ensuring that more of the expensive amplifier energy becomes music. A good place to start seemed to be that old chestnut: cabinet wall-bracing. We began by asking the following questions:

- Does loudspeaker-enclosure wall bracing have any useful effects?
- What is the nature of those effects, and are they in respect of the diverted energy?
- Does the bracing merely help reduce the effects of diverted energy, or does it actually help maintain the original purpose of the loudspeaker?
- Does it matter where the bracing is?
- Does it matter what the bracing is made of?
- Does it matter how the bracing is fixed

into the cabinet?
- Do the enclosure's wall materials have any noticeable effect on the efficacy of the bracing?

Lacking any test gear more sophisticated than a sound-level meter and several pairs of ears, the series of sociable "listening panels" took shape.* They occurred over a considerable period of time, beginning more than a decade ago, and were conducted entirely out of curiosity, with neither commercial backing nor particularly rigorous scientific method. Indeed, in scientific terms, their methods and results would be invalid. Although the results should be read as accounts of amateur dilettantes playing around at their hobby, the experiments were conducted in a spirit of open-minded curiosity, and their outcomes have been demonstrated as consistently repeatable in the intervening years.

INITIAL TRIALS

Our trials began fairly casually in pursuit of instant gratification of "what if..." types of questions. As these vague forays into uncharted territory appeared to be increasingly conclusive, it became obvious that a more thorough methodology would be needed if the results were to have any more utility than developing personal prejudices.

Many commercial boxes have very little bracing, and these seemed ideal candidates for "before and after" comparisons. More important for this exercise was to establish whether any pattern would emerge that distinguished between different designs of cabinet reinforcement and different combinations of materials. We asked:

- Does the bracing act as a chassis on which the cabinet is constructed?
- Does bracing serve to stiffen cabinet walls, reducing their potential to be excited by vibration?
- Does bracing serve to make cabinet walls function as smaller panels (as the old "clamped plate" theory often implies), reducing the amplitude of vibration, but increasing its frequency?
- Does bracing serve to damp cabinet walls, reducing the amplitude of excitation by friction losses?

The answers to such conundrums have already been pursued by a number of researchers, who reached a variety of often conflicting conclusions, despite using equipment and methods much more sophisticated than our own. The range of their conclusions may indicate the variety of their original hypotheses under test. Our exercise hoped to begin without any hypothesis more elaborate than "Bracing in loudspeakers may have some kind of effect that might be useful in contributing to a loudspeaker's ability to reproduce music."

A number of existing loudspeakers were suitably butchered. Their construction materials varied between veneered chipboard, veneered fiberboard, veneered softwood, and solid hardwood. Each speaker was auditioned by a group of victims against a single reference that was chosen for its consistency above any other quantities. Each group included

*The ideas that prompted these experiments have often arisen from combinations of the ideas of other people. These ideas have been mercilessly plagiarized and hurled into the melting pot together. Considerable research has been conducted over the years into the behavior of individual panels. Some of this early-published research misleadingly assumed the loudspeaker-cabinet wall/panel to be a "clamped plate," but later research has demonstrated this was not the case (Chapter 7 of Martin Colloms' book *High Performance Loudspeakers* has an excellent summary and includes references well worth pursuing). Cabinet-wall analysis has become more sophisticated than the old trick of covering panels with powder and watching it move when a signal is applied!

the owner(s) of the speaker to be modified. For each example, the same group would listen to the same speaker in its before, after, and in-between versions.

The same music would be played for the same group. The music would simply be the familiar favorite recordings of group members, plus some "difficult" pieces that had previously separated speaker sheep from speaker goats. Partnering equipment would obviously be identical, as would position, volume settings, temperature, time of day, and zero alcohol.

RESULTS

The point of the exercise was to establish the subjective effect of each modification. The information from each panel member remained expressed in the form of opinions. Although some attempts were made to ascribe collective numerical values to various aspects of subjective performance, I rejected these attempts because the most useful information seemed to arise through the opinions of the listeners.

Attempts to quantify such qualitative data were marked by a tendency towards homogeneity and by the loss of the nuances of listeners' opinions, but the numerical scores which group members awarded to particular performance parameters were used to validate the opinions.

All the listeners in all the groups noted significant differences between examples of different bracing types and materials. The locus of the affect lay predominantly in the articulation of each speaker. Complex wideband music was described as more affected than simple solo or small ensemble pieces. Bass content (both amplitude and frequency) seemed almost proportional to the magnitude of the effect, e.g., "dub reggae" produced the clearest listener comments compared with more ambiguity in the responses to a string quartet.

The noted differences were generally in the nature of clarity and the ability to differentiate between certain aspects of the music and the performance. Frequency-response sweeps showed no discernible changes, and I did not have access to cumulative spectral-decay measurements.

I have not explored hypotheses to explain these differences, although they may be due to a reduction in the "noise" of cabinet coloration. They may also result from improvements in the rigidity of the drive-unit mounting, implying that its position in space is better defined at audio frequencies. The main problem with hypothesis generation under these circumstances is that each change is only one change in a *system*, and its effects may be interdependent with other aspects of the system.

INFLUENCE OF MATERIALS

One of the most interesting results, particularly for the home builder, was that in most examples the influence of materials was greater than that of design or location of bracing structures. The effects of wood and types of composite board were consistent and repeatable in this respect, demonstrating a hierarchy of effectiveness. The results of the two metals tested (aluminum and steel) were too inconsistent to draw any useful conclusions. Mid-panel front-to-back or side-to-side braces were ineffective enough to elicit no comments regarding consistently observed differences.

The listeners found that vertical braces had more effect than horizontal or even circumferential ones. All the example speakers had longer vertical than horizontal dimensions, so this last finding may reflect this fact. A number of designers, including Robin Marshall (formerly founder and designer at Epos Loudspeakers and subsequently designer at Mordaunt Short Loudspeakers), have reported the greater effectiveness of bracing the long dimension of a panel rather than the shorter, because this pushes the first and subsequent panel resonant modes to higher frequencies. Our experience supports this.

The choice of adhesive was found to be as critical as the choice of bracing material. The success of each material was consistent among cabinet types and materials, the differences between them varying in magnitude similarly to the fundamental differences caused by adding bracing. The popular PVA wood glue (often known as "white glue") proved to be one of the least effective with every type of wood or composite bracing material.

Hot-melt craft glues were sufficiently inconsistent in performance to make conclusions dangerously speculative. Two-pack hard-setting adhesives were the most successful, including epoxy types. One that repeatedly outperformed the others was Evo-stick Aerolite 306, purchased as formaldehyde powder that must be mixed with water and a formicacid hardener. Domestic use of such materials is potentially hazardous, but it did sound good.

HARDWOOD SUPERIORITY

High-density hardwoods consistently outperformed lower-density woods or particleboard. Inexpensive softwood, e.g., red deal, was virtually ineffective as cabinet bracing. The performance of various forms of particleboard depended on the material of the cabinet walls, but they were always outperformed by high-density hardwoods. Aluminum and steel gave inconsistent results, varying wildly from excellent to worse than no bracing at all. Indeed, the metals were the only materials for which the brace design, in relation to each cabinet, was the more important factor.

Especially curious was the finding that very thin strips of the best hardwood example performed almost as well as more substantial pieces, depending on location. After discovering this, we conducted comparisons between some hardwood samples with the masses equalized. We used a denser material with a thinner section to ensure that its mass was distributed across each panel in a manner similar to a lighter wood. The denser examples still bettered the less dense, but this approach factored out differing masses.

We tried the hardwood braces using increasingly thin sections to find out whether there was a cutoff point in their effectiveness. When we attached thin strips to the rear of the driver baffle, we observed one of the most dramatic differences. These braces, only ¼″ thick, were joined with the two-pack adhesive, connecting between the drive-unit bolts (T-nuts) and the lower corners of the front baffle where the stand is located on the outside. The result was so outstanding that we tried additional samples.

BRACING MY SYSTEM

The thin-brace series of experiments culminated in modifications on my own favorite active speaker system, a pair of 80-ltr 18mm MDF enclosures already substantially braced horizontally with MDF. The baffle was already braced with the same MDF as the cabinet, and the additional hardwood braces were merely ¼″ thick.

At the time, this was my main audio system, to which I listened almost every day before the modification without noticing any shortcomings. The differences were clearly detected by the listeners, who all described the same phenomena: vocal lines were clearer and easier to follow, individual instrument parts could be followed more explicitly in large ensembles, and bass-heavy entries became less likely to overwhelm other more delicate instruments. I'd be embarrassed if this ends up sounding like a consumer hi-fi magazine product review, so here is a summary of the observations:

- bracing had a significant and repeatable effect on loudspeaker performance;
- the effect was described in similar orders of magnitude as driver selection;
- materials seemed to be more significant than any other factor;
- adhesives were as influential as the bracing material itself;
- hardwoods consistently outperformed softwood and particleboard;
- a hierarchy appeared to exist even among hardwoods, roughly proportional to density (even when mass was equalized);
- metals deserve more research, being at

the extreme ends of the observed effects;

- brace-thickness differences seemed not as significant as material differences;
- in our samples, vertical braces were more effective than any other types tried;
- circumferential bracing was less effective, despite being the most common advocated; and
- mid-panel-to-mid-panel braces across the cabinet void were the least useful.

CONCLUSION

The home builder has considerable influence over the effectiveness of the loudspeaker in the domain of cabinet construction. Common practice in the mass-market commercial world may have as much to do with the wisdom and traditions of furniture manufacture than established acoustic criteria. However, the cost of wooden-cabinet manufacture is likely to direct larger manufacturers to concentrate their research on other methods of construction, which may bring about improvements over anything you can achieve at home.

Our experiments do indicate that received wisdom may be suspect regarding traditional cabinet construction. Cabinet-bracing tests did demonstrate specific audible differences. The types of bracing proved to be more significant and more clearly noted than has often been described previously. The differences were similar in nature to those often observed between high-quality raw drivers. This is one area where both the small manufacturer and the home builder can make their most cost-effective personal design statements.

The home DIYer has the time and the wherewithal to make loudspeaker systems that more effectively convert the input-signal energy into sound—and this is the purpose of the loudspeaker.

Sometimes two heads are better than one. Here, this veteran duo investigates how a passive crossover affects bass directivity.

Mating Subs and Satellites via Passive Crossovers

By Charles Hansen and G.R. Koonce

INTRODUCTION, BY CHARLES HANSEN

My stereo system speakers consist of a pair of NHT SuperOnes and an SW2P-powered subwoofer. The subwoofer amplifier (SA2) is not built into the speaker, but is a separate unit, with a variable low-pass (LP) active filter and polarity switch. I suppose this approach was easier for NHT than redesigning the original version of the speaker, the SW2, with its built-in passive crossover (CO). It also gives you greater flexibility, since you can add an additional sub to the amplifier by using the SW2Si slave speaker (no internal CO).

A problem arises with my present setup when I need to test power amplifiers for *Audio Electronics* and *Glass Audio*. For a proper audition, I need to connect the amplifiers to a set of full-range speakers to evaluate their stereo imaging and soundstage and their overall response from low bass to high treble. To do this, I need to trundle the units over to my cousin, who happens to have two of the original NHT SW2 subwoofers, and connect the test amplifiers to his own NHT system.

This is all well and good, since we have a great time doing it, but with family obligations and work, it is not always easy to set up a mutually convenient audition schedule. Also, if the amplifiers are particularly massive and fragile, like the Atma-Sphere M-60 Mk. II OTL tube amps I once tested, it becomes quite a physical chore.

Looking for a solution, I sent an e-mail to G.R. Koonce, asking if it were possible to design a simple passive crossover that I could connect between my power amplifier and the SuperOnes and SW2P/SW2Si pair. Herewith is our trail of e-mails, beginning with his first reply, which shows the complexity of what I thought was a simple request.

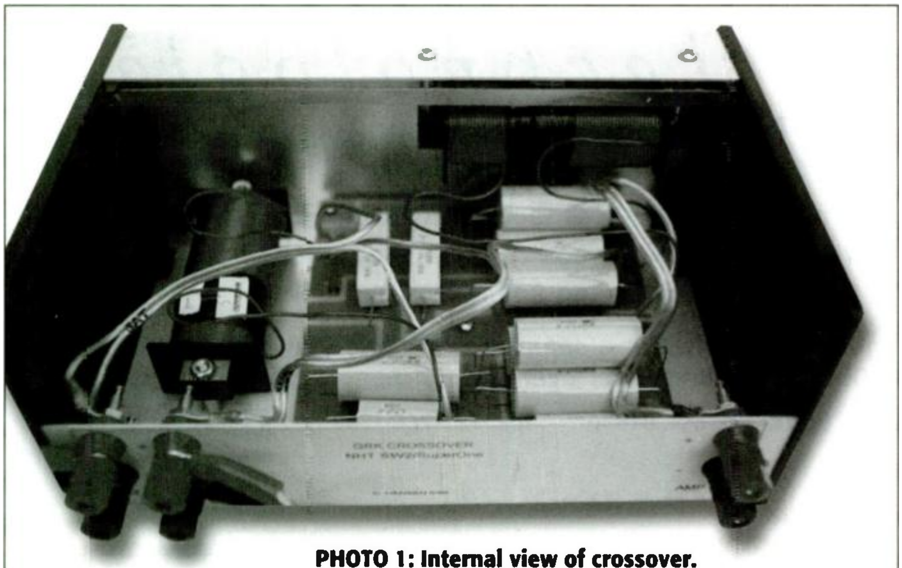


PHOTO 1: Internal view of crossover.

G.R. KOONCE 5/19/99:

What does it take to match the SuperOne 86dB/2.83V/m? It should be simple, but it is not, because:

1. Manufacturers list the SPL in different ways. Some show power sensitivity (actually efficiency) in dB/W/m, while others show voltage sensitivity as dB/2.83V/m. The two are equal for an 8Ω woofer. You also get different results depending on whether the manufacturers do the test at a single frequency or wideband. Any variation in cone mass or magnet strength results in a value different from the catalog value.

2. Then there is the diffraction-spreading loss (DSL). At high frequency, a speaker box radiates only into the forward hemisphere. When the frequency drops to where the box width matches the wavelength, the box becomes omnidirectional, and you get a theoretical 6dB drop in forward on-axis response. If the satellite upper end stops at 400 to 800Hz, then probably only the woofer will see DSL. My experience for a floor-standing box is that up to a 3dB correction for DSL sounds natural, while a full 6dB sounds unnatural.

3. There is the matter of personal taste. Many speakers are designed with the midrange suppressed a couple of dB to give a laid-back or "Cambridge" sound. If the midrange level matches the bass level, this results in a higher-presence "California" sound, which I think is more realistic.

4. The resistance of the woofer LP CO network will cost about 0.5dB in woofer sensitivity.

This all indicates that the woofer sensitivity should be in the 86.5dB–89dB/2.83V/m range. Run the input impedance curve for the SuperOne from 100Hz–2kHz. We need data to design a SuperOne high-pass (HP) CO.

C. HANSEN 5/19/99:
My NHT SuperOne speakers are spec'd as follows:

- 2-way, 7.4-ltr acoustic suspension, (no port or duct);
- 6.5″ woofer, 1″ ferrofluid-cooled dome tweeter;
- Crossover: 2.2kHz, 6dB/octave high-pass, 12dB/octave low-pass;
- Response: 57Hz–25kHz ±3dB;
- 86dB/2.83V/m, 25W min, 150W max;
- 8Ω nominal, 6Ω min;
- Outside dimensions 11.65″ H × 7.25″ W × 8.5″ D.

The bottom of the woofer flange is ½″ above the base of the cabinet exterior. Both drivers are flush-mounted and on center. I use them on 24″ speaker stands.

GRK 5/20/99:
The fact that the SuperOne has its woofer mounted at the bottom is good news. The SW2 could replace your speaker stands. In addition, you can slide the SuperOne forward and back a little to change the time delay between the woofer and the rest of the system, which can have a great effect on sound.

Based on the impedance curve of the NHT boxes, we could try to pick a crossover frequency that matches the DSL to the SW2 box width and stays away from the SuperOne crossover. The fact that the SuperOne crossover is up at 2.2kHz is also good, since it should not bother the impedance down in the 500–800Hz range. For the crossover design, we need the SuperOne impedance showing both magnitude and phase. The SW2 sensitivity of 89dB/2.83V/m should be a good match for the crossover.

CH 5/21/99:
I don't know whether you need this information, but the SW2P subwoofer is spec'd as follows:

- 1.76ft³ vented (duct tube with a tapered mouth);
- 10″ long-throw polypropylene cone woofer;
- Response: 25Hz–500Hz ±3dB;
- 89dB/2.83V/m, 50W minimum, 200W maximum;
- 6Ω nominal, 4Ω minimum;
- Outside dimensions 16″ H × 16″ W × 16″ D.

The 150W subwoofer amplifier (separate, not built into the box) has an adjustable 35–150Hz third-order LP filter. It also has a selectable 12dB/octave HP filter for the satellites, but I don't use it, because it adds coloration to the highs that I don't care for. I run the SuperOnes direct from my power amplifier.

GRK 5/22/99:
The impedance of a system with a crossover does vary quite a bit, mainly at low frequency, because of the driver's motional impedance and box effects. A closed box should have a single high-impedance peak at some low frequency, and then another anomaly near the crossover frequency. The phase below the low-fequency (LF) peak should be inductive, and then it should go capacitive for a while just above the peak. Then it will rise through zero and go inductive with the transition near the 6Ω impedance minimum around 200Hz.

Once you near the 2.2kHz crossover, there is no way to predict the phase. This is why doing a passive crossover to the SW2 in the 80–100Hz range is so challenging. Hopefully, the impedance is fairly flat and resistive in the 500–800Hz region.

For a realistic "California" sound with proper midrange presence, we wish the woofer to match the SuperOne, and the SW2 sensitivity of 89dB/2.83V/m should work well. A vented box such as the SW2 has two peaks in its LF impedance, and the frequency of the minimum impedance will be between them. The fact that it goes to 500Hz is a definite plus.

CH 5/23/99:
Enclosed is data for the SuperOne im-

TABLE 1
CROSSOVER VALUES VS. FREQUENCY

FCO	250HZ	300HZ	350HZ	400HZ
L1 (mH)	6.3	5.25	4.5	3.9
C1 (μF)	64.3	53.6	45.9	40.2
L2 (mH)	5.9	5.0	4.24	3.7
C2 (μF)	68.2	56.8	48.7	42.6

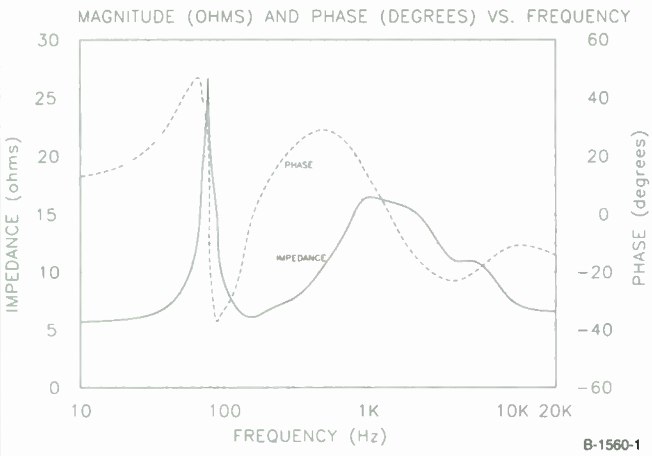


FIGURE 1: SuperOne satellite-speaker impedance magnitude and phase.

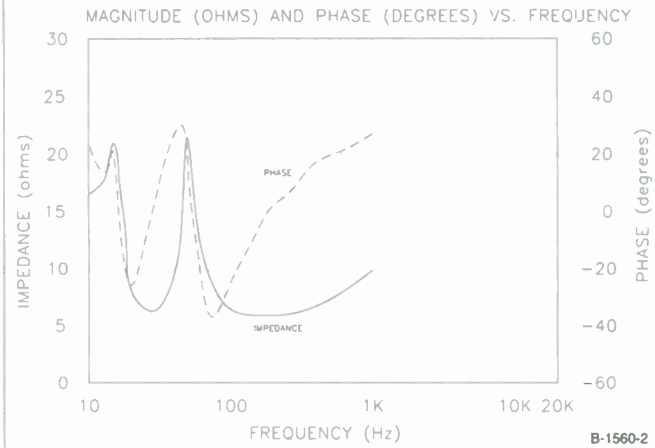


FIGURE 2: SW2 subwoofer impedance magnitude and phase.

pedance and phase from 10Hz–20kHz (*Fig. 1*), using 0.1Ω for sensing current. I captured the sine waves on my Pico ADC-216 DSO, and inverted the scope-current waveform to correct the polarity difference, which gave me the time between sine zero-crossings (Δ μs) to calculate phase angle. Scope ground is at the speaker's black terminal, which is the "low" side of the voltage across the speaker and the "high" side of the series current-sensing resistor. Since both scope channels share a common input ground, this results in an inverted current waveform.

The boxes for the SW2Si and the SW2P are the same, with only a driver and no crossover. The earlier SW2 has the passive crossover. I can get directly to the driver from the binding posts. As far as I know (from the published specs for all three SW2 variants), they use the same driver.

I will run a set of impedance-phase curves for the SW2P box I have.

GRK 5/24/99:

The SuperOne data seems to be closest to a "resistor" in the 250–500Hz range, though with some phase angle. With the SW2 going to 500Hz as the data indicates, there is a good possibility of mating the two with a passive CO.

We need the impedance and phase of the SW2P or SW2Si without the crossover. You also must drive the subwoofer directly (without the power amplifier and its active crossover) and see whether it really goes to 500Hz with a sound you like. Keep in mind that as the CO moves up in frequency, the placement of the SW2 versus the SuperOne becomes critical. The best effect, again, will be with the SuperOnes sitting on the SW2, doing away with their stands.

CH 5/24/99:
Enclosed is the SW2 impedance and phase data at the voice coil, from 10Hz-1kHz (*Fig. 2*). The test setup is the same as for the SuperOne.

GRK 5/25/99 AM:
I will start to work on a CO design based on the impedance and phase data for the SW2 and SuperOne. The normal approach would be to measure the response and impedance for both boxes and export the data as files. Then a modeling program would predict the acoustic response of the pair and let you develop a CO. We don't have this option.

We must work the "old" way. You must make each portion of the system look like a resistor. If you think of a driver as being a series resistor and inductor, then you can put a conjugate network, called a Zobel, across it to make it look like a constant resistance. Unfortunately, because of eddy currents, a driver really looks like a series resistor and a semi-inductor whose impedance rises as the square root of the frequency. The Zobel developed by the equations may not work well, so sometimes you must play with the Zobel values by trial and error.

Once you get the loads nearly resistive, then you can use filter theory to produce an LP and HP that does what you want. We will start with a second-order, which lets us manipulate the shape of the response near the CO frequency by playing with the Q of the networks. This gives you some flexibility (or more variables, depending on how you look at it).

The second-order has the problem, in theory, that it produces a null right at the CO frequency. The recommended cure is to invert the connection to the upper-end drivers. This ignores the fact that what you really care about is the acoustic CO shape. When designing the electrical filters, you ignore the driver responses (magnitude and phase) and the effect of the physical separation between them. In the old days, we developed the Zobels, designed the LP and HP, and then tweaked the component values and driver polarities—and in a case like yours, the sub position—until we achieved acceptable sound. Sometimes you must go back and start over with a different CO frequency or order. This is what you are in for.

I will calculate the starting values for the Zobels and HP and LP, but the probability is low that they will work right away. It will help if the SuperOnes sit on the SW2s, since we are making the sub into a woofer, and its position will now affect the imaging. You can add or subtract capacitance, and add or remove turns from the inductors to vary the initial values by 10% to 20%. Since you have an LC bridge, making measurements is easy. Doing it once may be fun. About the fourth time, you can see why computer design or the electronic active crossover is so popular.

GRK 5/25/99 PM:
I have gone through the computations and have the initial CO values. Both the SuperOne and the SW2 impedances indicate the CO should be in the 200-500Hz range. The higher the frequency, the smaller the CO components, but the lower the frequency, the less the effects of placement on the SW2. *Table 1* lists the values for four frequency choices. All the designs are second-order Butterworth CO sections.

First the Zobels. These are the same for any CO frequency. What you might do is look at Z_{in} in the 200-500Hz range and see that these Zobel values give nearly a resistive input (phase less than ±10°) of about 7Ω for the SuperOne, and 6.6Ω for the SW2.

The SuperOne Zobel is 7Ω in series with 25μF. The SW2 Zobel is 7Ω in series with 20μF. The resistor wattage should be half the average wattage you plan to put into the system. The capacitor voltage should be

$$V = \sqrt{7W_{peak}}$$

where W_{peak} = peak power (watts).

The CO for the SuperOne is C1 in series with the input, then L1 in shunt with the Zobel. The resistance of L1 is not as critical as it is with the SW2, and probably up to 0.8Ω is okay. Air core would be the best choice, but the biggest Perfect Lay Winding type offered is 5.5mH and 0.63Ω.

The CO for the SW2 is L2 in series with the input, then C2 in shunt with the Zobel. The coil should have low resistance—an absolute maximum of 0.4Ω.

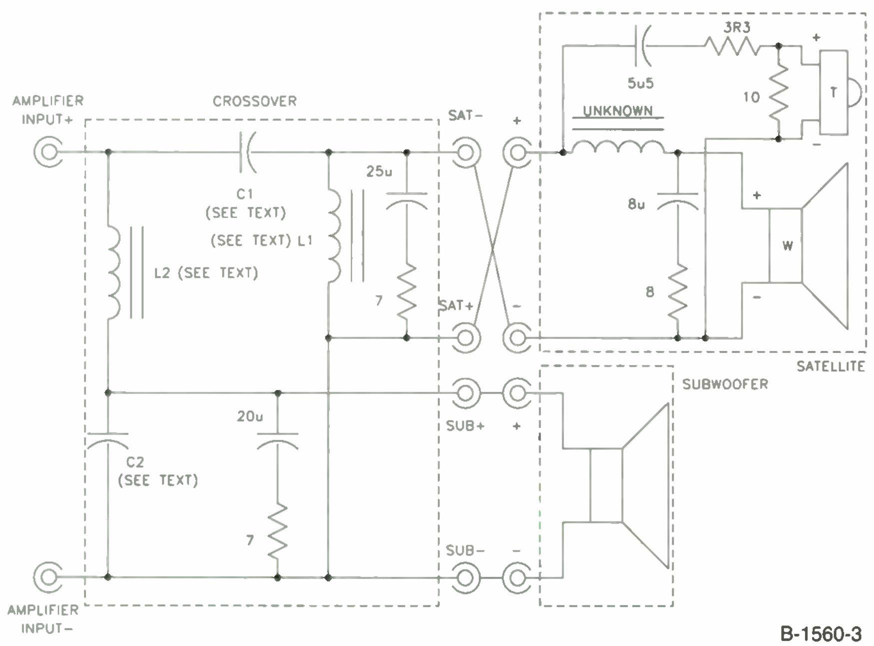


FIGURE 3: Schematic of crossover, NHT SuperOne (satellite) and NHT SW2 (sub).

Buy the Madisound Sledgehammer steel-core inductors, which stay linear out to high currents. Stay away from the available ferrite bobbin-core coils, which I have tested, for they take only a couple of amps. Even though you may have wound power-converter cores, I would be careful about winding your own inductors. I tried all sorts of cores, and most don't work. The core for a CO must have very low permeability (the coils take almost as much wire as an air core), or they will saturate and distort. I think the problem with the present ferrite bobbin cores is that their permeability is too high, and therefore they won't take any current without distorting.

Some CO coils are wound on plastic bobbins with a slug of ferrite in the center hole. When you punch out the slug, the inductance drops only to ½ or ⅓ of the inductance level with the core in place. Also, remember that the core material must be good over the entire audio-frequency range. If you get the big steel cores from Madisound, you can unwind them to any value you wish.

By the way, to arrange two coils so they do not crosstalk, you must situate the lines through the centers of the two cores so they are at right angles in both planes. If the two coils (assuming both are steel-core) are lying side by side, then just rotating one 90° horizontally so they lie on the same plane will not do it. If they are side by side, you must rotate one up through 90° so its core is vertical, and the lines through the core centers are neither parallel nor intersecting.

You will also need a bunch of 20µF Mylar caps and a couple of other smaller values to put in parallel to make the required total capacitance. Because of the low CO frequencies, you are playing with larger cap values than I normally work with. There are cheaper and smaller nonpolar electrolytics, but I have never liked their sound. For woofers, I have had success with 67% nonpolars in parallel with 33% Mylars.

CH 5/27/99:

I was surprised I needed a larger Zobel on the SuperOne than the SW2. I took the SuperOne apart, and here are the CO values:

Tweeter: 5.5µF 100V NP electrolytic and 3.3Ω 10W in series, with 10Ω 10W shunt across the voice coil. Based on your judgment of the sound of NP caps, I may replace the 5.5µF NP with one of the film types and see whether there is any improvement. Size is an issue, since the CO board is mounted on the SuperOne terminal cup, and must fit through the opening in the rear of the box.

Woofer: inductor in series, with 8µF NP 100V and 8Ω 10W Zobel across the

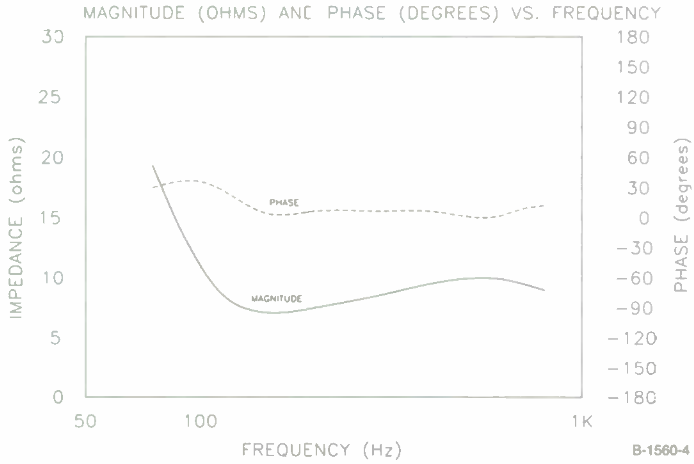


FIGURE 4: Impedance and phase for SuperOne with GRK Zobel network.

voice coil. The inductor core is a laminated iron bar 7.5 × 6mm cross-section and 4cm long, with a nylon bobbin. The coil is wound with AWG-20 magnet wire. I disconnected the woofer-coil positive lead and measured 7.6μH from the binding post to this lead. That seems way too low, given the amount of iron and number of turns. My L-C meter may be fooled by the R-C sneak circuits still connected. If the value is important, I can unsolder one end of the inductor and measure it.

(See *Fig. 3* for a schematic of the proposed complete system, including the internal CO in the SuperOne.)

I wonder if you could band-pass (BP) the 6.5″ woofer in the SuperOne?

GRK 5/27/99:

I was also surprised that the SuperOne required a larger Zobel cap than the SW2, but according to your impedance data, it has a higher voice-coil inductance. The difference in the HP and LP values is because the SuperOne is a bit lower in resistance–if the Zobel works correctly!

On the SuperOne CO, the HP is first-order, followed by a fixed L-pad. The values you give do not sound unreasonable. The LP is also first-order with a Zobel. I suspect the reason the Zobel for the whole box comes out so high is that we are seeing that input inductor. I have never before tried to Zobel a whole box and then add an HP, so things may go astray. I simply took the data you sent and assumed it was a woofer. The 7.6μH coil value is not reasonable. For a first-order at 2.2kHz, the inductor should be about 0.6mH. I don't think we need this value for the present approach.

We need additional components (an HP) on the SuperOnes, since we plan to use it only from the new external CO frequency up. The SW2 will handle the lows up to this new CO frequency, making them into a three-way system.

You do raise the point that you could modify the CO on the woofer of the SuperOne into a BP, and make this woofer into a midrange. I would try the way we are now going for two reasons: first, the most difficult CO network to design is the BP when you lack response data (magnitude and phase) for the driver; second, you could no longer use the SuperOnes as stand-alone speakers.

Hopefully, based on your impedance measurements, we have taken the internal CO into account, but, as I noted previously, I have not tried this approach before. The internal CO may interact, but I hope to avoid this by keeping the external CO frequency very low.

CH 6/15/99:

I received the parts from Madisound last week. I installed the 25μF, 7Ω Zobel on the SuperOne and ran the impedance test you suggested (*Fig. 4*). There is a less than 10° phase angle from 135Hz–750Hz. Impedance varies from 7.1Ω at 150Hz to 9.9Ω at 500Hz.

Next, I installed the 25μF, 7Ω Zobel on the SW2, and ran its impedance test (*Fig. 5*). There is a less than 10° phase angle from 145Hz–750Hz. Impedance varies from 6.4Ω at 150Hz to 7.9Ω at 550Hz.

I will build up the complete crossover for one channel, using the 250Hz values, and test it ASAP. I can borrow a Radio Shack analog sound-level meter. Is this suitable for running response tests? If necessary, I can move the whole test setup out to the back yard to eliminate room effects. I have the CBS test CD-1, and the *Stereophile* CD-3 test CD, which has chromatic scales and warble tones. (Both of these CDs are available from Old Colony Sound Lab, 1-888-924-9465, as P/N CD1 and CDSTPH3, respectively.)

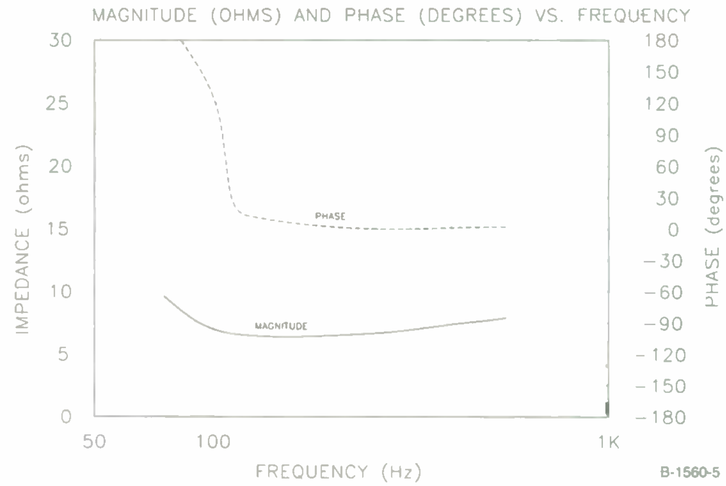


FIGURE 5: Impedance and phase for SW2 with GRK Zobel network.

GRK 6/17/99:

The impedances you report with the Zobels do not sound too bad. It would be nice if the magnitudes stayed more constant, but at least they stay resistive. The design for the SW2s predicted 6.6Ω, so the LP may be about right. However, the design for the SuperOnes predicted about 7Ω, and it looks as though 8Ω may be closer to the mean. This would indicate the HP cap drops to about ⅞ of what I sent you, and the inductor becomes 8/7 larger, so correct the design from 7Ω to 8Ω. I would probably start with the original values, though.

Give the CO networks a try and see how they sound. You can simply breadboard them with clip leads until things are working. Work with two systems, so you can listen in stereo. Getting the system to image properly is important. The COs are sufficiently simple that playing with them both should not be a problem.

The RS SLM (Radio Shack sound level meter) is useful if you know what you are measuring! The key is the test signal. In a room with sinusoids, you will find about a 20dB variation as you move around the room with low frequencies. You might find the level in a corner to be higher than that at a distance of 1m from the box. Assuming the SW2s and SuperOnes are flat over the range we plan to use them (not always a good assumption), then you should only need to get the electrical inputs to each of them to achieve the proper second-order Butterworth shape. This neglects placement problems, and so on. In theory, you should start with the SW2s in proper polarity, and the SuperOnes inverted. From there it is a matter of experimenting.

Another way to find out the right phase and show the "best" sound you will attain is to implement an active 250Hz second-order CO[1] and drive a sub and the SuperOne with the two channels of one amplifier, so you know that the gains and inversions match. If you can't make that sound right, the chances are poor of doing it with a passive CO.

You can test the LP and HP into a resistive load of about 6.6Ω (subs) and 7Ω (SuperOnes). If they don't work correctly into the resistive load, something is wrong with the coils or caps. If they work into the resistors, but not the speakers, then we have some more work to do in compensating the speaker impedances.

Since you are building only one set of speakers, I would work by ear in your listening room. Only if things are a mess would I resort to back-yard testing. Testing a finished system is a fine art and can be really misleading. Joe D'Appolito has published a book on just doing acoustic tests.[2] Unless you wish to learn all the ins and outs of that topic (and invest in the test equipment), I would try to avoid acoustic testing. If you intend to learn about acoustic testing, then buy Joe's book; it is very good.

CH 6/22/99:
I built up a set of your suggested 250Hz crossovers. That was 5.9mH and 68μF on the sub (I used the stock 6.0mH with 70μF). The SuperOne was 6.3mH and 64μF (I used the stock 6.5mH with 64.7μF). The Zobels are the same as in the impedance-versus-frequency test. I made sure the inductors were at right angles to each other, in two different planes, and as far apart as possible.

I previously used the single sub in the corner. With two subs out on the floor and closer to my listening position, the bass is just as pronounced, and I think a little better defined, but with a slight bit less LF extension. There is just a bit of a dip in the room response on the warble tone centered at 125Hz.

GRK 6/22/99:
Listen to the subs alone. They should sound okay, with good definition. Also listen to the SuperOnes alone to see if they are contributing bass. You also should verify that both subs and both SuperOnes sound the same.

When you pull the sub out of the corner, you will probably lose extension and amplitude. It sounds as though the two subs produce about the right level, which is good. It is very hard to predict room effects. Many sources state that a sub in the corner produces the smoothest response. However, as you have already discovered, you need them out by the SuperOnes, since they are now woofers going to 250Hz. If the HP and LP seem to working correctly, the 125Hz problem is probably a room effect. Normally the only cure is placement of the boxes.

CH 6/23/99:
I placed the SuperOnes on top of the SW2s, with their grille boards in the same plane, which produces a very balanced sound. Listening to just the SuperOnes shows they are now virtually devoid of bass. The SW2s by themselves have a smooth bass response, which extends into the low mids.

I tried changing the SuperOne Zobel by the ⅞ ratio you suggested, without any convincing reason not to use the original values. In fact, everything from the SuperOne sounds very nice.

The 125Hz dip seems to be sub-placement related. I can change the dip frequency by changing the spacing between the subs—the 125Hz is with them placed about 7′ apart. If I put the two subs side by side in the middle, the bass dip goes away, but then the directional differences between satellite and sub in the low mids is noticeable. I guess I have some more placement work to do on the two-sub setup. A single sub in the corner is easy to deal with by comparison.

Interestingly, the dip frequency does not seem to change with my listening position. Even standing behind them, I hear pretty much the same bass response. This is much different from my usual single-sub-in-the-corner setup, where listening position greatly affects the bass amplitude versus frequency.

Back to the crossover—I measured volts versus frequency at a constant 0.1A current using a bilateral current-source amplifier I built to make life easier with all this impedance measurement. It uses a cookbook bilateral current source circuit from the National Semi Apps Manual,[3] with an LT1010 high-current buffer inside the overall feedback loop to increase the output-drive capability. It is modeled after my curve-tracer-adapter base-current step generator,[4] but with a 10Ω output resistance for 100mA/V. It can also provide a voltage output by opening the positive-feedback loop.

In addition to the log of frequency intervals, I recorded the frequency where the overall impedance peaked (47Hz), where the volts at the satellite and sub were equal (240Hz), and anything else that looked interesting.

I intend to route a couple of PC boards for the caps and resistors, and to package each channel in an enclosure with input/output terminals.

GRK 6/23/99:
Clearly the COs are working, but I don't think your data gives the true story. The AC bilateral current-source approach is handy for measuring impedance, but it is not the way to measure a CO. The CO design is based on a 0Ω source, not what a current source offers. Also, the speakers are generators, and the Qs of the system depend on the resistance across the voice coil, which should be a minimum. Thus you need the 0Ω source when testing and playing. I think some of the peaking will go away when driven by a low-impedance source.

For the fun of it, I would repeat the test with the CO driven by an amplifier (voltage type), holding about a fixed 1V output. I think the LP and HP responses may look even better.

I would also be interested in what you find in terms of imaging and overall sound with two subs as woofers playing stereo, versus the single sub in the corner. Many articles say there is no imaging at low frequencies, but many other articles say there is!

I have no specific recommendation on the 125Hz dip except to play around with speaker placement. You might improve it if the room is not symmetrical with respect to all the walls. If your listening position will permit it, try the systems at a slight angle to the rear walls and not equidistant from the side walls.

GRK 6/28/99:
I do not agree that your bilateral current source has 10Ω output impedance in current mode. Yes, there is a 10Ω resistor in series with the output of the LT1010CT, but the positive feedback will raise the output impedance to a very high value; i.e., a current source. This means that when testing the CO, the HP and LP can crosstalk with each other, and any internal voltage generated by the drivers will not be damped, but passed to the other network. Will this make a difference?

I suspect the CO measurements will be different with a voltage source, and it is a worthwhile exercise to compare these, since I have never tested the response of a CO driven by a high-impedance source. We know that the performance of a speaker system, basically the woofer, varies according to the source impedance with which you drive it. That is why a low resistance in the LP coil is so important. Back in the 1950s, Altec made mono tube amps with adjustable R_{out} knobs. They had a combination of voltage and current feedback that changed R_{out} while keeping the gain constant. They would even let R_{out} go through zero to negative values, all while staying stable.

I tested one set to negative R_{out}, and the output level went up as you added load! The woofer bass quality definitely changed as you turned that knob. When we design the box for a woofer, the coil resistance is factored into the design.

I have measured the input impedance of a full system both with my impedance meter, which has an output impedance of about 5k (to act like a current source), and with the Audiosuite tester, which is a 10Ω source (a 10Ω sampling resistor in series with a voltage-amplifier output). These agree. Sometimes the peaks near the driver resonance are a little different, but I always wrote this off to the fact that Audiosuite is a "sampled" tester, which does not test continuously with frequency, but as a set of linearly spaced frequencies.

If a network is linear, it would seem that it should be independent of how you measure it. It should measure the same gain no matter how you apply a specific voltage across its input terminals, but with those drivers acting as generators, I'm not sure. You have raised an interesting question, and I'm curious as to what you find. It would answer whether it is valid to test via a current source should you have to do any CO work in the future.

CH 6/29/99:
If you don't mind turning this into a science project, I think I will run the same CO impedance data five ways—the LT1010 constant-current source, and the original method using my 80W power amplifier and series resistors of 0.1Ω, 1Ω, 10Ω and 100Ω. I'll send you the data as

soon as I finish. I'll ignore phase for now and measure only voltage and current magnitudes. I also wish to play some more with speaker placement. I am quite satisfied with the overall sound of the system, but I believe I can detect bass imaging (by unplugging one of the SW2s) down to about the 65Hz "C". Next I'd like to do some controlled tests for bass imaging.

I didn't mean to imply that the current source had a 10Ω output impedance. That is the current-setting resistor, set between the negative and positive feedback points. It determines the output current per input volt.

GRK 6/29/99:
Yes, I think some experimenting would be useful. Along with measuring Z_{in} with the five approaches, I would also record the LP and HP outputs so you have their gain curves. When measuring with a voltage drive, you should watch the level when you get to high frequencies. I've found that even 100W amplifiers don't like high frequencies, and sometimes fry at just a few watts. Even if the amplifier likes it, the tweeter may not. I certainly would stay at 1V or less and turn it down if it sounds too loud.

Let me know what you find. It is an interesting challenge to linear-network theory.

CH 7/1/99:
In the CO test trials, I measured the CO input voltage and current, and the voltages at the SuperOne and SW2 inputs. Connections to the speakers were short—3′ of #12 Monster cable from the amplifier, and less than 1′ from the CO to the speakers. I calculated Z_{co}, LP gain, and HP gain. Are the results in line with what you expected?

I am just about done with the first crossover enclosure. I used a Ten-Tec aluminum box. I mounted L2 on the top-front-left, and L1 on the bottom-mid-right, turned 90° from L2. There is a steel tie bar (3⁄16″ × 1⁄16″) running across the chassis, about ½″ from L2. I measured the inductance change due to this bar, and it was about 0.06mH (1%).

GRK 7/2/99:
I just found an article in *Audio*[5] that tries to explain why you get a suckout at some bass frequency, based on the spacing of your two subs. I don't understand the mechanism yet, but possibly it will give some idea of how to at least partially fix it.

There are two things to worry about when you bring a magnetic material close to a crossover coil. One you have covered—the inductance change. The other is Q. With air-core coils, you can destroy the Q with little change in inductance. With your steel-core coils and the low crossover frequency, I doubt the steel tie bar is a problem.

Certainly the long steel-core coils and most air-core coils used for crossovers have very long return paths. Someone tested the air-core coils, and in a *Speaker Builder* article[6] noted that side-by-side coils had bad crosstalk even with 6″ between them. I loved the ferrite-bobbin-cored coils when the good ones were available (coils still are available, but they sound bad and won't take any current), since they had a much shorter air return path and were less sensitive to other coils and iron objects nearby.

[Note by CH: I asked my friend Kapal Gandikota, an engineer who really understands electromagnetic theory, about the possibility of an inductance drop in L2 due to the steel tie bar. His reply follows:

Inductance is directly proportional to the cross-sectional area of the gap, and inversely proportional to the length of the gap. The effective area of the gap includes the fringing of the flux lines. If the gap length is too great, then the fringing effect is large, and the measured inductance may be much larger than the anticipated or calculated value.

In some cases, especially when the air gap is long and magnetic length (i.e., the length of the core) is very long in comparison to the core's width and breadth, small reductions in air-gap length can lead to a large reduction in fringing effect. This can sometimes result in a small reduction of inductance (contrary to expectation) and vice-versa. However, this (fringing) effect does not come free. The flux density in the gap is lower than in the core, making the stored energy in the gap less for the same core-saturation flux density.]

I was trying to make the point that crossover coils have rather low Qs, which you must take into account when testing them. With low Q, the impedance peak does not occur at the resonant frequency defined by the L-C values, and I was curious as to how you correct for this in your LC meter. I'm not sure this is a problem until the Q becomes very low (single digit), but I raised the issue. When I test coils for effects of nearby objects, I just watch for a change in inductance and don't really note which way it goes. I must watch for this in the future and see what I get with different coil types.

I guess I am not surprised that there is an image at the lower frequencies. They claim if you get the crossover down to 80Hz or below, you can't locate the sub, but I have never been convinced of that.

I haven't had time to study your test data in detail yet.

GRK 7/3/99:

I looked at your tables, and the results are very interesting. Within the limits of test accuracy, it appears to me that the gains of the LP and HP are independent of how you measure them, or at least the differences are slight (*Fig. 6*). The current source measurement even agrees with your previous data.

CH 7/9/99:

I heard a fairly strong double-frequency tone right near the 255Hz sine-wave test frequency during both the 0.1 and 1Ω resistor tests. It was definitely not coming from the drivers, but from behind the speakers. It went away when I unplugged the sub. Fiddling with the enclosure cover reduced it. It was coming from the L2 inductor in the CO, the one in series with the sub. It is mounted on short plated-steel standoffs, so I will look into that issue after the tests are complete.

For these tests, I used the right-side CO—its CO frequency is 255Hz, versus 239Hz for the left one.

GRK 7/10/99:

I don't like those steel standoffs you used for mounting your coils. I prefer an insulator with a small-diameter brass screw to minimize any eddy-current effects. I normally use short pieces cut from wooden dowels.

CH 7/12/99:

I've already changed the L1 and L2 standoffs to brass, and the screws to nonmagnetic stainless steel. I wasn't very smart to use more magnetic metals in a magnetic circuit. I no longer have the overtone.

I will also do more imaging-direction tests with one set of satellite/sub speakers. I can use my signal generator on tone burst at low frequencies and a background 400Hz tone to set a baseline HF direction. I can then plot apparent acoustic LF angle versus frequency, as compared to the fixed HF signal direction. Does this seem reasonable? I don't think I need the crossover. I can feed the 400Hz to the satellite and the LF bursts to the sub.

GRK 7/12/99:

I don't see any problem with your image-direction test approach, but I'm not sure it addresses the "problem." I don't think anyone is claiming that you can't form an image at low frequency. I believe work has been done to show that at low frequency the ears pick direction by arrival time. The point that people seem to make about not requiring two subs is that there is no low-frequency directionality in the recorded (source) material. I don't know whether this is due to the type of mikes used, or even whether it is true, but that seems to be the claim. Your test may produce interesting results, but I don't think it will silence those claiming one sub in the corner is best.

CH 7/13/99:

I ran a directional imaging test on the SuperOne and SW2 with several people. (The hardest part was getting them to understand what I intended them to hear.) The first test was with the sub 60° to the left and the satellite on center. I fed a fixed 400Hz tone to the satellite from my sine-wave generator, and connected my function generator to the sub. I set it for 0.5s sine bursts at 1s intervals. Unfortunately, the "pop" between bursts gave a directional clue, so that even down to 50Hz I sensed it was coming from the left. I switched to a continuous LF tone, but the room effects were distracting.

The best results came by using the frequency-sweep mode, narrowing down the sweep-range boundaries until the sweep's lower frequency was clearly nondirectional, and its upper frequency clearly directional.

I could hear the bass shift from nondirectional to directional between 60 and 65Hz. Some of the others thought it was a bit higher, but everyone heard obvious directional imaging by 75Hz. The results with the sub positioned 60° to the right were the same. One person said the 400Hz on-center tone was distracting, and that the results were better without it.

GRK 7/14/99:

Your directionality test results are interesting. It seems as though the 80Hz number everybody seems to use should be more like 60Hz. I'm not surprised that the edge click with the pulse gave direction, since I believe arrival time is the key to low-frequency imaging. They claim that at high frequency (I think 1k–2k was the dividing point) the directional information is in the relative phase of the two signals.

CH 7/14/99:

In retesting, the HP and LP network gains were essentially the same as the prior data (*Fig. 6*). There were variations in the impedance results of up to 10% (at the minimum), but when I tried to plot them, the graphs weren't visually much different. ***Figure*** 7 is a plot of the test using the power amplifier with 0.1Ω series resistance. The maximum impedances resulted with both the current source and the 100Ω series resistor, and were about 7% higher.

GRK 7/14/99:

I received and looked at your latest data. It appears, within the limits of test accuracy, that the SW2's woofer is linear. Also that its impedance peak is the same no matter what source resistance you use in the measurements. This agrees with the data on my two small woofers, and I think it also agrees with linear network theory.

The data for the complete system still shows some variation, but I can't see any trend in it. There is still a tendency for the high-resistance source to show a slightly greater impedance, but it is not consistent. I would think the variations are measurement limitations, and we have no proof at this time that the system impedance is a function of the source impedance used to measure it.

GRK 8/2/99:

After giving *Fig. 1* some thought, I believe it should look like a closed-box speaker in series with an inductor. Therefore, what I described last time would have inductance added to it and tend to look inductive most of the time, and at the normal zero points. The shape of the magnitude peak does look better.

Figure 1 shows that at high frequency the SuperOne's Z_{in} magnitude drops to about 6.5Ω, whereas *Fig.* 7 shows that the revised system drops to 3Ω. The reason is probably the Zobel across the SuperOne's input. This is doubtless no problem for most amplifiers, but it could become one if you tried to drive a second set of speakers on the same amplifier. I suspect that a small inductor in the Zobel, making it a series L-C-R, would cure this. The inductor would take the Zobel out at perhaps 4k, so for 7Ω it might be about 300µH. I don't think you need to try this unless you would like to, but we should mention the problem.

I don't believe a coil in the 200–300µH range would change the HP response by any major amount, since it is more than a decade away from the CO point. The Zobel network would now have a resonance at about 1.8k, but the Q would be around 0.5, so I don't think this would cause a problem. As with any resonant circuit, readers should be cautioned that if they try this, they should test it carefully. They should also make the inductor as small as required to get

FIGURE 6: HP and LP network gains.

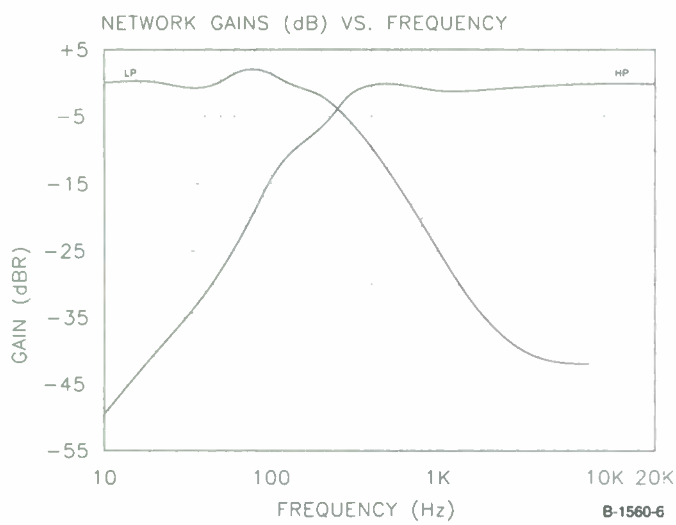


FIGURE 7: Impedance versus frequency—crossover with NHT speakers.

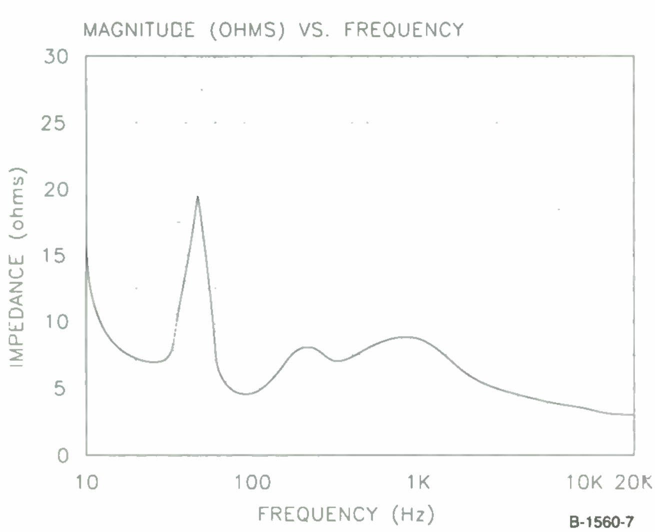
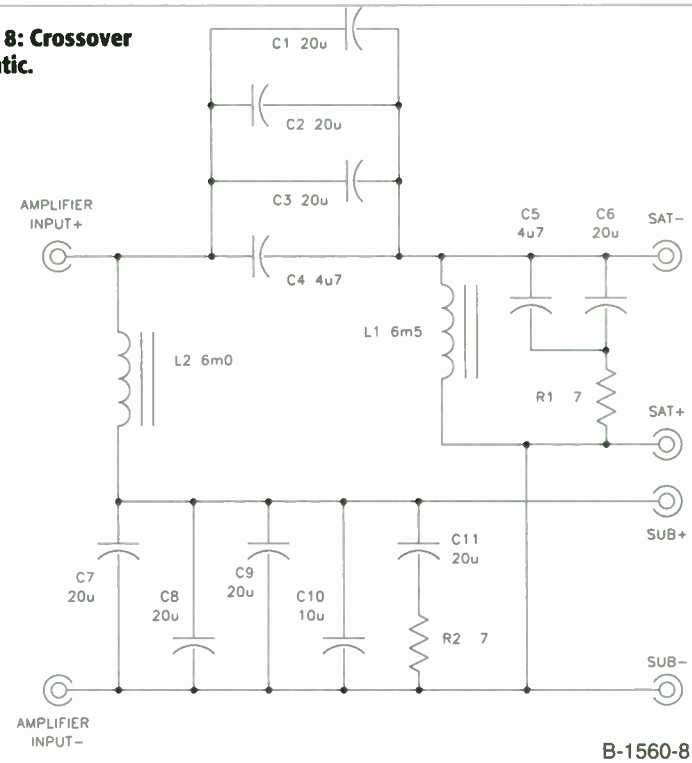


FIGURE 8: Crossover schematic.



the system high-frequency impedance up to where they intend it to be.

CONSTRUCTION

The schematic for the final crossover is shown in *Fig. 8*. The parts cost for both crossovers was $235, and the parts list appears in *Table 2*.

I used a 10.1″W × 4.4″H × 6.5″D Ten-Tec enclosure for each crossover. *Figure 9* shows the chassis layout top and side views. I mounted L2 on the top-front-left, and L1 on the bottom-mid-right, turned 90° from L2. The capacitors and resistors are mounted on one 6″ × 6″ single-sided blank PC board. I routed tracks on the

TABLE 2
CROSSOVER PARTS LIST

DESIGNATION	VALUE, DESCRIPTION	MADISOUND P/N
C1–C3, C6–C9, C11	20µ 150V Carli mylar	M20
C4–C5	4µ7 150V Carli mylar	M4.7
C10	10µ 100V Mitsubishi mylar	10MFD
L1	6m5 Sledgehammer inductor	SL6.5
L2	6m0 Sledgehammer inductor	SL6.0
R1–R2	7Ω 15W 5% non-inductive WW	15R7
		DC ELECTRONICS
J1, J3, J5	Red 5-way gold binding post	7012
J2, J4, J6	Black 5-way gold binding post	7013
	Ten-Tec MW-10 enclosure	MW10
	⅝″ nylon tapped spacers	887
	6″ × 6″ single-side PC board	22-261

PHOTO 2: Crossover connected to speakers.

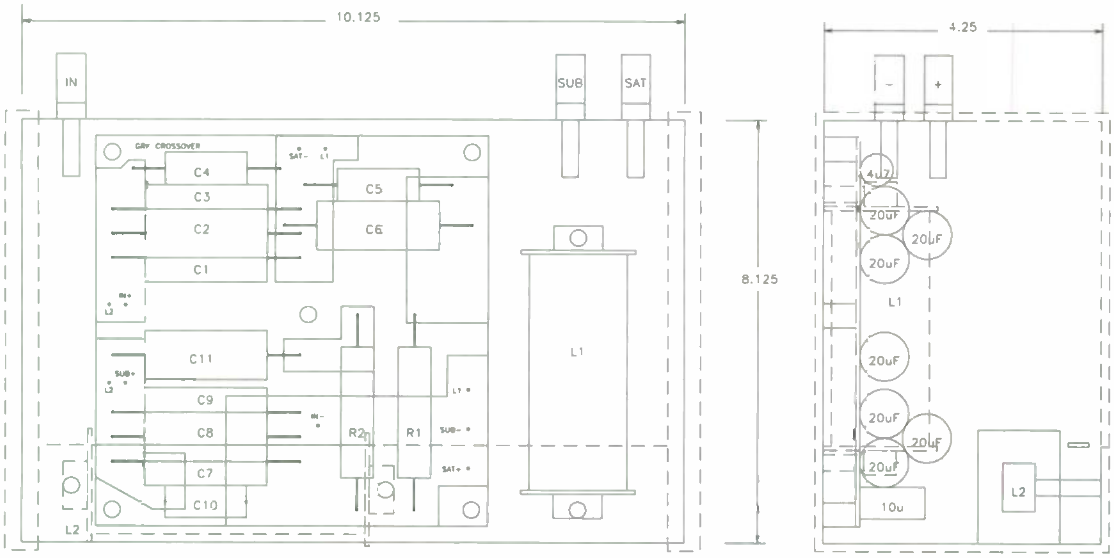


FIGURE 9: Crossover chassis layout.

B-1560-9

copper with my Dremel™ router.

As you may recall from the text, I originally used steel screws and standoffs for the two inductors, which caused a problem, so I changed to plated-brass standoffs and stainless screws. These are specialty items, so you could try nylon spacers instead. Make sure you use nonmagnetic stainless or brass hardware. The 300-series (Austenitic) stainless steels are nonmagnetic. The 400-series (Ferritic and Martensitic) grades are magnetic.

I used tapped nylon spacers to mount the boards to the chassis, with ten short 6-32 nonmagnetic stainless screws, and the holes have sufficient clearance for a flat washer.

I used 16-gauge speaker wire from the six jacks to the board. The inductor leads are connected directly to the board. *Photo 1* shows an internal view of one of the finished crossovers, and *Photo 2* shows the crossover connected to the NHT SuperOne and SW2 speakers.

REFERENCES

1. G.R. Koonce, "Tunable Active Crossover Circuit," *SB* 2/95, p. 20.

2. Joseph D'Appolito, *Testing Loudspeakers*, Audio Amateur Press, 1998.

3. National Semiconductor 1986 Linear Applications Handbook, AN-31, p. 86.

4. C. Hansen, "Semiconductor Tester," *Popular Electronics*, May 1999, pp. 31–41.

5. D. Weinberg, H. Ferstler, "Bass versus Space: Room Acoustics and Good Sound," *Audio*, July-August 1999, pp. 28–33.

6. Mark Sanfilipo, "Inductor Coil Crosstalk," *SB* 7/94, p. 14.

SOURCES

DC Electronics
PO Box 2303
Scottsdale, AZ 85271-3203
800-467-7736
www.dckits.com

Madisound Speaker Components
PO Box 44283
Madison, WI 53744-4283
608-831-3433
info@madisound.com

Now Hear This (NHT)
2950 Lake Emma Rd.
Lake Mary, FL 32746
800-732-6866
www.recoton.com

SB Mailbox

AUTHOR UPDATE

There is an error in your publication (*SB* 7/99) regarding the website which complements my book (*The Life and Works of Alan Dower Blumlein*, p. 55). You published this website as www.jayjaybee.com. This is, in fact, the URL for the website constructors. The correct URL is www.doramusic.com/blumlein.

You may be interested to know that this site currently contains 60% of Alan Blumlein's patents and will be updated so that all 128 patents will be there by the end of November. I also intend to have all of Blumlein's binaural recordings available as downloadable .ra3 or MPEG3 files as well as the binaural film clips (which were the world's first "stereo" films). These should be ready by late December.

Robert Charles Alexander
Watford, England

SOUND-SPEED TESTS

Don Jenkins' experiments reported in *SB* 7/98 ("What Really Happens in a Stuffed Line," p. 32) indicate that the speed of sound through 0.66 lb/ft^3 polyester stuffing is about 68% of its free-air speed at frequencies near 150Hz. Bradbury's analysis predicts a value closer to 50%. Jenkins asks whether any other *SB* readers have made similar tests.

In a forthcoming article on transmission-line loudspeaker system design, I intend to address the behavior of stuffing materials in detail. At this time I will just make four brief statements:

1. As Bradbury noted, measuring sound speed through fibrous materials is difficult and uncertain. My own tests of 0.5 lb polyester in a three-foot pipe confirm a value of about 0.68 in the 150Hz region.
2. Yes, the general effect is real. It is well-documented in technical literature.
3. Yes, effective speed is frequency dependent.
4. Yes, relative sound speed does influence the response of transmission-line systems, but not nearly as much as you might think.

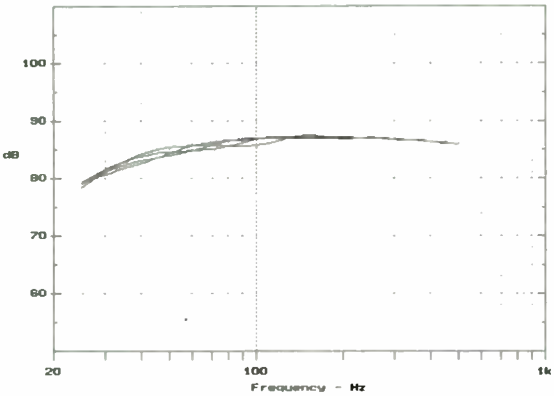


FIGURE 1: Bailey 1972 simulation. Relative sound speed varied from 0.8 to 0.4.

With regard to the last point, remember that Bailey's transmission line is a damped, nonresonant design. It is not a tuned pipe. Pipe output contributes to system output over a range of perhaps two octaves, but its contribution amounts to less than 3dB, and exact phase relationships are not critical.

To illustrate this observation, I set up a computer analog of Bailey's 1972 commercial transmission-line system. This was derived from dimensions and frequency-response tests reported by Geoffrey Letts in his 1975 honors thesis at the University of Sydney School of Electrical Engineering.

The graph in *Fig. 1* overlays four frequency-response curves corresponding to relative sound speed values of 0.8, 0.63, 0.5, and 0.4. Sound speed is held constant; it does not vary with frequency. For this particular design, a value of 0.8 gives the closest match to measured response. It is obvious that the low-frequency cutoff of this well-behaved transmission line is not dictated by effective pipe length even though Bailey may have believed this was the case.

In terms of sound speed, it also turns out that there is no magic stuffing material for transmission lines. You can use fiberglass, polyester, cotton, wool, Acousta-Stuf, and so on. Once packing density has been set for acceptable passband ripple, sound speed pretty much takes care of itself. This is not an opinion. It is based on measured performance.

G. L. Augspurger
Perception Inc., Consultants in Acoustics
Los Angeles, CA

STUFFING EFFECTS

Ketil Parow's article on the All-Fun Horn (*SB* 5/99, p. 10) was really great. It was a perfect balance of "how to build one just like mine" and "here is the mathematical basis for the design so you can build your own." More of this type, please!

Rick Schultz' Rhino article (*SB* 5/99, p. 26) was also dear to my heart, since horns and transmission-line speakers are my two passions. Especially intriguing was Rick's claim that he could design a TL based on Thiele/Small parameters. Unfortunately, the article did not include the formulas Rick used to derive his design. Could you please try to impose upon him to share with all of us the formulas he has derived for calculating length, volume, aperture, and so on, from T/S values?

I am sure it would not compromise even pending-patent applications, since designs for noncommercial use (with formulas) have been published using patent info from Bose and Polk, to name just two. Since others (John Cockroft comes quickly to mind) have not been able to relate TL performance to T/S parameters, I am sure the publication of Rick's formulas could lead to quite a bit of experimentation and debate—to the delight of us all.

Mark Parker
Hancock, NH

Rick Schultz responds:

Thank you for your comments. I also enjoy *SB*'s articles, but the letters section is the best free exchange of ideas I have come across. Our passion for stereo/HE speakers is not an exact science, and this exchange is vital to improving our home systems.

The *SB* 5/99 issue is truly exciting, particularly Marshall Leach's 1979 *JAES* preprint and review of his recent book. Mr. Parow's article about his Lowther horn was a nice contrast to my discussion of compression and oscillation horns. I know the DX3 can work in a quarter-wave system built for sensitivity and have better loading with flatter aperture output.

The letters in *SB* 4/99 contained an exchange between Jakulis, Mattern, and Jenkins. Jenkins' point is that the speed of sound through enclosure air is unaffected by stuffing. Although the air-stuffing mixture has been considered homogeneous in the past, it is not. It is a complex heterogeneous "blob." I suspect the stuffing itself is heterogeneous due to variations in length and diameter. The various materials in the blob have different effects on sound. This complex blob blurs, broadens, and reduces quarter-wave (and harmonic) resonance while damping midbass output from the aperture.

The blob certainly adds mass (inertia) to the acoustic system. It probably produces a drag on compression waves passing the solid material. It may produce less drag on oscillation waves, or it may oscillate itself; I don't know which. I doubt diffraction around the fibers is a consideration due to the small diameter of the material compared to wavelength. Stuffing is a frequency-dependent load.

Stuffing adds inertia to the inertia of the aperture. That means my calculated aperture area (Sa) would increase for a stuffed system. The blob has its own compliance characteristic. My calculated air volume (Ve) would require a smaller air-stuffing blob. Light stuffing (less than 0.5 lb/ft^3 polyester or 0.25 lb/ft^3 fiberglass) has slight effect on efficiency and sensitivity. However, moderate or heavy stuffing will change everything.

In stuffed systems, the driver is interacting with a new inertia and blob compliance. It is no longer in "free air." Stuffing must alter driver Q_{TS}. This is why stuffed enclosures are usually trial-and-error. If only the effects of stuffing could be measured without the interference of an enclosure. See *Fig. 2* for a proposed experiment, with test driver TSP using the standard density of stuffing (0.5 lb/ft^3 polyester or 0.25 lb/ft^3 fiberglass) at increasing lengths from the driver. *Figure 3* is a variation for testing density. Measure TSP as density increases over a practical fixed length (5'?). Perhaps you can test wool, polyester, Acousta-Stuf, and fiberglass. The results will probably depend upon the driver's free-air Q_{TS}, so you should test several different drivers. I don't have the equipment to do this...is anyone inclined to try this?

There is an excellent *JAES* preprint titled "Loudspeakers on Damped Pipes" Parts 1 and 2. Mr. G. L. Augspurger, Perception Inc, has developed a very precise model of the effects of damping material in a transmission line (see "Sound-Speed Tests," previous page). It also models unstuffed quarter-wave systems. While my emphasis is on efficiency, his is on null-free response comparable to an infinite baffle. I suspect the best quarter-wave system lies somewhere between our models. [*These articles on TL design by Mr. Augspurger will appear in upcoming issues.—Ed.*]

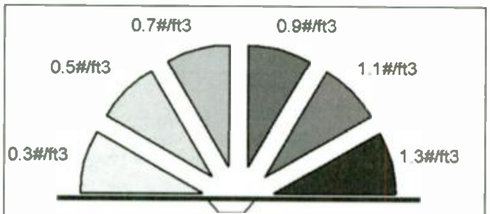


FIGURE 3: A proposed experiment to measure the effects of stuffing density on driver Thiele/Small parameters. As in *Fig. 2*, the stuffing is set as a dome of material. The radius of the dome is the same for each measurement, perhaps 5'. The amount of stuffing within this dome varies to provide the stated densities, which apply to most material, but the values for fiberglass should be halved.

I believe I am on the right path, but the Rhinos have some problems. I know I can improve them. I prefer to hold back the details of the design until it is improved. It would be nice to find a sponsor to help with this process. My main obstacle is the cost of drivers and time/expense of testing. I have had some very bad results on my past and current projects, but I have learned much. With two more projects, I think I will have unstuffed quarter-wave systems nailed down. (I am not sure of the economic value of my model, but with my oldest child off to college, I do not want to jeopardize the value of my intellectual property.)

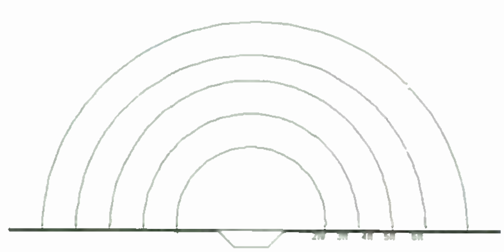


FIGURE 2: A proposed experiment to measure the effects of stuffing material on driver Thiele/Small parameters. Though the diagram is two-dimensional, the stuffing is set as a dome of material. The density of the material should be as uniform as possible.

TL TANGLE

I was eager to read "Transmission Lines: The Real Story" when *SB* 6/99 arrived. However, I found the article so confused and confusing that I thought some clarifications were in order. First, I find astonishing the author's assertion that a nonoptimally stuffed transmission-line speaker is really not a transmission line. Would he assert that an imperfect car is not a car?

His similar statements about ported speakers are also baffling. Whether referred to as bass reflex, ported, or vented, they are all functionally and physically identical; i.e., they all consist of an enclosure and some kind of port that function together as a Helmholtz resonator used to extend the bass response.

Calling all pre-Thiele/Small ported speakers boom boxes seems an unprovable assertion and a disservice to all the designers who toiled away with their rules of thumb and their many experiments. No doubt some designed awful speaker systems, but others succeeded by dint of hard work and a measure of luck to design some good ones. The author's statement seems especially bizarre when he later proposes a nonanalytical, empirically based design technique for transmission lines.

The author also states that vented speakers by necessity must ring and have poor transient response. Both these statements are somewhat misleading. I simulated the following system: speaker f_s = 30Hz, Q_{TS} = 0.37, V_{AS} = 49 ltr, cabinet volume = 27 ltr, cabinet tuning = 30Hz, and cabinet Q = 10. The impulse response looked so perfect that it wasn't even worth reproducing. *Figure 4* shows the step response of this system, while *Fig. 5* shows the same system with the port blocked off, i.e., a sealed system of the same volume.

The ported system has a bit more overshoot and just a hint of ringing. The alignment of the sealed system is very close to critically damped, and if its transient response can be characterized as very good, it would be difficult to characterize the transient response of the ported system as bad. It would not be clear to me that the difference between the two would be audible without a double-blind test.

The author also makes a cryptic comment that vented designs have a Q_C of one. Because he doesn't define the variable, we don't really know to what he refers. It can't be the cabinet Q, because that is usually made as high as possible. As both Thiele and Small showed, the

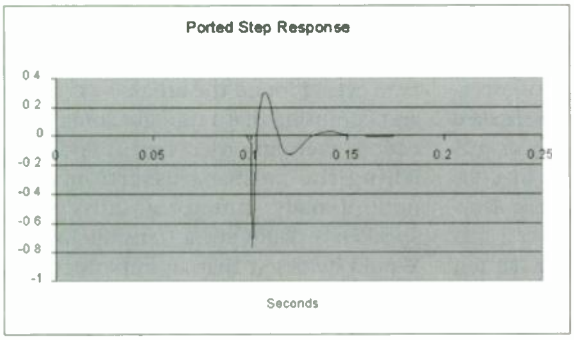


FIGURE 4: Ported step response.

cabinet Q can be infinite without causing response problems. Q_C would not seem to be the system Q, because that is variable, depending upon the box volume and tuning as selected for a specific driver.

The author errs in his descriptions of musical instruments. He seems to imply that the clarinet's fundamental resonance is at four times the bore length because of the instrument's bell, and that the flute's is at two times the bore length because of its lack of a bell. These assertions, along with Fig. 5 (assuming it attempts to show resonance in a pipe with one closed end), are false.

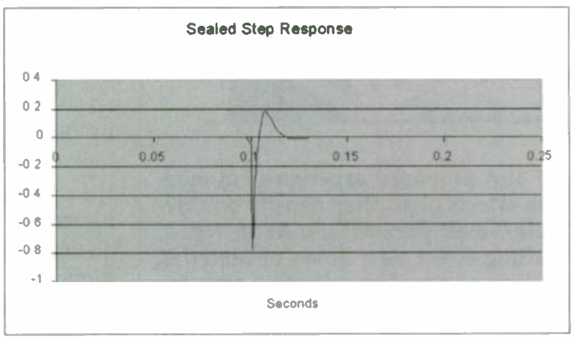


FIGURE 5: Sealed step response.

The author also incorrectly asserts that organ pipes have resonances at all integral fractions rather than just at odd integral fractions. Some do and some don't; it depends on the design of the pipe.

First, let's clarify how a closed-end pipe resonates. At resonance, a standing wave will develop in the pipe. Sound is a compression wave, and at some point or points along the standing wave there will be a node (or nodes). This is a place where pressure change is at a maximum and velocity change is at a minimum.

Between the nodes will be antinodes, where pressure change is at a minimum and velocity change is at a maximum. At resonance, there will be a node at the closed end of the pipe: by definition, there can be no velocity at the blocked end. There will be an antinode at the open end. The longest wavelength that can satisfy these boundary conditions is four times the pipe's length. The next shortest wavelength that satisfies these criteria is three-fourths the pipe's length, and the next are at ⅘, ⁴⁄₇, and so on. Thus we see the real reason for the odd harmonic series.

A pipe open at both ends has a fundamental resonance with a node at its middle and two antinodes at each opening, which is a wavelength twice the bore length. If a flute has a fundamental resonance at twice the bore length as the author asserts, it is most likely because the large hole that you blow across to sound a note makes the flute look like a pipe with two open ends.

The author also states that a TL has a different response from an organ pipe because it is excited at its closed end, whereas organ pipes are excited at their open ends. This statement is also misleading. A simple experiment will prove my point.

Take a short length of pipe—a piece of ½″ or ¾″ copper water pipe about a foot long works very nicely for this experiment. Block one end off and blow across the open end to make it sound. You will hear a distinct note whose wavelength is four times the pipe's length. Hence we see that even when excited at its open end, the pipe will still resonate at the same frequencies. It has no choice—its resonance is determined by its geometry, not the spot at which it is excited.

The author also confuses things by using the term harmonic in an unconventional way. He calls the first harmonic that frequency that is twice the fundamental, but in virtually all cases, the first harmonic is considered to be the fundamental frequency. I could cite so many examples of this that it would become boring, but let me mention one. Symmetrical waveforms are always considered to have odd harmonics, and their harmonics are at 1, 3, 5,... times the fundamental frequency. Thus, the peaks shown in the response in Fig. 7 would be considered the 1st, 3rd, 5th,... harmonics in common parlance.

The author also mentions the TL has a line response with a −18dB slope. This detail is confusing, because he doesn't specify over what frequency range this response variation occurs. Inspection of Figs. 7 and 8 clearly shows that the response is 12dB/octave, which makes the line a second-order system.

While I have many criticisms of the author's presentation, I do not wish to leave the impression that I intend a wholesale condemnation of his work. I suspect that readers will learn many valuable tips about designing TLs as this series continues.

Roy Mallory
Bedford, MA

A. Monk responds:

Thank you for a stimulating letter. The TL has historically provoked strong opinions, and I hope your observations will result in equally strong analysis by other readers. If your observation that the article is confusing is upheld, then I have failed in my attempt to define the TL's characteristics. I hope your opinion is somewhat premature, since only the first part has been published and you will temper your judgment when you have a chance to read the complete presentation.

Perhaps my attempt to answer the objections you raise will permit me to clarify some points that were abridged in the first part and help other readers with what appears to be confusing.

I needed an accurate, yet nontechnical, definition of the TL to differentiate it from similar design methodologies which were grouped under the ¼ lambda category. The article is about defining it, but initially I chose to go with the statement that the TL is what Bailey did when he built and characterized the first TL line. The critical design issue is the concept optimization of the TL's response, i.e., a cusp-type response and minimization of the phase-angle response at the shifted F_R.

Had I initially used this definition, which gives a very precise quantification of an optimum TL line length versus fiber density/type response, it would have amounted to gobbledygook. This cannot be understood without an analysis of the fiber characteristics in the line. From this concept it follows that any line not optimized cannot be considered to have Bailey's defined TL response.

Perhaps you have an alternate definition; if so, I would be interested in hearing it. By all means publish your analysis; the TL literature will benefit. If your objection is based on logic or a theoretical model, then we are in disagreement, since in my criteria a concept must be supported and conform to empirical data. Your statement that "The author's statement seems especially bizarre when he later proposes a nonanalytical, empirically based design technique for transmission lines" is a misunderstanding of the proposed model as based on experimental data.

However, if your preference is for a simulation mathematical model, I have included a reference to J. Backman's paper which includes such a model. It is well worth the time to study it; however, you need some familiarity with mathematics as well as computer modeling to understand it; thus it is beyond the scope of the majority of amateur speaker builders. In my view the problem with this model is that the mathematical model has not been verified and corrected by experimental data, as the author points out at the end of the paper.

Your primary objection seems to be tied to the vented box design and performance characteristics and the "boom box" nomenclature used for the response of the bass reflex. This was Bailey's characterization and used for the bass reflex before the Thiele and Small analysis for the woofer was defined and from whose work the vented box evolved. Thus the lumping of bass reflex, ported, and vented as Helmholtz variants is misleading both historically and analytically. As to the poor transient response of the vented box, this is somewhat controversial, because there is no accepted definition in the amateur community of transient response and how we perceive it. I attempt to illustrate this question in the second part, but I suspect that you will not be very happy with the responses shown.

Your responses for a step response of sealed and ported designs is very interesting and, in my opinion, contrary to your statement that it very well illustrates the poor transient response of the ported box. After the initial positive peak, the data for the ported box shows a very large negative peak and a smaller positive peak before settling out. This is the classical example of a ringing response, i.e., poor transient response. The given plot for a sealed response has no extended energy storage; i.e., it is a critically damped system.

I used the analogy of the woodwind instruments to illustrate the concept of resonance as related to the geometry of the pipe. I did not intend to draw a rigorous correlation to the TL. Each must be analyzed by its characteristics as defined by experimental data.

I have taken the data on the clarinet and the flute from the chapter of *The Physics of Wood Winds* from "The Physics of Music," a *Scientific American* preprint. If I have misquoted, the fault is mine, but it has very little bearing on the TL analysis.

I'm guilty of idiosyncratic usage of the term harmonics, but I would not change it, since the mathematics of calculating the fundamental and relating to the nulls and peaks was, in my view, consistent and useful, and I could not think of a more appropriate term. To define the harmonic response of the TL line by nulls versus peaks is even more confusing, since the system response peak will not correspond to a line-related resonance peak.

The Helmholtz absorption nomenclature is not useful. While my usage is a deviation from the academic definition, you should be able to make the mental translation if you so choose. Sorry for the confusion, but the subject matter is not epistemology or adherence to a particular academic jargon.

You bring up an interesting example in the resonance of the pipe blocked at one end. Your ability to analyze its resonance by aural spectrum analysis is much finer than mine, since I must rely on instrumentation, but it is an experiment I have done and describe in the third part of the TL article, where I examine the question of whether the TL can be classified as a Helmholtz. My opinion is no. The data signatures are distinct, and, again, each type must be defined by the experimental data correlation. Generalizations are unproductive.

Thank you for a thoughtful and detailed letter. I suspect that our disagreements are superficial, since we both seem to value verifiable data over subjective assertions.

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Speaker Builder 1990 to 1998, $120 postpaid. New Peerless 4½″ poly midrange TO105 $50/pr.; Philips 2″ dome mid AD02110 $30/pr. new; Audax 34mm dome mid HD13D34H, $30/pr. new; Seas 76mm dome mid H304 $50/pr. new; eight 4″ woofers, 40-1022, like new, $10/each. Manuel (505) 898-5967

Boulder Ultimate preamp ($4500), $1800. Mach 1 Acoustics Reference DAC ($2400), $540. Two Mach 1 Acoustics Isobarik subwoofers, each sub contains two Eton 11-580 11″ woofers, $270 plus shipping each. Marc (603) 860-3085

Denon amp, 2000W/ch, $400. Harman-Kardon PT2500 tuner/preamp, $200. Both for $550, both in mint condition. Nick (559) 583-6511.

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Bruel & Kjaer 2203 sound level meter with 4145 microphone, in working order, $400 USD. Carl Lavertu (514)-396-8574.

Audio Concepts "GEM" monitor-type speakers in lighter oak finish. Near mint, $600, negotiable. David (801) 731-2123 AM, or E-mail sndrsut@aol.com.

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Driver Report

CSX 145 H FROM PEERLESS

By Vance Dickason

This Driver Report focuses on a driver from the Peerless (Denmark) CSX series, the 5″ Model CSX 145 H.

PHOTO 1: The Peerless CSX 145H.

Features: CSX-series products have a stamped surface-mount frame (one of the nicest stamped frames in the industry), composite sandwich paper cone, inverted dustcaps, aluminum shorting ring for lower distortion, flat linear compliance spider, and rubber surround. The CSX 145 H is a 5″ woofer with a 26mm diameter voice coil suitable for midbass/midrange applications and as a small woofer in a two-way speaker.

Measurements: With the LMS analyzer, I produced the free-air impedance curve shown in *Fig. 1*. I then transported this data to LEAP software to produce the T/S parameters given in *Table 1*. After verifying that the sample data correlated well to the factory specification sheet, I used this information to create a sealed-box computer simulation in a 0.08ft^3 box (50% fiberglass fill), and a 0.21ft^3 vented box tuned to 53Hz. *Figure 2* shows the response of both enclosure types simulated at 2.83V. The sealed enclosure yielded an f_3 of 102Hz at a –3dB phase angle of 92° (box Q_{TC} of about 0.7), while the vented box produced an f_3 of 51.3Hz. *Figure 3* depicts the associated group-delay curves for both simulation designs.

I increased the input voltage of the simulation to a level which would pro-

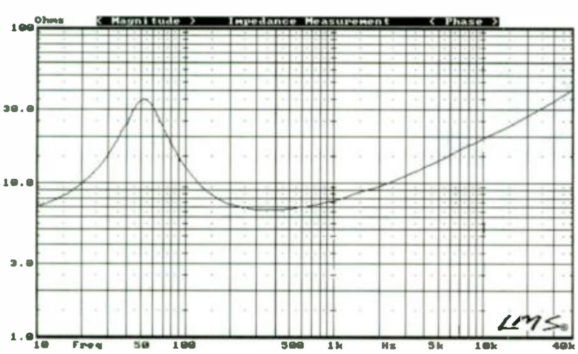

FIGURE 1: Peerless CSX 145 H impedance plot.

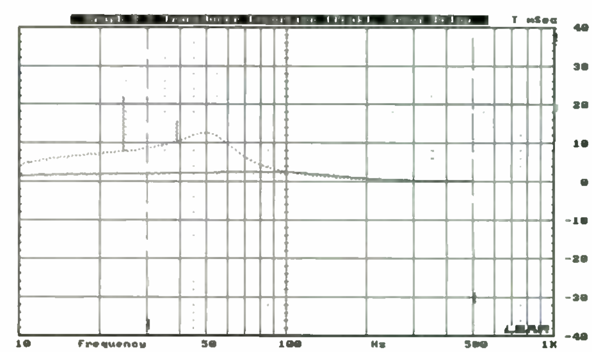

FIGURE 3: Group-delay curves for *Fig. 2* (solid = sealed, dot = vented).

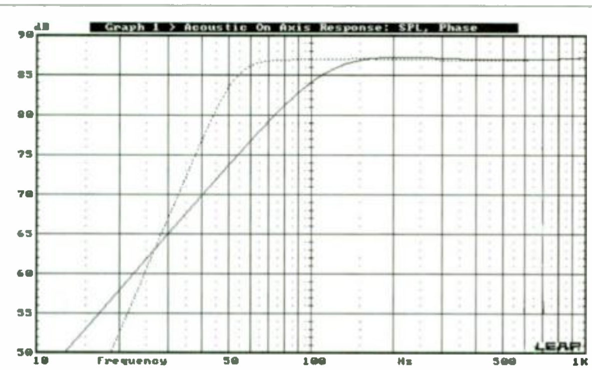

FIGURE 2: CSX 145 H 2.83V box simulation (solid = sealed, dot = vented).

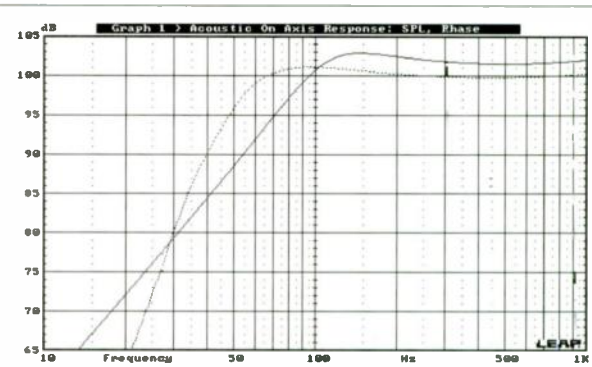

FIGURE 4: CSX 145 H box simulation at 21V (sealed) and 15V (vented) (solid = sealed, dot = vented).

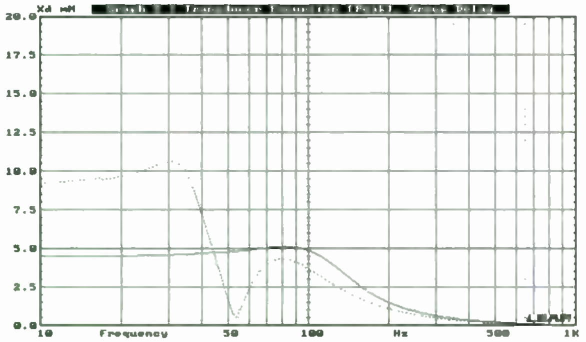

FIGURE 5: Cone-excursion curve for *Fig. 4* (solid = sealed, dot = vented).

FIGURE 7: CSX 145 H SPL comparison for two samples.

FIGURE 6: CSX 145 H on- and off-axis frequency response (solid = 0°, dot = 15°, dash = 30°, dash/dot = 45°).

TABLE 1
PEERLESS CSX 145 H PARAMETERS

	SAMPLE A	SAMPLE B	FACTORY
f_S	52.6Hz	52.1Hz	48Hz
R_{EVC}	5.92	5.92	6.1
Q_{MS}	1.99	2.04	1.78
Q_{ES}	0.41	0.40	0.42
Q_{TS}	0.34	0.34	0.34
V_{AS}	8.9 ltr	9.1 ltr	12.5 ltr
Sens.	87.8dB	87.9dB	87.5dB
X_{MAX}	4.5mm	4.5mm	4.5mm

duce a cone-excursion maximum equal to X_{MAX} + 15%, defining the linear operating envelope for the product. SPL, as shown in *Fig. 4*, increased to about 103dB for the sealed box with a 21V input, and 101dB at 15V for the vented box. The vented box was limited due to the excursion below 40Hz exceeding the X_{MAX} + 15% criteria (*Fig. 5*). The CSX 145 has a long voice coil for a 5″ driver and produces somewhat higher output levels than many of the 5″ drivers previously featured in this column.

I mounted the driver in an enclosure with an 11″ × 6½″ baffle to produce the on- and off-axis curves shown in *Fig. 6*. The woofer gives a smooth response to 5kHz, with a 6dB peak in the response at 9kHz. Off-axis response at 30° indicates that a crossover as high as 3.5kHz would work well for this driver and yield a very flat response. *Figure 7* compares the SPL of both samples, showing a good match out to 8kHz.

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Showcase

THE NARDEUX CENTER CHANNEL

By John Badalamenti

To complement the Nardeux N-300 loudspeakers ("Tools, Tips & Techniques," *SB* 3/99, p.58) for home-theater Dolby surround sound, you need the Nardeux center loudspeaker for a perfect match. I started with the idea of a common design, but soon realized that I needed the intervention of Joe D'Appolito, who suggested a design that was the ideal way to go. With a 3-way system, I could pack four drivers into the specified dimensions.

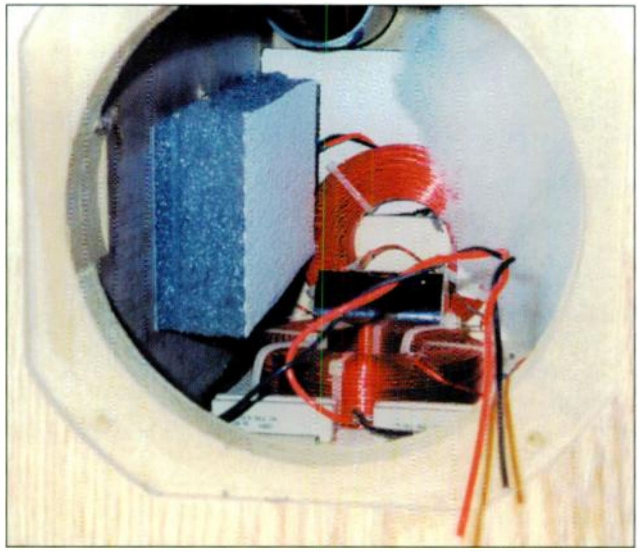

PHOTO 2: Low-pass crossover mounted on left side.

LOADING THE CABINET

Photo 1 shows the crossover assemblies. The low-pass crossover is mounted onto one board (*Photo 2*), while the mid- and high-pass are mounted onto another board (*Photo 3*). Each crossover board is mounted inside each side of the cabinet for weight distribution. The crossover specs are show in *Fig. 1*.

The internal wiring is 16AWG and 18AWG D.H. Labs silver-plated pure copper wiring. Note the "Blackhole" mounted in place inside the cabinet and the Dacron lightly placed all around internally (*Photos 2* and *3*).

The cabinet is loaded with FOCAL shielded drivers. The midbass drivers are 7K4411Bs (two units). The midrange driver is the 5K413S, and the tweeter is the T90TdxB. The loaded cabinet weighs in at a hefty 65 lbs. The overall measured impedance at the banana input terminal is 9.8Ω. The finished cabinet (front and rear) is shown in *Photos 4-6*. The cabinet specs are shown in *Figs. 2* and *3*.

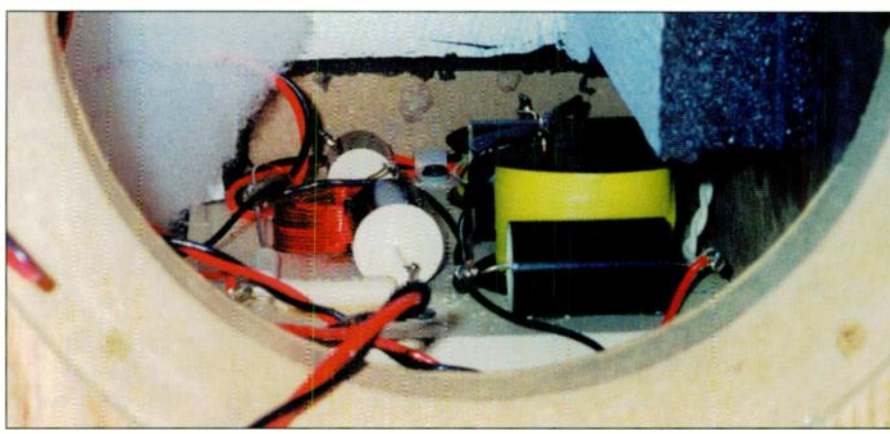

PHOTO 3: Mid- and high-pass crossover network mounted on right side.

PHOTO 1: Low-pass crossover network (left), mid and high crossover network (right).

According to Joe's test results, the frequency response is flat within ±1dB—making it the perfect center-channel speaker for the Nardeux N-300s. The cabinets' response is as low as 45Hz.

The high-quality cabinet construction is by Zalytron Industries. Note the beautiful red-oak finish to match the front-channel speakers. I purchased all crossover parts and drivers from Zalytron.

I believe this is an excellent project

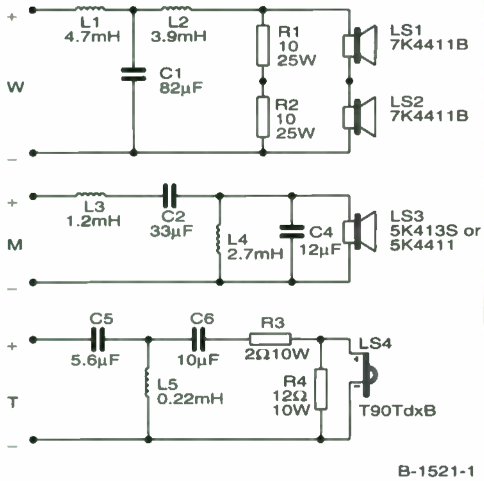

FIGURE 1: Crossover design.

PHOTO 4: Finished Nardeux center channel without grille.

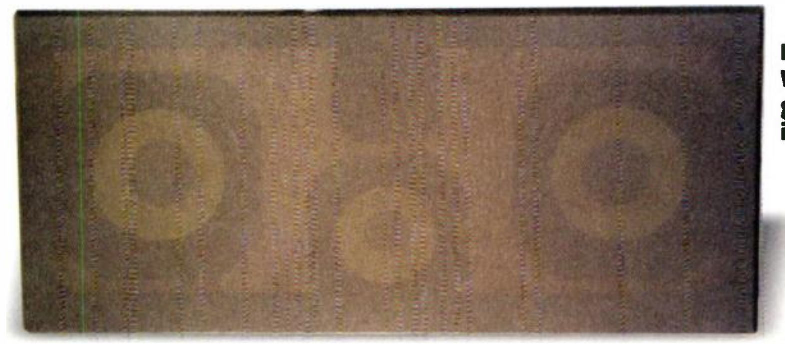

PHOTO 5: With grille installed.

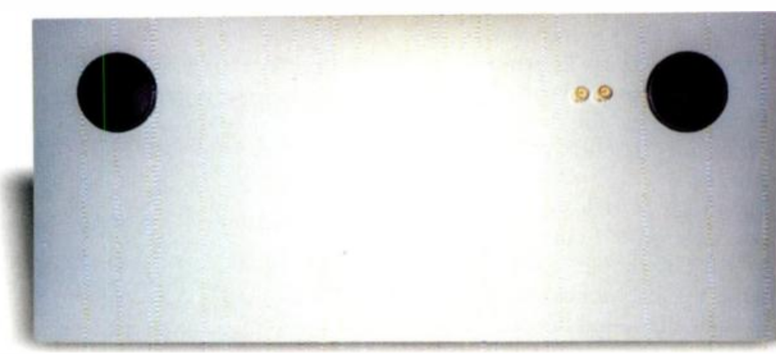

PHOTO 6: Rear view of cabinet. Note vent ports and Axon terminal posts.

for anyone seeking state-of-the-art performance with a reasonable, realistic approach. The assembly was not very difficult; with some practice and some basic hand tools, any novice can enjoy the rewards of amateur speaker building. So impress yourself and many others for many years to come by building something you can be proud of.

If you have questions on this subject my e-mail address is j.badalamenti@worldnet.att.net.

This Nardeux Center Channel is dedicated to the NARDEUX STEREOMATIC 302S from Loches France and my brother, Andrew, who inspired me from the beginning with stereo equipment and a whole lot more.

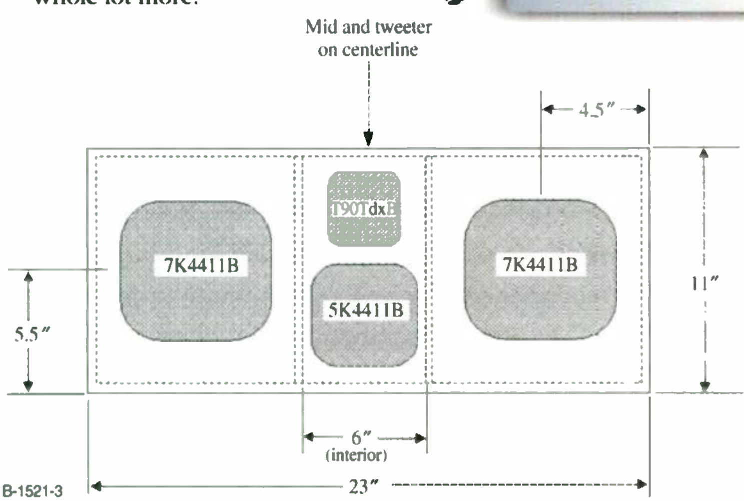

FIGURE 2: Cabinet specs (front view).

Badalamenti center channel top view.
Line rear, sides, top or bottom (not both) with Black Hole.
Line rear of mid/tweeter sub-enclosure with Black Hole.
Fill sub-enclosure with lightly compressed high loft Dacron.
All material 3/4″ MDF except front baffle
Front baffle 1″ MDF

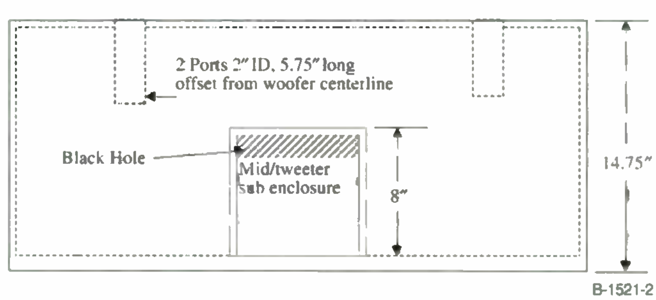

FIGURE 3: Cabinet specs (top view).

NEW

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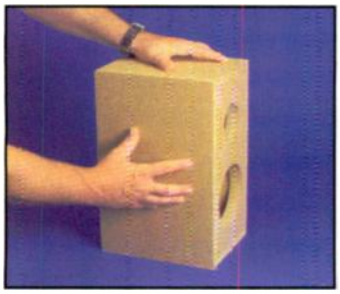
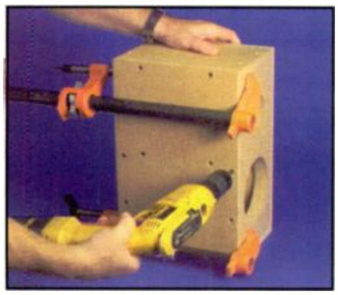
"I Built It Myself!"

By using technology previously only available to high end speaker manufacturers, we are able to offer audiophile speaker cabinets at reasonable prices. Now you can build an audiophile quality speaker system without expensive woodworking tools or hours of labor. We have developed a series of cabinets designed by audiophiles and computer manufactured. The cabinets are made of 1" MDF (medium density fiberboard) to reduce cabinet resonance and coloration. They are precision milled with a CNC router to incredible exacting tolerances. Dado and slot joinery are used throughout to provide incredible strength. Internal braces reduce unwanted cabinet resonances. All you have to do is assemble, glue, and finish to your liking.

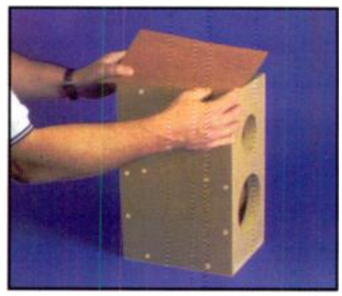
Blank Baffles: The cabinets come with pre-cut baffles for standard drivers. Blank baffles are available for you to rout yourself for flush mounting or custom drivers.



Step 1
Assemble cabinet on your workbench and pre-drill and countersink screw holes. Modify driver holes if necessary at this time.



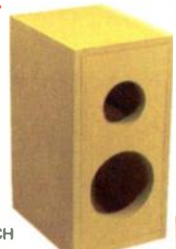
Step 2
Brush Polyurethane glue (available at most home centers) on both sides of all joints. Assemble cabinet (clamp if possible) and install screws.



Step 3
Fill screw holes with putty let dry and sand. Apply your veneer or laminate.

Step 4
Varnish or paint cabinet to your liking. Install crossover, rear terminals and drivers.

.25 Cu. Ft. Bookshelf Cabinet



Reminiscent of the classic British mini-monitor, this cabinet is well suited as a compact 2-way audiophile system or home theatre setup. .25 cu. ft. internal volume. Includes 1" Dado brace. 1" Front Baffle is pre- cut to accept most 5 - 6 in. mid/woofs and tweeters. 100% 1" MDF Construction. Internal dimensions: 5" W x 12-3/4" H x 7" D ◆External: 7" W x 13-3/4" H x 9" D ◆Woofer hole: 4-5/8" ◆Tweeter hole: 3" ◆Net weight: 12 lbs.

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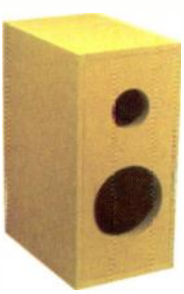
#300-703 (Blank Baffle, 1" MDF) .... **$3.95** EACH

.55 Cu. Ft. Cabinet



This is the perfect cabinet for any single 6-1/2" woofer and 5-1/4" single or dual woofer arrangement. .55 cu. ft. internal volume including 1" MDF Dado brace. 1" front baffle pre-cut to accept 6-1/2" woofer and tweeter. Internal dimensions: 6-1/4" W x 15-1/2" H x 10" D ◆External dimensions: 8-1/4" W x 17-1/2" H x 12" D ◆Woofer hole: 5-5/8" ◆Tweeter hole: 3" ◆Net weight: 20 lbs.

#300-708 .... $46.80 EACH
#300-707 (Blank Baffle, 1" MDF) .... **$5.95** EACH

Note: All of our MDF cabinets are shipped "knocked down" ready to assemble. They include detailed assembly instructions and finishing recommendations.

.75 Cu. Ft. Dual Woofer Cabinet

Build the popular D'Appolito style dual woofer speaker system. The front baffle is pre-cut to accept two 5-6" mid/woofs and one tweeter. Popular tall European style cabinet with two internal braces to help eliminate unwanted panel resonance. .75 cu. ft. internal volume including 1" braces. Internal dimensions: 6-1/4" W x 21-3/4" H x 10" D ◆External dimensions: 8-1/4" W x 23-3/4" H x 12" D ◆Woofer holes: 4-5/8" ◆Tweeter hole: 3" ◆Net weight: 26 lbs.

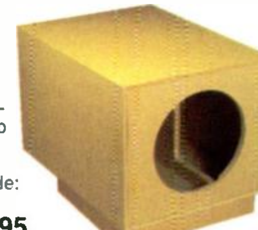
#300-714 .... $59.80 EACH
#300-713 (Blank Baffle, 1" MDF) .... **$7.95** EACH

1 Cu. Ft. Esoteric Speaker Cabinet

When completed, this cabinet will rival audiophile systems costing thousands! Designed to accept dual 6-1/2" mid/woofs and a center tweeter in a D'Appolito configuration, or buy the optional blank front baffle to design your own system. Two internal braces to help eliminate unwanted cabinet resonance. Internal dimensions: 6-1/4" W x 21-3/4" H x 13-3/4" D ◆External dimensions: 8-1/4" W x 23-3/4" H x 15-3/4" D ◆Woofer holes: 5-5/8" ◆Tweeter holes: 3' ◆Net weight: 31 lbs.

#300-718 .... $69.50 EACH
#300-713 (Blank Baffle, 1" MDF) .... **$7.95** EACH

3 Cu. Ft. Subwoofer Cabinet



Finally, a high quality, high performance MDF subwoofer cabinet at an affordable price. This cabinet utilizes 3/4" MDF (medium density fiberboard) not particle board. MDF has far superior sound deadening characteristics than particle board, plywood, or OSB. 3 cu. ft. cabinet is perfect for 10"-15" subwoofers. Internal Dado brace to reduce cabinet resonance. Inside dimensions: 16-1/2" W x 14-1/4" H x 22-1/2" D ◆Outside dimensions: 18" W x 15-3/4" H x 24" D ◆Woofer hole: 11-1/8" ◆Net weight: 43 lbs.

#300-728 .... $89.95 EACH

TO ORDER CALL TOLL FREE 1-800-338-0531

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TITANIC 1200

Titanic 1200 Subwoofer

The Dayton Loudspeaker Company assembled a team of the best speaker designers in the industry with one thought in mind ... to build the finest A/V subwoofer on the market! These engineers were chosen because of their expertise and pioneering designs of long excursion, high output subwoofers. A dozen prototypes were built and rigorously tested before we were satisfied with the final design parameters. With specifications like 16Hz Fs, 14mm Xmax (linear), 90dB SPL (2.83V) and 350W power handling, we're sure you will agree that this subwoofer is truly, Titanic!

Only the finest components are utilized: a heavy cast aluminum basket, a talc filled polypropylene cone, SantopreneR rubber surround, an ultra high power voice coil assembly with a high temperature resistant ApicalR former, and quality, high tech adhesives to hold it all together. All of these components cost more, but are worth it!

ONLY $149.80 ASK FOR #295-400

Lets talk Performance!

The 12" Titanic was designed primarily as a sealed box woofer. One possible design is a 3 cubic foot sealed box with 50% fill. This design produces deep, accurate bass to 25Hz. You will be overwhelmed by the incredibly realistic low frequency that this driver will produce. Go ahead, put the T/S parameters into your favorite box design program and see for yourself what the Titanic 1200 can do!

Try it for 30 days ... If you don't feel this is the best A/V subwoofer you've ever heard, we'll refund your money! The Titanic 1200 is covered by our 5 year warranty.

Thiel-Small Parameters

◆Power handling: 350 watts RMS, 450 watts max. ◆Voice coil diameter: 2" ◆Voice coil inductance: 1.96 mH ◆Nominal impedance: 4 ohms ◆DC resistance: 3.66 ohms ◆Frequency response: 16-400 Hz ◆Magnet weight: 84 oz. ◆Fs: 16 Hz ◆SPL:90dB 2.83V/1m ◆Vas: 9.894 cu. ft. ◆Qms: 8.22 ◆Qes: .42 ◆Qts: .407 ◆Xmax: 14.2 mm ◆Net weight: 14.6 lbs. ◆Dimensions: A: 12-1/8", B: 11-1/8", C: 6-9/16", D: 6", E: 2-3/4"

TITANIC Subwoofer Cabinet

ONLY $119.80 ASK FOR #300-729

This cabinet was designed to make the Titanic woofer "Sing." 3 cu. ft. cabinet with a substantial 1" MDF braces to couple all cabinet sides and help eliminate unwanted panel resonance. One inch thick MDF (medium density fiberboard) is acoustically superior to particle board, plywood, and OSB. Dado and Slot joinery coupled with screws and polyurethane glue (not included) make this cabinet virtually "Dead." The MDF exterior is easily painted or veneered to your liking. (Finishing Instructions Included Free). Inside dimensions: 17-3/4" cubed ◆Exterior dimensions: 19-3/4" cubed ◆Woofer hole: 11-1/8" ◆Net weight: 50 lbs. ◆Shipping knocked down.

250 Watt Class D Subwoofer Amplifier

- Perfect mate for Titanic Subwoofer
- High efficiency amplifier
- Made in the U.S.A.
- Advanced Class-D Topology
- Reliable/stable operation
- Low distortion
- High pass filter for satellite speakers

ONLY $249.80 ASK FOR #300-795

All subwoofer amps are not created equal!

Today you have many choices when it comes to an amplifier for your next subwoofer project. In our quest, we have seen them all and tested many. The heart of any subwoofer system is the amplifier; do you really want to trust your "Audiophile grade" subwoofer system to an amplifier made in China? This American made, **TRUE 250 WATT** amp produces earth shaking bass, especially when mated with our 12" Titanic subwoofer. Class D amps are nearly 95% efficient creating less heat making them more reliable and stable while producing less distortion. This amp features both high and low level inputs, phase reversal switch, level control, auto on/off (activated by input signal), and a second order (12dB) electronic low pass filter that is continuously variable from 40 to 160 Hz. It also features a 6dB high pass filter (90Hz @ 8ohm) output for satellite speakers. The amp has overload and fuse protection for years of reliable use. The highly efficient amplifier sums the right and left stereo inputs to a mono output, so that only one amp is required per system. Rated power output: 250 watts into 4 ohms @ 0.1% THD. Signal to noise ratio: 100dB (A-weighted). Dimensions: 10" W x 10" H x 2-1/2" D. Net weight: 4 lbs.

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