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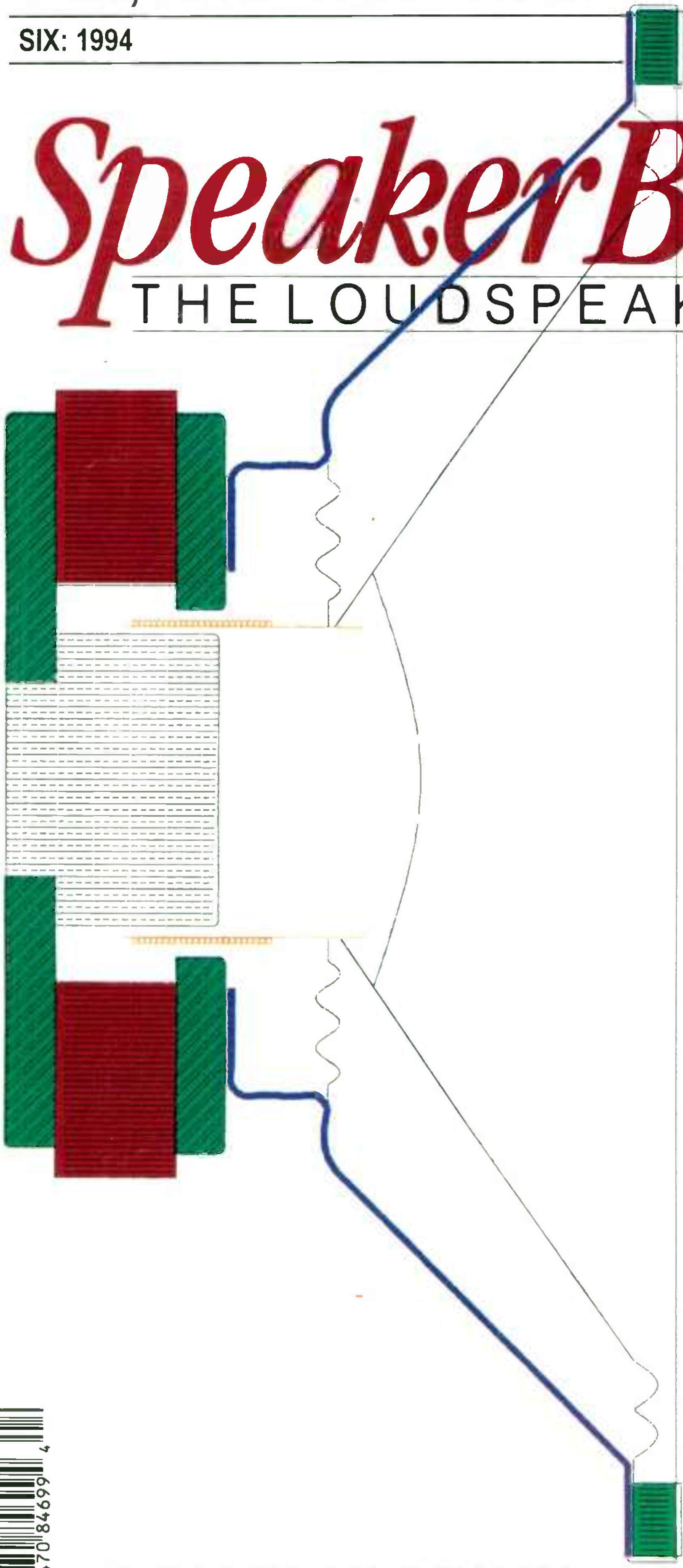
Speaker Builder

THE LOUDSPEAKER JOURNAL



HONEYCUTT:

SPEAKER GRAD SCHOOL

JENKINS:

DEMYSTIFYING DAMPING

WASLO:

IMP DOES SPLs

GALO:

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BULLOCK ON BENSON: A BETTER THIELE/SMALL?

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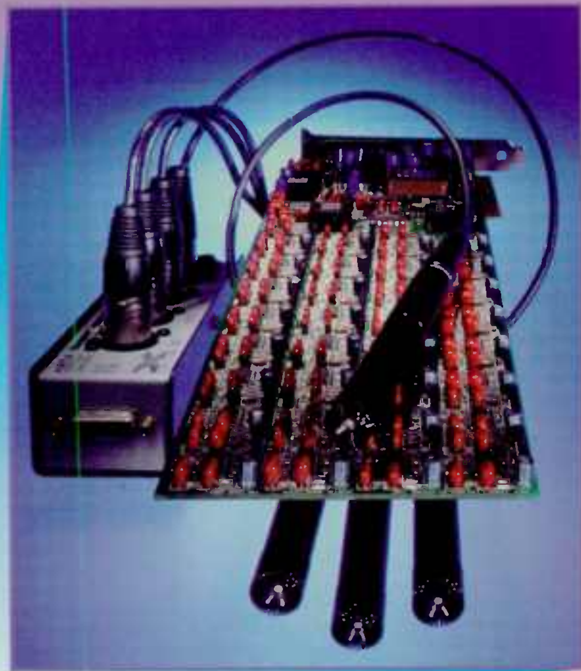

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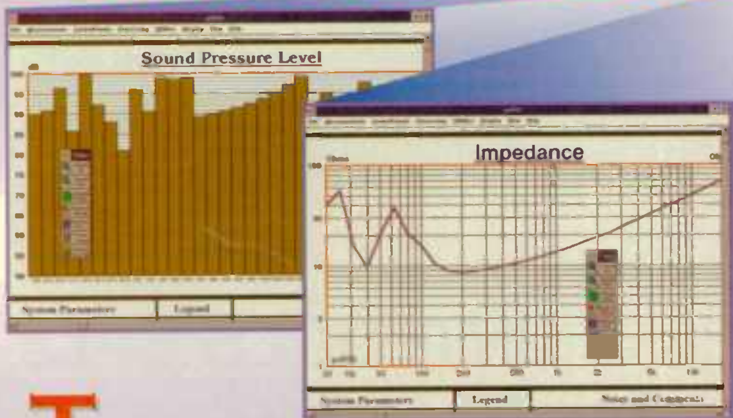

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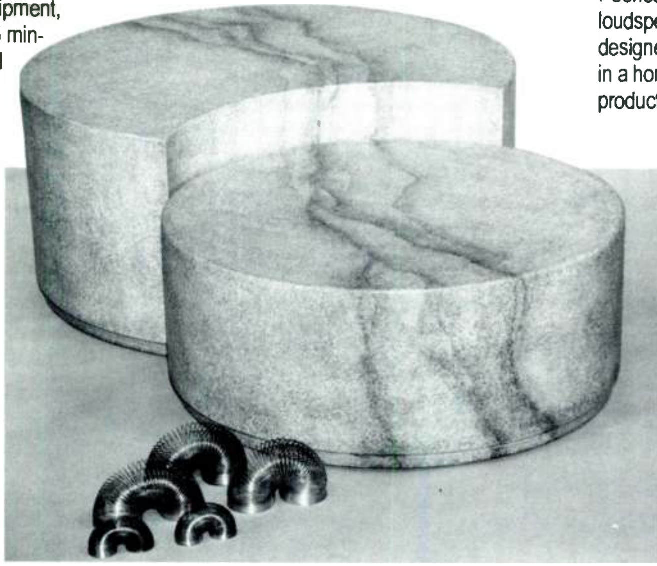
Good News

➲ STONE AGE

You can give your speakers a textured stone look with the Texturelac spray finish. Applied with standard spray paint equipment, Texturelac dries within 15 minutes. Stucco, leather, and pebble patterns are also available from Abilene Research and Development Corp., PO Box 294, Hewlett, NY 11557, (516) 791-6943, FAX (516) 791-6948.

Reader Service #108



■ LOUDSPEAKER U.

Menlo Scientific's Loudspeaker University—West will take place in November at the Airtel Plaza Hotel in Van Nuys, CA. Following the introductory (Nov. 15–16) and advanced (Nov. 17–19) workshops, there will be an adhesives workshop Nov. 20, and a ferrofluids workshop on Nov. 21. For more information call Mike Klasco, Menlo Scientific, (510) 528-1277.

■ HOME THEATER

Peerless introduces a line of loudspeaker units especially designed for the center speaker in a home theater system. The product line consists of three woofers (5", 6½", and 8") and two 1" dome tweeters (one is wide angle) and meets the demands of Dolby Pro-Logic Surround Sound and THX systems. Peerless of America, 1764 Rosalia Dr., Fullerton, CA 92635, (714) 990-3390, FAX (714) 990-3391.

Reader Service #101

■ m.a.r.s. ENCOUNTER

Fried Products announces a new ambience recovery system which promises concert hall sound in your living room. m.a.r.s. (McShane ambience recovery system, developed by veteran designer Chuck McShane) has been incorporated into Fried Products' loudspeakers. The new technology involves dual voice coil drivers, and an interconnect between the two speakers feeding L-R, R-L information back and forth is said to result in an improved illusion of the locality and presence of a live performance. Fried Products, 1323 Conshohocken Rd., Norristown, PA 19401, (800) 255-1014, (610) 277-1014, FAX (610) 277-4390.

Reader Service #102

■ VIDEO SPEAKER

KLH's new V-210 powered subwoofer is suitable for both home theater Surround Sound and hi-fi home stereo systems. With a 120W amplifier, 10" woofer, and selectable settings for frequency response, volume, and phase, the V-210 ($499) accurately reproduces the low frequency sounds of recorded music, movie sound tracks, and video game sound effects. Contact KLH Audio Systems at 11131 Dora St., Sun Valley, CA 91352, (818) 767-2843.

Reader Service #104

■ IN CHARGE

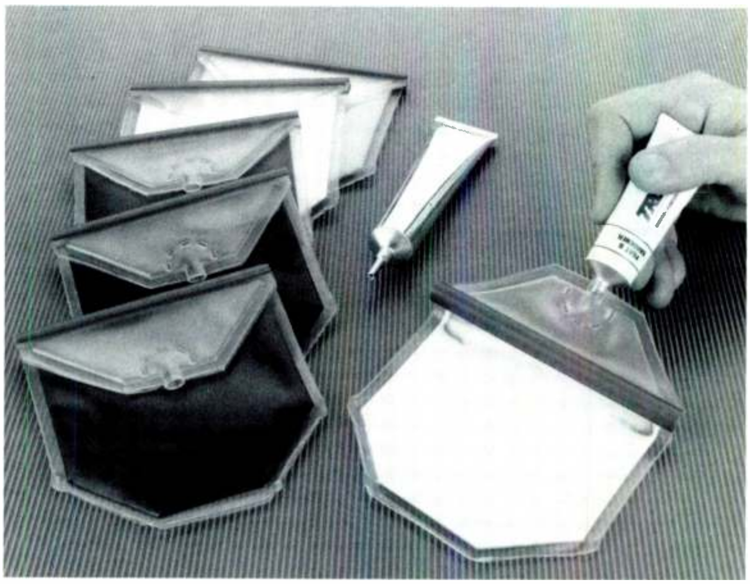
For the electrostatic loudspeaker builder, Amantis Audio offers a complete line of components, including ready-to-use interlocking electrostatic panels, high voltage power supplies, equalizers, and electronic crossovers. For more information, contact Amantis Audio, Inc., 184 N. Main St., Champlain, NY, 12919, (514) 858-7604, FAX (514) 858-7605.

Reader Service #110

➲ MIX IT UP

The Syon Packaging Brochure features a tube and bladder mixing and dispensing system and the Acu-Pak film package for delivering two or more part liquid and resin components in ratios from 1:1 to 100:1. Other packaging systems include a combination of plastic tubes, film pouches, and metal containers, with or without dispenser nozzles. Syon Multicomponent Packaging is available in 2g–5kg sizes from Syon Corp., 280 Eliot St., Ashland, MA 01721, (508) 881-8852, FAX (508) 881-4703.

Reader Service #107



■ BABY BROTHER

Parasound's HCA-606 6/5/4-channel amplifier has a circuit topology and operating features similar to its precursor, the HCA-1206, but has lower power output capabilities. This latest model, for home theater surround systems or multizone stereo installations, is rated at 65W × 6 into 8Ω with less than 0.1% THD, and 105W × 6 into 4Ω all channels driven. Channels 3/4 and 5/6 can be bridged to over 200W into 8Ω. Parasound Products, Inc., 950 Battery St., San Francisco, CA 94111, (800) 822-8802, FAX (415) 397-0144.

Reader Service #105

■ SACRED SOUND

Joseph De Buglio's book *Why Are Church Sound Systems and Acoustics So Confusing?* sheds some light on the sometimes obscure subject of church audio. The updated version, available in soft cover or binder editions, includes new charts, articles from other industry professionals, and more illustrations. Order from JdB Sound Acoustics, 63 Lockerbie Ave., Toronto, ON M9N 3A3, Canada (416) 252-6624.

Reader Service #109

Speaker Builder (US ISSN 0199-7920) is published every six weeks (eight times a year), at $32 per year, $58 for two years; Canada add $8 per year, overseas rates $50 one year, $90 two years; by Audio Amateur Publications, Inc., Edward T. Dell, Jr., President, at 305 Union Street, PO Box 494, Peterborough, NH 03458-0494. Second-class postage paid at Peterborough, NH and an additional mailing office.

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← ***Reader Service #36***

Good News

■ LOOK WHO'S FORTY

Acoustic Research turns 40 this year, and will celebrate with a gala party in late October at New York's Grand Central Station, the site of its demo room for many years. Company founders and former employees (such as Ed Villchur, Henry Kloss, and Roy Allison) will attend. Consumer invitations are limited, so send your request to Acoustic Research, 535 Getty Ct., Bldg. A, Benicia, CA 95842.

■ HAPPY 20th

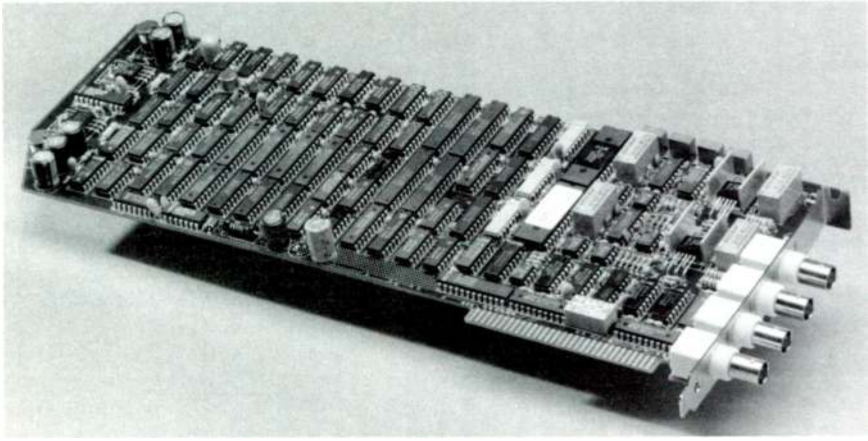
October marks the 20th anniversary of Miller & Kreisel Sound Corp., and also celebrates the birth of the subwoofer category. Founded in 1974 as both a speaker manufacturer and an audiophile recording label, M&K Sound plans a special 20th Anniversary model satellite speaker, the S-1C, which features a new application for its push-pull dual driver system. M&K Sound, 10391 Jefferson Blvd., Culver City, CA 90232, (310) 204-2854, FAX (310) 202-8782.

Reader Service #113

■ CUSTOM CLEAN

Cleaning your potentiometer became much easier with the introduction of the Guzintah. Use this device with any chemical spray cleaner to internally clean the resistive element and wiper without having to take apart your equipment. Different thread sizes are available in unified or metric thread. P.K. & Associates, 6238 SH 109, Delta, OH 43515, (419) 537-3128.

Reader Service #115



⊂ MAKING WAVES

The PCI-311 and PCI-312 Arbitrary Waveform Generators occupy one PC expansion slot and combine the performance of stand-alone generators with the convenience of the personal computer. The single channel PCI-311 ($1,495) and the dual channel PCI-312 ($1,995) feature 12-bit output DACs with output rates to 50 megasamples/second, and 12V P-P into 50Ω. Optional BenchTop software ($495) provides a graphics user interface. Contact PC Instruments, Inc., 9261 Ravenna Rd., Bldg. B11, Twinsburg, OH 44087, (216) 487-0220, FAX (216) 425-1590.

Reader Service #103

■ NEW DISTRIBUTOR

Polydax Speaker Corp. has named Speaker City of Burbank, CA, as an authorized distributor of the Audax name and product line. With more than 30 years of experience in the audio field, Speaker City specializes in home and car audio products and offers personalized listening demos. Contact Speaker City USA, 115 South Victory Blvd., Burbank, CA 91502, (818) 846-9921, FAX (818) 846-1009.

Reader Service #112

■ A GOOD BUY

Electronix Express introduces the latest version of its product catalog. The 1994 edition features a variety of electronic equipment, tools, and components for audiophiles. To obtain your 120-page catalog ($2), contact Electronix Express, 365 Blair Rd.,Avenel, NJ 07001, (800) 972-2225, FAX (908) 381-1572.

Reader Service #119

⊃ FREE CATALOG

MCM Electronics' latest catalog (#33) contains over 2,800 new items, including semiconductors, speakers, project accessories, connectors, test equipment, computer products, audio, TV, and VCR. Of particular interest to service technicians and electronic enthusiasts, this catalog also introduces three new product categories, including amateur radio, CCTV, and wireless security systems. To request your copy, write to MCM Electronics, 650 Congress Park Dr., Centerville, OH 45459, (800) 543-4330.

Reader Service #117

■ MUTE MEDIUM

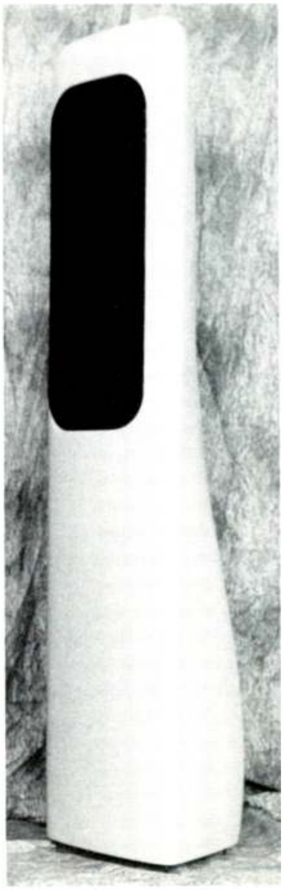
New England Audio Resource is now offering two new speakers encased by a veneer of Corian®, DuPont's acoustically inert solid surface material. The manufacturers claim that Corian, laminated in a ½" layer over medium density fiberboard, eliminates colorations caused by conventional speaker cabinets, and drastically reduces cabinet resonances. For more information, contact N.E.A.R., 12 Foss Rd.,Lewiston, ME, 04240, (207) 795-0609.

Reader Service #114

⊃ ROSINANTÉ

Rosinanté has a new theory called The Evolution. This 8Ω system with 93dB sensitivity will handle 50–400W and has a claimed frequency range of 28Hz–22kHz. It fits easily into the average-sized room and is available in white or black marble finish. Rosinanté, 602 Acorn Plaza, Eudora, KS 66025, (913) 542-3922.

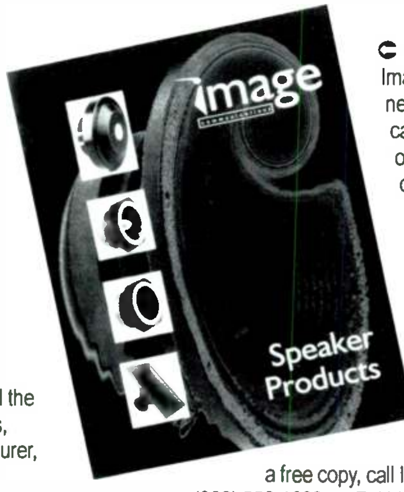
Reader Service #106



Good News

■ **ICC CATALOG**
International Components Corp. (ICC) has released the 1994 version of its Electronic Components and Accessories Catalog. Products include video and telephone cables, connectors, antennas, and speakers, as well as a variety of tube types. ICC, 105 Maxess Rd., Melville, NY 11747, (800) 645-9154, FAX (516) 293-4983.
Reader Service #118

■ **STUDIO SERIES SPEAKERS**
Duntech Audio has recently released the PCL25 loudspeaker, which produces, according to the Australian manufacturer, accurate sound reproduction from a compact design. Incorporating "pulse-coherent" technology, the unit includes twin 6.3" dynamic bass drivers flanking a 1" dome tweeter. Features include response to below 40Hz (-3dB at 55Hz), 89dB SPL, 4Ω nominal impedance, and 500W power rating. Price is $1,995 / pair. Duntech, 111 South Dr., Barrington, IL 60010, (708) 526-1681, FAX (708) 526-1669.
Reader Service #111



SPARE PARTS
Image Communications' new Speaker Products catalog lists many types of complete speaker components including Eminence loudspeakers, drivers, and diaphragms; Acoustician drivers, horns, tweeters, diaphragms, and accessories; and McCauley loudspeakers, drivers, and horns. For a free copy, call Image at (312) 585-1212 or (800) 552-1639, or FAX (312) 585-7847.

■ **BASS BOX**
The AlumaBass Professional and AlumaBass Star are Bullfrog's latest loudspeaker enclosures for bass guitar systems. Housing the company's AlumaPro speakers, which feature an anodized gold finish, each cabinet is constructed of ¾" composite-plywood covered with black carpeting. Bullfrog, Inc., 1503 Prairie Ave., South Bend, IN 46613, (219) 233-4151, FAX (219) 233-3521.
Reader Service #116

■ **SOUNDPROOF**
Fireflex acoustic foam panels are designed for sound absorption in high heat environments, or where fire-rated sound proofing is required, such as churches, schools, and theaters. It is also ideal for lining enclosures in high-temperature areas for sound proofing. Made of Melamine® foam, each panel measures 24" × 48" × 2". NetWell Noise Control, 6125 Blue Circle Dr., Minnetonka, MN 55343, (612) 939-9845, FAX (612) 939-9836.
Reader Service #121

■ **TOOL SOURCE**
The latest Woodworker's Supply catalog is full of tools and accessories for both amateur and professional woodworkers. Find a wide variety of sanders, saws, planers, bit sets, handles, knobs, veneers, and much more in this $2 catalog available from Woodworker's Supply, Inc., 1108 North Glenn Rd., Casper, WY 82601, (800) 645-9292.
Reader Service #120

About This Issue

Ribbon loudspeakers, with their small design and low efficiency, traditionally presented a problem for speaker builders. In this issue, developer **John Meyer** overcomes these difficulties with his plans for a large ribbon module beginning on p. 10. This construction project promises good results for DIY enthusiasts of all abilities.

Loudspeaker history is rich in interesting personalities, ingenious developments, and timely breakthroughs, not to mention a fair amount of happenstance. In an informative, entertaining style, contributing editor **Gary Galo** brings insight and order to this story in the first part of his look at how we started and how far we've progressed ("Loudspeakers: A Short History," p. 18).

For a more up-to-date industry profile, turn to **Richard A. Honeycutt**'s extensive report on Loudspeaker University (p. 32). Some will argue that this three-day gathering of industry pros was long overdue in sharing information and addressing industry concerns. Our patience is rewarded, however, as the author provides a glimpse of some new technologies we can anticipate.

Many questions about the damping factor exist, including misconceptions regarding how to determine its value. **Donald Jenkins** tackles this complex topic (p. 26), and offers a technique to calculate a system's true damping value.

Bill Waslo is our guide on another journey with IMP. Using the latest version of this versatile program, he shows us how to extend the flexibility of our measurement techniques in determining absolute SPL sensitivity (p. 22).

Woodworking wizard **Bob Wayland** presents another installment on building quality enclosures (p. 42). This time he reveals the secret to making clean, strong corner joints with splines.

With the introduction of a new Mac program entitled The Listening Room, it just became a little easier to determine optimum listener/speaker locations. **Victor Staggs** puts this acoustics program through its paces beginning on p. 44.

Plus, **Robert Bullock** uncovers the significant work of J. Ernest Benson on speaker design and performance, and **G. R. Koonce** provides a simple modification to add vernier tuning to a function generator.

Speaker Builder

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

–JOHN STUART MILL

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A one year subscription to *Speaker Builder* is $32. Canada please add $8. Overseas rate is $50 per year.

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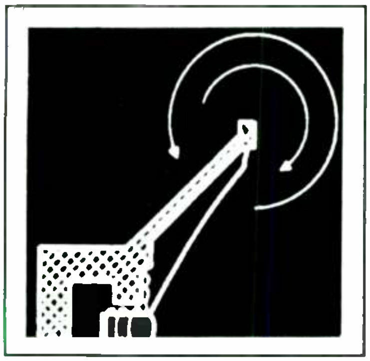

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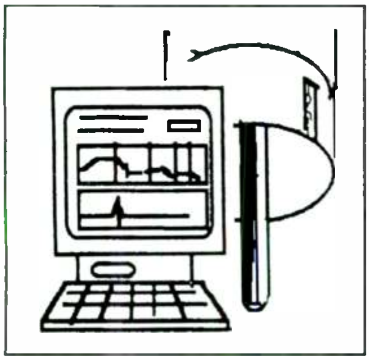

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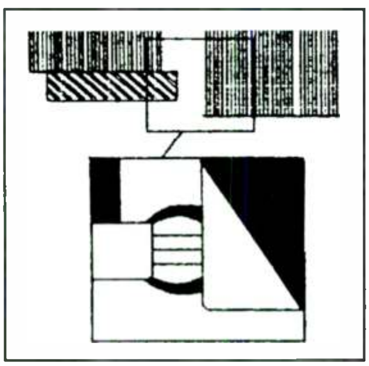

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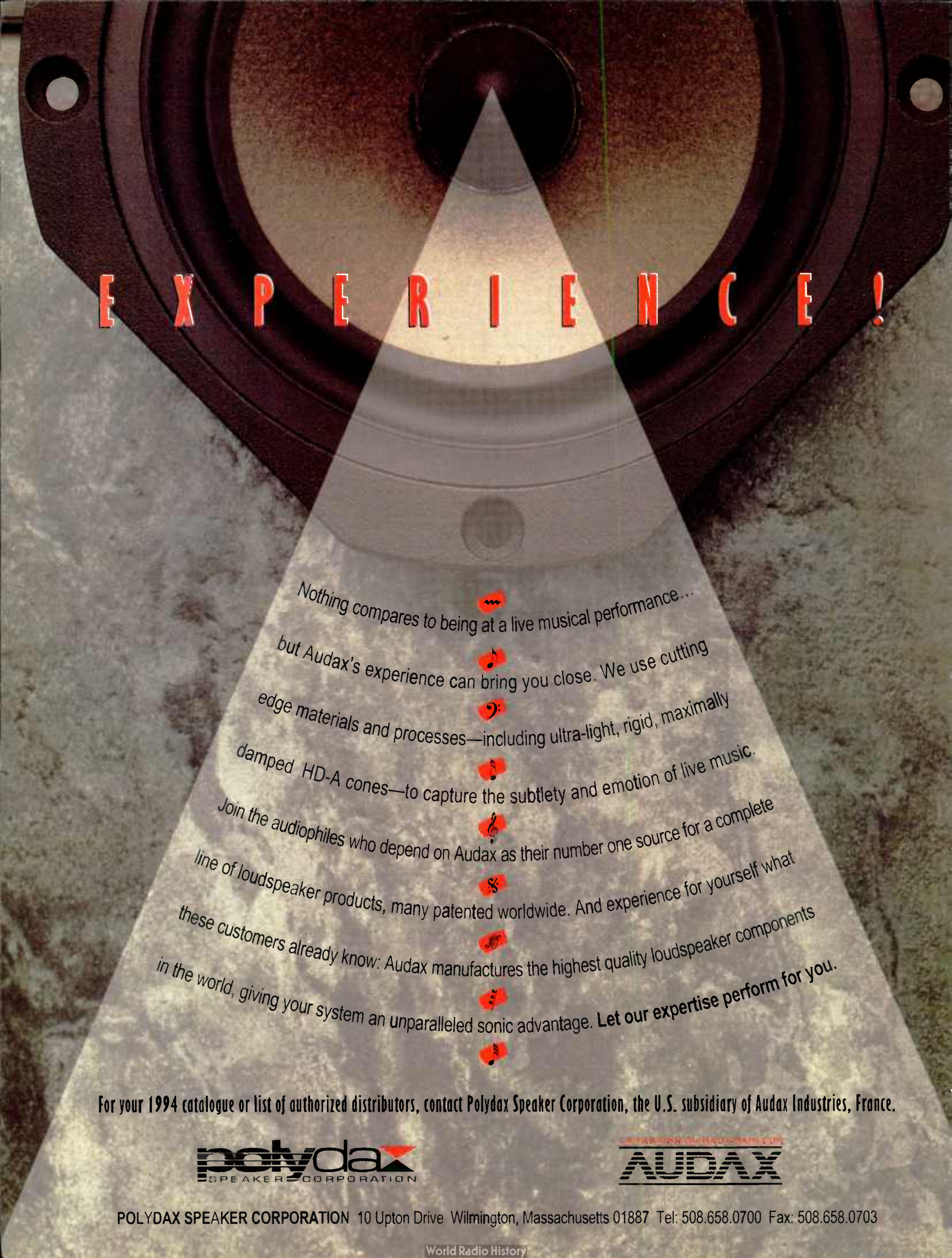
Speaker Builder

THE LOUDSPEAKER JOURNAL
VOLUME 15 NUMBER 6 AUGUST 1994

DEPARTMENTS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Editorial

Any Art in This Science?

I recently spent a fascinating evening in our nearby metropolis (in New Hampshire that's over 10,000 people) listening to a professional artist talk about how he works. What was startling about the whole matter to the nearly 100 of us in the audience was his heavy reliance on what I can only refer to as craft techniques to produce his exceedingly beautiful acrylic paintings.

Ed Gordon, the artist, was perfectly open and candid about how he works. But first I'll try to draw, in words only, the picture he painted. Visualize a ladder-back maple chair with a rush seat resting on a wide-board lightly stained wood floor which has been varnished and polished to a high gloss. The chair sits nearly two feet in front of a half-open, mullioned window through which a lawn and a large fruit tree are clearly visible. Adjacent to the right of the window is a tall bookcase containing books, a pair of glasses, and a paperweight, with white panelled doors with brass knobs fronting the deeper, bottom half of the case. The afternoon sunlight streams into the room from an out-of-the picture window on the right, making a brightly glowing light-pool on the floor under the chair.

The colors are brilliant and every detail in the picture stands out sharply, but not quite photographically. The painting seems to glow with a kind of soft iridescence, as though the light represented in it is radiating from within it. My memory of it is still vivid even after nearly six months' time.

All his hearers involuntarily uttered a barely whispered gasp when Gordon blithely unrolled a full-sized pencilled sketch of the drawing done on a sheet of Mylar®, confessing that his first step was to transfer the lines to his picture's tempered masonite base. The dismay became even more palpable when he produced Polaroids® of the scene. The room became even quieter as the artist said he had painted this same picture several dozen times.

Gordon went on to explain that he used whatever practical shortcut techniques he could devise to make his painting easier. He had switched from oils to alkyd paints years earlier for the practical consideration that the latter dry overnight whereas oils take days, even sometimes weeks. He had painted other scenes, of course, to the evident relief of some. But in doing an interior of a restored 17th century saltbox house, he had a rubber stamp made to reproduce the stencils running around the walls, making some fellow-artists visibly uncomfortable. Art, apparently, isn't supposed to contain mechanical (read scientific) methods. It is expected of artists that their work be a mix of inspiration and skill, not to say magic.

Audiophiles and speaker builders often ask the classic question about the mix of art and science in both audio design and construction. Respectability and repeatability rest on science in audio. The art is allowed in to cover guesses where we don't know enough to be scientific. But is that all?

Reader Service #21

Audio amateur types go about with a slightly defensive air about their handiwork too much of the time. I think this arises often because we do not pay all that much attention to the aesthetic question, or we only consider it after the project is completed. I confess I was at first moved to laugh somewhat scornfully at photos of Bowers & Wilkins' new shape for their new Nautilus speaker (*SB* 4/94, p. 4). The four-way looked like something (aka the Giant Snail) Dr. Doolittle might seek in the South Seas.

But that new speaker demonstrates a process which takes the end aesthetics of the design into account at the inception of the project. Whether the shapes were chosen first and the driver technology shaped, Procrustean-like, to fit the spaces, I do not know. But at the least, recognizably natural shapes were chosen for the Nautilus which may well have acoustic merits nature thought up some long while ago.

By the end of Ed Gordon's talk and his excellent answers to some questions from skeptics, critics, and even some angry purists who made it clear they regarded him as a certifiable philistine, I found I really liked and responded to his work. Some in the audience seemed less reluctant to quibble when it came out that Gordon sells all the versions of his demonstration picture he can find time to paint at $8,000 each. Others were less willing to cavil when we learned he had just won first prize in *The Artist*'s national landscape competition.

As I looked again, and yet again, at the beauty of the painting on the stage, it took on a warmth, a genuineness of tribute to reality that no amount of knowledge about the author's tricks, if that's what we choose to call them, could diminish. The picture gathered a kind of momentum of beauty which was far more, for me at least, than just a bunch of clever techniques. It paid genuine homage to the reality it represented.

My story about Ed Gordon and his somewhat practical, almost mechanical, approach to painting is, in some ways, the reverse of our situation as audiophiles. We err, I think through neglect, on the side of the practical, get-it-done pressures in making our loudspeakers. Isn't there a place for something beyond the technical, something more than a quick coat of paint or even a painstaking veneer job or an especially attractive grille?

Our dazzling sonic performance and utility ought sometimes to be matched by an equivalency of beauty. I can't say how we might achieve this. But I think all of us should give some quality time to thinking about it.—E.T.D.

A LARGE RIBBON YOU CAN BUILD

By John Meyer

About 20 years ago, I walked into a stereo shop in Toronto and paused just inside the door. I was looking for the guitarist, since the store seemed to be hosting one of the "live versus our state-of-the-art equipment" demos that were so popular in those days.

There was no live performer, only some large black panels raised about 3′ above the ground. These turned out to be Dayton Wright electrostatics, which were my first exposure to film drivers. I was hooked. I attended many listening sessions and several meetings with Mike Wright, the brilliant but erratic (don't these characteristics always seem to go together?) designer. Since I was a part-time designer myself, any audience with "god," as Mike was known in the audiophile community, was as highly coveted as finding the operating manual from an abandoned alien spaceship.

PHOTO 1: Module 8, with 8″ ribbon crossing over to an 8Ω 5″ Peerless mid-bass at approximately 1.3kHz. Ribbon impedance is 11Ω; sensitivity is 85dB.

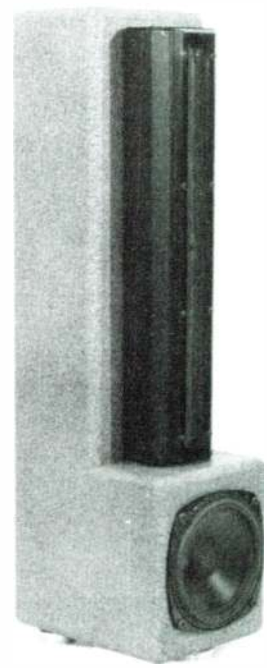


PHOTO 2: Module 15, with 15″ ribbon crossed to a 4Ω Peerless 5″ at approximately 1.2kHz. Ribbon impedance is 6Ω; sensitivity is 87dB.

PROMISE AND PITFALLS

Despite the breakthrough nature of their sound, some electrostatics contained many problems in those days. Reliability, impedance, sensitivity, placement, frequency response, and several other issues made ownership suitable only for those with both deep passion and deep pockets. Ribbons also existed, but the first available to the OEM market was the Strathearn panel. This unit offered the same openness, transparency, definition, and air as the larger panels, but came in a smaller package at a much lower price. Nirvana!

Not quite. The magnets had a habit of voting with their feet, so to speak, by jumping into the gap and rupturing the diaphragm. A transformer was required, and so many parts in its construction rattled that the "music" continued long after the band stopped playing. A spectral time delay plot of one of the early units provided hours of interesting study.

But the fundamental promise of planar drivers could not be denied. With extremely low mass and driven over their entire diaphragm area, film drivers avoid the problems associated with their heavier cone and dome siblings. Not only do cones and domes have far greater mass, but they also have far less directly driven area (the apex of the cone and the perimeter of the dome) with which to achieve perfect control. Planar film drivers are also inherently phase coherent over their entire bandwidth, whereas cones and domes have a built-in phase "smear" due to their depth (or profile).

Loudspeakers are merely air pumps. The speaker with the greatest fidelity is the air pump producing the fewest errors (room interactions aside). Film drivers have the greatest potential for making the fewest errors.

THE PROJECT

When we began to develop our own ribbon drivers (8″, 15″, and 30″) in the mid '80s, we wished to capture the musical characteristics which were so seductive in film drivers, but in a design that was easy to drive, easy to place, and, above all, reliable. Hence the inspiration for the Newform Research line of speakers, kits, and raw ribbon drivers.

This construction project is quite simple, yet the results rival the large film drivers in their strong points and greatly surpass them in their weak points. It involves building a

ABOUT THE AUTHOR

John Meyer developed the first flat aluminum honeycomb diaphragm loudspeakers (Coherence) to be produced in North America in the late '70s. He has a patent on enclosure technology which allows a one-piece multi-laminate shell to be produced reducing "wall talk" over a good MDF enclosure by 9–27dB. Ribbons have always been a source of fascination and frustration and lately of joy. Future plans include a very large wind generator to avoid a repeat of last winter's marrow crystallizing experience.

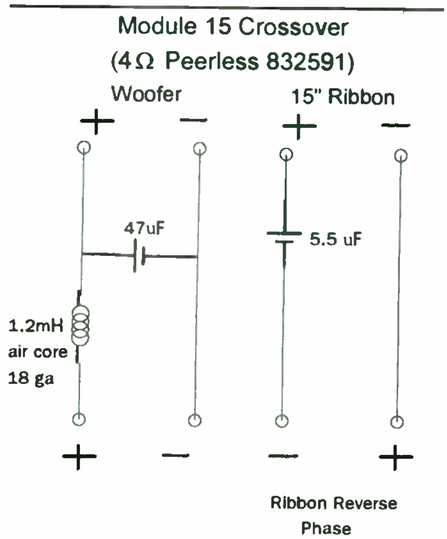


FIGURE 1: Module 15 crossover (4Ω Peerless 832591).

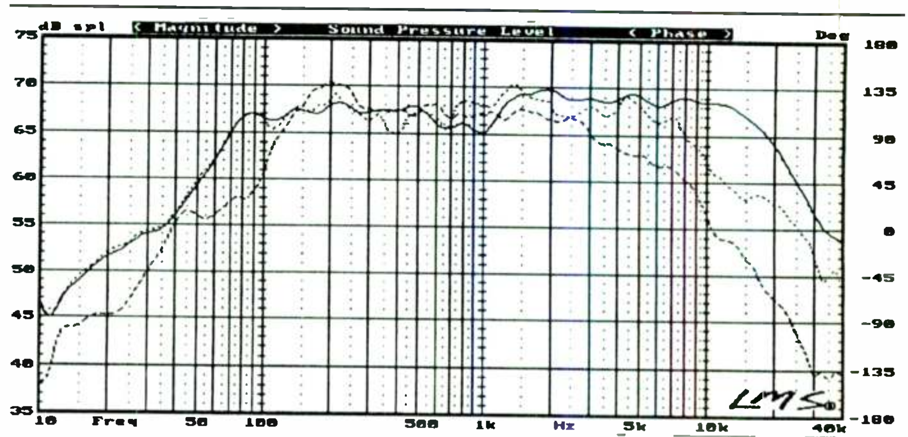


FIGURE 2: Module 15 curves (0° = dash, 45° = dotted, 90° = dash/dot). Note that the curves are extremely tight out to 3k, which contributes greatly to the soundstaging capability.

ribbon module to cover the range from 100Hz–20kHz, with the bottom end covered by the subwoofer of your choice. This combination works as well in an audiophile's system as it does in a home theater. In fact, in five years there probably won't be a distinction between the two. The modular approach allows easy placement for performance as well as aesthetic considerations.

Room setup and speaker placement is critical, almost as important as the loudspeakers themselves. The two main performance considerations are the bass and the soundstage. It is a very pleasant surprise when both these factors can be optimized in the same location. The modular approach allows you to solve the two problems independently by placing the ribbon modules for optimum soundstage and placing the subwoofer for optimum bass.

One more caveat concerns the dreaded "spousal approval factor" (SAF). The house-proud non-audiophile typically resists placing large irregularly shaped and un-color-coordinated objects thrust at odd angles into the main living areas. Modules provide more placement options and a vastly lower profile. Problem mitigated!

DESIGN THEORY

The module consists of a 15″ ribbon driver crossing over to a 5″ mid-bass driver at 1.25kHz (*Fig. 1*). The 5″ unit, housed in an L-shaped MDF (medium density fiberboard) enclosure, is mounted in the front panel of the "L" foot while the ribbon sits on the foot, staggered back slightly. The enclosure is only 6″ wide and just tall enough to allow mounting of both drivers. Overall, this project is like building a very simple two-way, except the dome is replaced by the much larger ribbon and the cabinet is slightly more complex.

Crossover slope is 6dB/octave for the ribbons and 12dB for the bass drivers for minimum signal processing and maximum coherence. The 1.25kHz crossover point is a full two octaves below the point at which the 5″ driver begins to beam or break up. For this reason, both integration and speed are excellent. The family of curves (0°, 30°, 60°, and 90°) is also extremely tight (within a 5dB window out to 3kHz), which, when combined with the system's minimal acoustic profile, yields a very broad, well-focused, deep soundstage (*Figs. 2* and *3*).

In a normal two-way the woofer or mid crosses to a tweeter, at 2.5k at least. By that point, even a 5″ will be down 18dB at 90°. Of course, 6½″ and 8″ drivers become worse even earlier, as you can see on the *Fig. 3* graph, where the huge off-axis suckouts display this characteristic. The soundstage suffers accordingly.

As is typical of a ribbon or panel driver, the vertical dispersion is poor. These systems are definitely made for the seated listener. However, tilting the speakers and/or using the 30″ ribbon can largely eliminate any dissatisfaction with the standing position.

In a home theater application the poor vertical dispersion can be a big benefit, since it reduces stray vertical reflections which degrade the "sound localization" capability so vital to the surround effect. The system's limited vertical dispersion, in other words, is indicative of a "well-controlled radiation pattern."

Many people think off-axis consistency is also critical to a proper soundstage. That is, the on- and off-axis family of curves (leaving out the verticals) should be tight until the inevitable higher frequency rolloff splits them up. This consistency yields a deeper and more focused soundstage because differ-

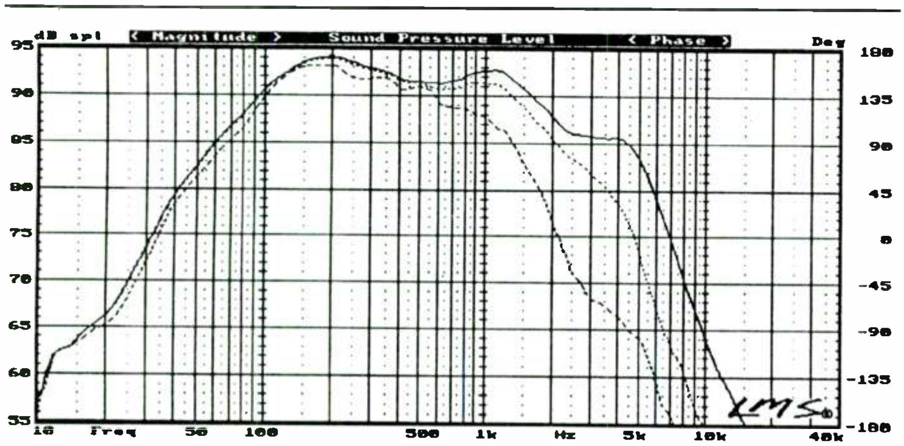


FIGURE 3: Off-axis response. Even a 5″ driver is down over 12dB at 2k at 90° and 18dB at 3k. That is why a low crossover is so critical to consistent off-axis response and hence to the integrity of the soundstage. The larger the bass driver, the greater the midrange suckout off-axis. I used a single inductor and smoothed the curves heavily to get rid of the trash and concentrate on the issue.

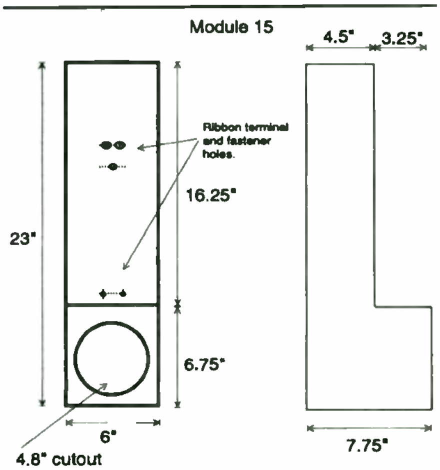


FIGURE 4: Module 15 dimensions, showing holes for bolts and ribbon leads.

ent elements of the sound spectrum will not be displaced in their spatial location relative to other elements. The desire for a tight family of curves was a prime reason we designed our ribbons to work well down to 1k. We thought an 8" woofer (as used in our full-range systems) could be used reliably up to that point.

TABLE 1

MODULE 15 (AND MODULE 8) PARTS (FOR A PAIR)

MDF or PB	¾"
woofers	2 of Peerless 832591—4Ω (832592—8Ω, for Module 8)
wood screws	8 of #6 for the terminal plates 8 of #10 for woofer mounting
capacitors	2 of 47µF (24µF for Module 8), 2 of 5.5µF
inductor	2 of 1.2mH (2.3mH for Module 8)
terminal plate	2 of 4 binding post type (3¾" x 4¾")
stuffing	1 of 1lb bag loose polyester stuffing
grille	perforated steel 1/8" holes, 1/8" centers
ribbon shims	beveled rubber washers
grille cloth	the usual type, color of your choice
miscellaneous	wood glue for cabinet silicone for terminal plate foam tape for woofers

RESULTS

Our full range designs have more than lived up to our expectations, but the soundstage produced by the ribbon modules is truly exceptional. They use 5" midbass units crossing over to the ribbons at between 1.2k and 1.4k (there are three models). The cabinets are only 6" wide. The modules produce an absolutely exceptional soundstage. Focus and depth are consistently excellent on a broad number of sources. We were surprised that they surpassed our full range systems in this respect, and did so decisively because of the very narrow cabinet and the 5" versus 8" driver.

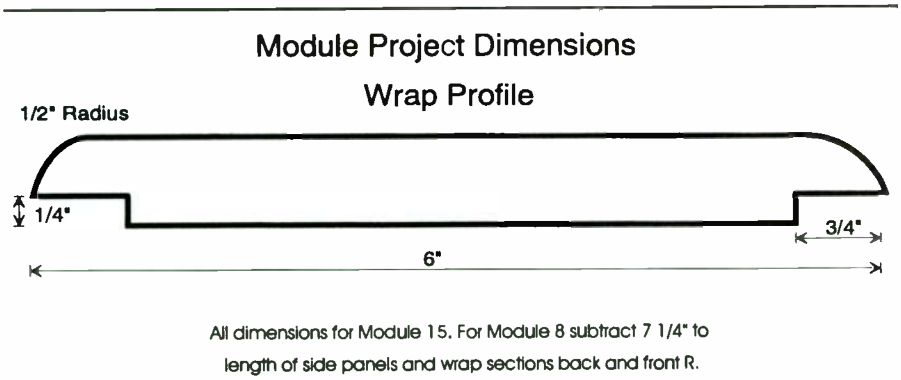
Given the excellent sound-staging capabilities, you might be tempted to use the "Phantom" setting on your Pro-Logic decoder (please don't talk to me about THX), without using a center channel. If you think you really need the center, the ribbon is largely shielded, requiring only some modification to the bass driver. Three options are available: You can use a shorting magnet on the back of the woofers' backplates, you can put some steel shielding inside the cabinet, or you can use the Peerless shielded 5" 832591/2 models.

With the center channel, as well as the rear channels, you may wish to use the 8" ribbon for cost and ease of placement considerations. Note that the center and rear channels are only fed signals with the bandwidth of 100Hz–7kHz. However, future versions of Dolby (especially digital) may expand the bandwidth to full spectrum. The bottom end can always be "subbed," but all speakers should have full treble capability. Dolby also recommends that all speakers have the same characteristics for best effects and smoothest integration.

PHOTO 3: Empty Module 15 cabinet with granite finish and radiused corners.

ASSEMBLY

Electrically, this project, with its three crossover components, is extremely simple. The wood parts are easy to construct for someone with a table saw and a router. Precision here is an asset, but if you apply a



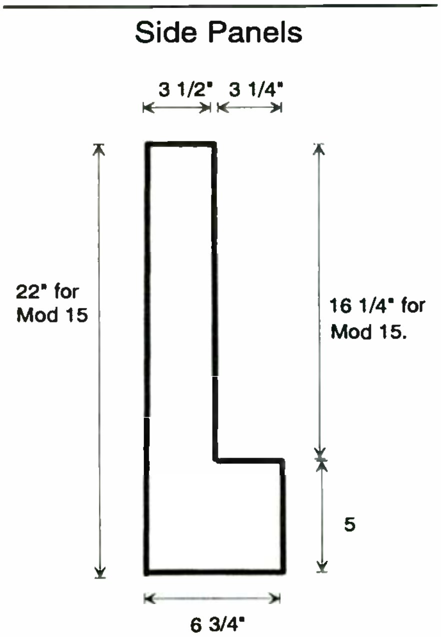


FIGURE 5: Module 15 dimensions for wrap (left) and side panels (right). For Module 8 dimensions, subtract 7¼" from the length of the side panels and wrap sections back and front R.

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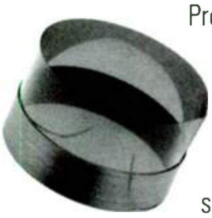

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CONES OF KAPTON

DUPONT

spray-on granite finish, you can patch your mistakes with no lasting ill effects. While MDF is a popular choice for your enclosure material, given the small unsupported enclosure surfaces, any good-quality particle board will do, depending on the finish you desire.

The enclosure requires no complex internal bracing or baffling obstacles, and the main time variable will be the finish. You can even cut the wood parts with a jig saw and fill and sand the seams for about the lowest tech approach possible. A black lacquer cabinet, however, will require the utmost precision at every stage. The project could require up to 20 hours for the many coats this finish will entail. With a spray-on granite, six hours is a reasonable estimate.

You must mount the ribbon to the upper front baffle using bolts (three for the R15 and four for the R8). These bolts (supplied with the kits) hold the ribbon above the "foot" of the enclosure, reducing resonances and noise. The bolts pull the ribbon back against the front of the baffle, where conical rubber washers isolate it. This further reduces anomalies. More work could go into the choice of isolation material, however.

The bolts are accessed through the terminal hole on the back. If the terminal hole is not at the right height, the bolts cannot be tightened by terrestrial hands. In the case of the Module 15, the ribbon leads are fed up from the crossover (mounted on the floor of the cabinet) through the two 7/8" holes which later house the ribbons' five-way binding posts. The wire leads coming out of the 8" ribbon's bottom cap are fed through a hole in the top of the "foot" to the crossover leads (*Fig. 4*).

Surrounding the holes on the baffle with foam tape and compressing it as the bolts are tightened prevents air leaks from the midbass air volume. The rest of the assembly is completely conventional using standard parts (Peerless 5" and four binding post terminal plates) from your local audio component dealer. See *Figs. 5* and *6* for project dimensions.

TABLE 2

TOOLS REQUIRED

- Table saw
- Jig saw
- Drill
- Router
- Sander?
- Soldering Iron
- Screwdriver

TABLE 3

TIME REQUIRED

Cut wood, including drilling holes	2 hours
Assemble the cabinets	1 hour
Finish the cabinets	varies: 15 minutes to a lifetime
Wire and mount the crossover	1 hour
Mount the ribbons	20 minutes
Stuff the cabinets	10 minutes
Mount the woofers	10 minutes
Affix terminal plate	10 minutes
Make the grilles	2 hours or buy them

TABLE 4

SUGGESTED SUPPLIERS

Midbass driver: Peerless Model 832591/2 from Solen. Approximately $30 US each.

Cabinet: StudioLab Audio, a small speaker builder in the Toronto area. They make extremely good basic enclosures in small runs or one-ofs on a custom basis, and offer their own line of speakers and a full line of parts. Their prices are competitive and they can supply a variety of finishes from black lacquer to veneer (although with these radiuses, veneer is not an option for this enclosure). Since most builders will probably wish to finish their own enclosures, the prices are for the bare MDF. Call for a custom finish. Cost for the bare enclosure is $65 US each, UPS delivered.

Ribbons: Newform Research Inc. Prices, per pair, are $330 (R8), $530 (R15), and $880 (R30). Prices are US$ and include UPS delivery.

Crossovers: Newform Research Inc. $26 US per pair, UPS delivered.

Grilles: Newform Research Inc. Perforated steel, epoxy powder coated and cloth covered. Call for prices.

Entire kit (everything except the cabinet):
Module 8 $428 / pair
Module 15 $614 / pair
Module 30 $944 / pair

Newform Research Inc.
PO Box 475, Midland, ON, Canada L4R 4L3
(705) 835-9000, FAX (705) 835-0081

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

StudioLab Audio Inc.
29 Bermondsey Rd., Toronto, ON, Canada M4B 1Z7
(416) 757-3265, FAX (416) 757-3266

SUBWOOFERS

Numerous plans exist for subwoofers, and you can produce some very good results for a modest investment. Peerless, for example, produces an 8" dual voice coil (831858) and publishes several different alignments. The 40-liter cabinets, with simple crossover, are easy to build. Therefore, you can achieve amazingly good bass either in kit or assembled form, so I won't deal with the subwoofer issue further.

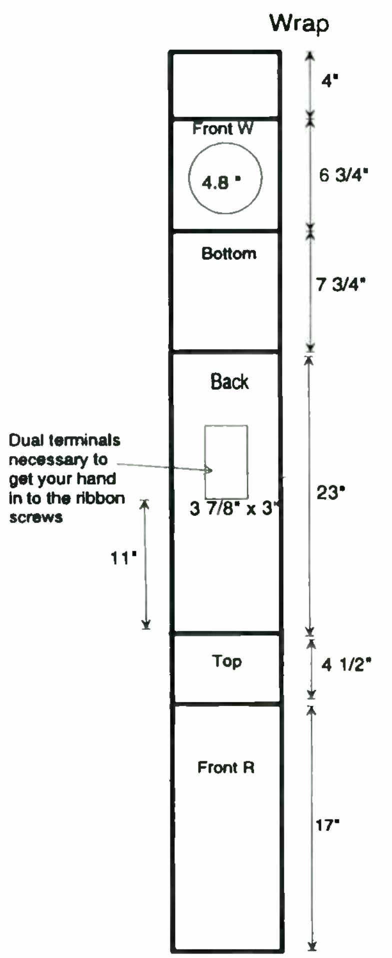

FIGURE 6: Ribbon dimensions.

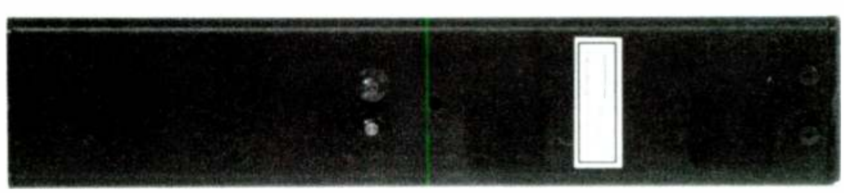

PHOTO 4: Back view of the Ribbon 15. Note binding posts and two bolts at the bottom with one larger one in the middle.

PHOTO 5: Ribbon 30 mounted on the Module 15 enclosure. This unit should be screwed onto a broad-based stand.

PLACEMENT

The ribbon diaphragms are only ¾″ wide, so they radiate horizontally like a ¾″ dome. Since dispersion is a function of piston diameter and wavelength, you can predict that more energy will radiate to the sides between 1kHz–3kHz than with any of your other speaker systems. The benefit of this is good sound staging (as well as the small midbass driver and the narrow enclosure). Depth, width, location, and focus are all there in living vibrations.

However, you must take care to break up the side waves if you have hard side walls such as drywall or windows. Otherwise, the side waves will produce a strong reflection at the listening position and foul the soundstage.

A strategically placed plant, bookcase, or tapestry will work wonders. You can cheat and use a mattress, but we are trying to achieve state-of-the-art, not state-of-the-bludgeon. Besides, some level of reflection is necessary for a proper soundstage.

Stands—16″ for the R30, 20″ for the R15, and 24″ for the R8—are definitely required since the ribbons must be at ear level. You may also wish to experiment with tilting. Being tall and heavy, this system requires you to pay close attention to the size of the base and the possible attaching of the speakers to the stands with screws.

This project is very straightforward, and takes little time. The results are comparable to the best efforts of brand-name manufacturers. Great sound can come from small packages (especially if you have a sub lurking in the shadows) and is much less expensive than anyone would have imagined even two years ago. Technology (and the hobbyist) march on.

Part 1

LOUDSPEAKERS: A SHORT HISTORY

By Gary A. Galo
Contributing Editor

In 1925 the most significant advance in the history of recorded sound occurred—the commercial introduction of electrical recording and reproduction. Two distinct technological breakthroughs made this event feasible.

First, the development of the transducer allowed equipment to transform mechanical energy into electric current, and vice versa. Electrical recording required two transducers—a microphone to convert sound vibrations into electricity and an electrical disc cutting head to turn the electricity into a mechanical vibration. The cutter head vibrations caused a cutting stylus to engrave a physical replica of the original sound wave into the record groove.

Electrical reproduction also required two transducers. The phonograph cartridge, or pickup, as it was originally called, traced the mechanical picture in the record groove and converted the resulting vibrations into electricity. Finally, the loudspeaker changed the electricity back into sound vibrations.

The second advance which occasioned recording and reproduction was the development of the vacuum tube amplifier. A microphone cannot provide enough voltage or current to feed a disc cutting head directly. The microphone signal requires considerable amplification to drive the cutting apparatus. Similarly, the phonograph pickup is equally incapable of driving a loudspeaker.

In both cases, the vacuum tube amplifier provided the means of increasing the voltage and current of the first transducer's electrical signal to a level sufficient to drive the second one. Today, solid-state amplifiers have generally replaced their tubed counterparts, although a survey of equipment reviews in *Stereophile* and *The Absolute Sound* indicates considerable interest in vacuum tube amplifiers in the high-end audio community.[1,2]

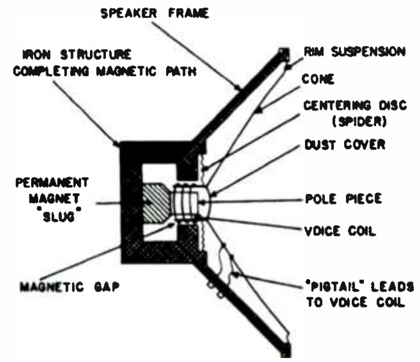

FIGURE 1: A side, cut-away view of the dynamic moving coil loudspeaker. (Figures courtesy of Dover Publications, *The Reproduction of Sound in High Fidelity and Stereo Phonographs*, by Edgar M. Villchur, 1965.)

MOVERS AND SHAKERS

Ironically, Joseph P. Maxfield and Henry C. Harrison of Western Electric, the inventors of electrical recording, were not the first proponents of electrical reproduction. The rights to their phonograph, which was acoustical, were sold exclusively to the Victor Talking Machine Company, which, in turn, marketed it as the Orthophonic Victrola. *Orthophonic Recording* was Victor's trade name for its new recordings.

Harrison and Maxfield believed that their new acoustic phonograph design would generate more than enough volume for home reproduction of electrical recordings.[3] The Orthophonic Victrola contained a sophisticated reproducer (for its time) capable of covering the wider frequency response of the new recordings. This reproducer was coupled to an exponential horn, a development which would have a significant impact on loudspeaker design as well (discussed in Part 2). By the late 1920s, radio receivers and all-electric phonographs with loudspeakers were readily available.

The basic principles of the moving coil electrodynamic loudspeaker, often simply referred to as the dynamic loudspeaker, were first patented in Germany in 1874 by Werner von Siemens.[4] In 1877—the same year Edison invented the acoustic phonograph—Lord John William Rayleigh described radiation theory for a circular diaphragm mounted in a baffle in his *Theory of Sound, Vol. 2.* In 1898 British physicist Oliver Lodge received a patent for a sound reproducer consisting of a moving coil attached to a wooden board. The modern dynamic loudspeaker is generally credited to C.W. Rice and E.W. Kellogg, who produced a working model in 1925.[5,6]

Figure 1 shows the essential elements of a modern moving coil loudspeaker, which has changed very little in its basic operation since Rice and Kellogg. A coil of wire, called the *voice coil*, is surrounded by a magnetic field and attached to a vibrating diaphragm, or *cone*. The voice coil and the cone are suspended by the *spider*, at the rear of the cone, and the flexible *rim suspension*, often called the *surround*, at the front. As electricity flows through the voice coil, it generates its own magnetic field, which interacts with the field of the permanent magnet, causing the cone to move back and forth.

During the early days of radio and electrical phonographs, single *full-range* loudspeakers, which attempted to reproduce the entire range of frequencies available on recordings and broadcasts, were commonplace. For the past 40 years, however, nearly all loudspeaker systems claiming high fidelity performance have used more than one driver to cover the entire audible frequency spectrum from 20Hz–20kHz.

The speaker enclosures described below are most commonly used with low-frequency loudspeakers, normally called *woofers*. Separate speakers named *midranges* and *tweeters* are normally used for the middle- and high-frequency portions of the spectrum. These individual loudspeaker components are called *drivers*. Prior to the widespread

An edited version of this article appeared in *The Encyclopedia of Recorded Sound in the United States*, published in 1993 by Garland Publishing, Guy Marco, editor. For this publication, I have made several revisions to my original manuscript.—GG.

use of multi-way loudspeaker systems (detailed in Part 2), some of the enclosures were used with full-range speakers.

ENCLOSED BOXES

A moving coil loudspeaker normally radiates sound from both the front and the rear, since the loudspeaker frame, or basket, is open in the back. When the loudspeaker cone moves forward, a compression is produced in front and a rarefaction is produced at the rear. Conversely, when the loudspeaker cone moves backward, a rarefaction occurs in front, along with a compression at the rear. Thus, the radiation from the front and the back of the cone are reversed in polarity with respect to each other (*Fig. 2*—this situation is sometimes described as "out of phase," though that terminology is technically incorrect).

At low frequencies the wavelengths are quite long with respect to the cone size. When the front and rear radiations meet, at the outer edge of the loudspeaker, cancellation occurs. This results in little low-frequency output from the loudspeaker, despite large cone movement in this region.

Since the first use of dynamic loudspeakers, it has been necessary to isolate the front radiation of the cone from the back radiation. The simplest method is to mount the loudspeaker on a large, flat *baffle (Fig. 2)*. The distance from the front of the speaker, around the baffle to the rear of the driver, determines the low-frequency cutoff point for the system. When the distance is less than one quarter wavelength of the frequency being reproduced, the output is greatly reduced.[7]

Many early radio receivers and phonographs used flat-baffled loudspeakers, as well as a variation on this technique—the *open back* enclosure. Early radios and electric phonographs were often housed in five-sided wooden boxes which contained both the electronics and the loudspeaker. The open back cabinets allowed sufficient distance between the front and rear of the driver, but with a much smaller front profile.

The open back enclosure normally suffers from cabinet resonance problems, which cause uneven low-frequency response. Generally, the deeper the cabinet, the more serious the resonance problems become. Olson summarized the low-frequency characteristics of a variety of open back cabinets in 1947.[8]

Prior to 1954, all loudspeakers had relatively stiff suspensions, which provided most of the control of the diaphragm motion. This was necessary with flat baffle and open back arrangements, where the rear of the cone radiated into free air. To achieve reasonably accurate low bass from a flat baffle, the baffle had to be unmanageably large. A 1929 article in the *Encyclopædia Britannica* described the "whole musical range of frequencies" as 25Hz–12kHz.[6] For the loudspeaker to respond to 25Hz, the baffle had to be 11', 4" square ... hardly a practical size.

An open back cabinet could be built with a much smaller front panel, but the cabinet resonances described above made it unsatisfactory for accurate low-frequency reproduction. Early loudspeaker designers soon realized that they needed fully enclosed boxes to achieve accurate bass with a front baffle of manageable size.

One consideration in designing an enclosed box loudspeaker is the rear radiation of the loudspeaker cone. There are two possibilities: (1) the rear radiation can be completely absorbed inside the enclosure; or (2) a portion of the rear radiation can emerge, through a port or vent, to reinforce the front radiation at low frequencies. This brief history of the major types of loudspeaker enclosures begins with the former possibility. Part 2 follows with a detailed look at vented enclosures.

A SPEAKER MILESTONE

The *infinite baffle* enclosure—popular through the mid-1950s—isolates the front radiation of the driver from the rear radiation at the lowest operating frequencies. Since it has never been possible to build a baffle of infinite size, such an enclosure was always approximated using a box of manageable proportions. One purist approach was to mount the driver on the wall of a room, in which case the room at the rear of the loudspeaker became the "enclosure," while the room at the front of the driver was the listening room. It was also possible to mount loudspeakers on closet doors. Several manufacturers built very large, stand-alone enclosures which approximated infinite baffles *(Fig. 3)*.

During the 1950s, Bozak was a leading proponent of such enclosures, exemplified by its B-310 system,[9] which contained four 12" woofers, eight tweeters, and two midranges in an enclosure with an internal volume of nearly 16 ft³ (36" wide × 19" deep × 52" high). Electro-Voice manufactured a version of the Patrician loudspeaker containing a single 30" woofer in an enclosure comparable in size to the Bozak.

In 1949, Harry F. Olson and J. Preston received a patent on the *air suspension* loudspeaker, which contained a woofer mounted in a small, sealed enclosure in which the air became part of the cone's suspension.[10] The RCA LC-1 system was the first commercial product based on the Olson and Preston patent.[11]

Olson's woofers were not substantially different from those used in large infinite baffles or open back enclosures. They had relatively stiff suspensions, with a resulting high free-air resonance frequency. When such drivers are placed in a small enclosure, the system resonance will be even higher than that of the driver in free air. This results in a loudspeaker system with relatively weak low bass, unless extremely large woofers are used.

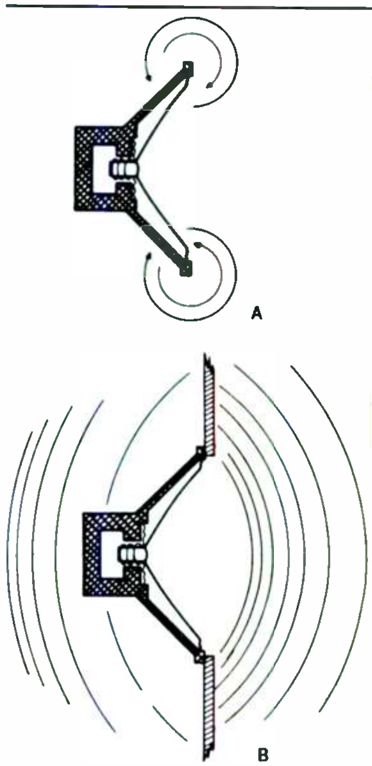

FIGURE 2: An unbaffled loudspeaker (A) showing cancellation of the low frequencies. Adding a baffle (B) isolates the front radiation from the back.

Perhaps the most significant development in the history of the high fidelity loudspeaker occurred in 1953 when American designer Edgar Villchur built the first *acoustic suspension* loudspeaker.[12] The results of his work were first published in *Audio Engineering* magazine in 1954, and a patent was awarded to him that same year.[13] The first high compliance woofer distinguished Villchur's loudspeaker from Olson's air suspension design. Villchur's woofer was a Western Electric 728B, a 12" driver which he modified by replacing the stiff outer suspension with a loose cloth surround. The new surround was made out of the ticking from an old mattress edge and sealed with butyl rubber to prevent air leaks.[11]

Modifications of existing drivers were hardly practical for production. The modified 728B became the model for the first high compliance woofer manufactured by Villchur's newly formed company, Acoustic Research, Inc. Villchur was assisted in the construction of his first acoustic suspension loudspeaker by his wife, Rosemary, whose contribution has often gone unnoticed, but should not be overlooked. Her experience as a draftsman during World War II proved important during the development of the driver suspension.[11,12] The invention of the high compliance woofer would later affect nearly every type of speaker system, and remains one of the most important contributions to the development of the high fidelity loudspeaker.

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COMPACT DESIGNS

Villchur's high compliance woofer had a very low free-air resonance frequency of less than 15Hz. When the woofer was placed in a sealed 1.7 ft^3 box, the low-frequency cutoff point became 38Hz.[14] The trapped air inside the small, airtight enclosure provided most of the support for the woofer cone, and served as a restoring force for it at frequencies near system resonance.

In an infinite baffle loudspeaker, as well as in Olson's air suspension designs, the compliance of the air inside the enclosure is greater than that of the driver's suspension. In an acoustic suspension loudspeaker the situation is reversed—the air compliance is less than that of the driver suspension *(Fig. 3)*. Prior to the acoustic suspension loudspeaker, a 38Hz low-frequency cutoff was unheard of in such a small enclosure.

The introduction of these small-box systems, known as *bookshelf* loudspeakers, played a major role in public acceptance of stereophonic recording. When "hi-fi" enthusiasts contemplated converting their monaural systems to stereo, adding a second 16 ft^3 enclosure to their living rooms was, in most cases, an unwelcome prospect. The compact size of a pair of acoustic suspension loudspeakers made their installation extremely practical.

One negative characteristic of acoustic suspension loudspeakers is their relative *insensitivity* (often referred to as *inefficiency*) compared to other designs. More amplifier power is required for a given playback level. As a result, a trend toward higher-powered amplification began in the late 1950s. In 1955, a 25W amplifier was considered a powerhouse. Today, amplifiers delivering ten times that power are commonplace.

AR's first commercial acoustic suspension system, the model AR-1, was introduced at the annual Audio Fair in New York in 1954 and hit the market the following year. Editor and engineer Roy Allison, who later became an employee of Acoustic Research and is now president of RDL Acoustics (after his prior company, Allison Acoustics, was purchased by an Asian firm), reviewed the product in the Oct. 1955 issue of *High Fidelity* magazine.[11,14,15]

The AR-1 was a two-way loudspeaker using a Western Electric 755A, 8″ full-range driver as a tweeter, along with Villchur's 12″, high compliance woofer. In 1955 AR intro-

Continued on page 64

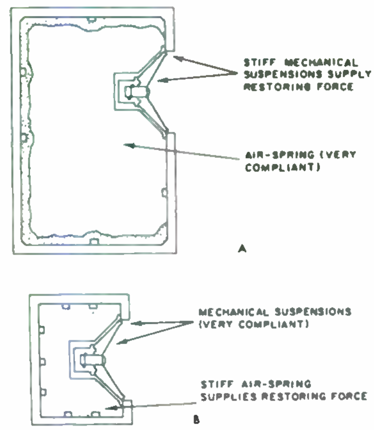

FIGURE 3: An infinite baffle loudspeaker enclosure (A) and an acoustic suspension system (B).

ACKNOWLEDGMENT

I wish to express my gratitude to C. Victor Campos for supplying much historical information on the acoustic suspension loudspeaker unavailable from any printed source. He also provided invaluable help as a proofreader during final editing of my original manuscript. Victor was a member of the technical departments at Acoustic Research and KLH during the 1960s and '70s. From 1986 to 1992 he was Director of Product Development at Adcom. He is currently Director of Product Development and Engineering at NAD.

ABSOLUTE SPL SENSITIVITY MEASURING WITH IMP

By Bill Waslo

When you measure a speaker's performance, your aim, most of the time, is to determine the frequency response. In general, you wish the speaker to respond uniformly to electrical inputs over selected frequencies. But that's easy to achieve: Just cut the speaker terminal wires—you'll get constant (in fact, zero) output at all frequencies.

Uh, wait a minute. Let's consider a few other speaker requirements, besides response uniformity. For example, you may wish to know not only the relative frequency response, but also the absolute sensitivity of the speaker. It's one thing to know that the response at 4kHz is 2dB below that at 1kHz; it's quite another to determine that at 4kHz the speaker generates 89dB sound pressure level (SPL) at 1m away when driven with 2.828V RMS. I'll discuss how to make such measurements using IMP and the version 2.0 software.

REFERENCES REQUIRED

Measuring absolute SPL requires care in maintaining reference levels. If gain or sensitivity variation in a measurement channel occurs, you'll need to account for it in the results. In addition, the test must be well-controlled, requiring you to specify or calibrate microphone distances and drive levels. When using the IMP/M system for sensitivity measurement, you must control several electrical variables.

First, you need to load a valid microphone correction file, not only to correct the response shape but also for the reference sensitivity value. In IMP ".DAT" microphone files, this information is usually in the first line, between quotes, and in units of mV/Pa (millivolts per pascal, referring to the value which corresponds to 0dB in the correction data).

Second, the analog probe and microphone level knobs on the front of your IMP represent potential unknown gain blocks in the test chain. They might be set to any position, and your poor blind computer will be unable to account for their effects—unless you tell it. So how is that done?

KNOB CALIBRATION

You need to calibrate the knobs on your IMP's front panel. IMP's internal fixed components have good precision, but the potentiometers are more mundane. The full-clockwise gain is known, but other settings need calibration.

You needn't send your IMP to the Bureau of Standards for calibration service. You already have an instrument for measuring relative levels—IMP itself—by which you can establish calibration markings around your level knobs.

First, align the knobs so that you can easily reestablish a repeatable full-clockwise knob position if you remove the knobs in the future. If you are using the special IMP case (from Old Colony or Liberty Instruments), turn the knobs clockwise (cw) to the stops, then loosen the set screws. Position the knob so that it aligns with the maximum cw index line on the front panel and retighten the set screws. If you made your own case, mark it with a permanent line to indicate where the knob points are when the control is at full cw. This point is "0dB," meaning no attenuation.

Configure your IMP as shown in *Fig. 1*. Note that probe2, not the usual probe1, is probing the amplifier output. You can leave a speaker connected if you wish. If you have the MLS option, go to the Acquire menu and configure for MLS(=on). If you do not have MLS, temporarily configure the [* Setup Averaging Impulse_mode] parameters for probe1, probe2, and MIC input to average only a single pulse response. [*In his IMP software, author Waslo follows the standard abbreviation for microphone.* SB *adopts an alternative style, mike, which we use throughout the article, except as it relates to program prompts and messages.—Ed.*]

Now set RATE to 61.2kHz, SIZE to 256 and INPUT to probe2. Set the probe level knob fully clockwise. With your power amp (or preamp) volume control at minimum, select [* Acquire Repeat]. Slowly turn up your amp's volume until about a half-window height MLS trace (or a near full-scale pulse) appears in the bottom IMP display.

Then press [Esc], and select [* Acquire Collect]. Transform the acquired data to frequency response via [* Transform Fft] and declare it as "cal" using [* Transform

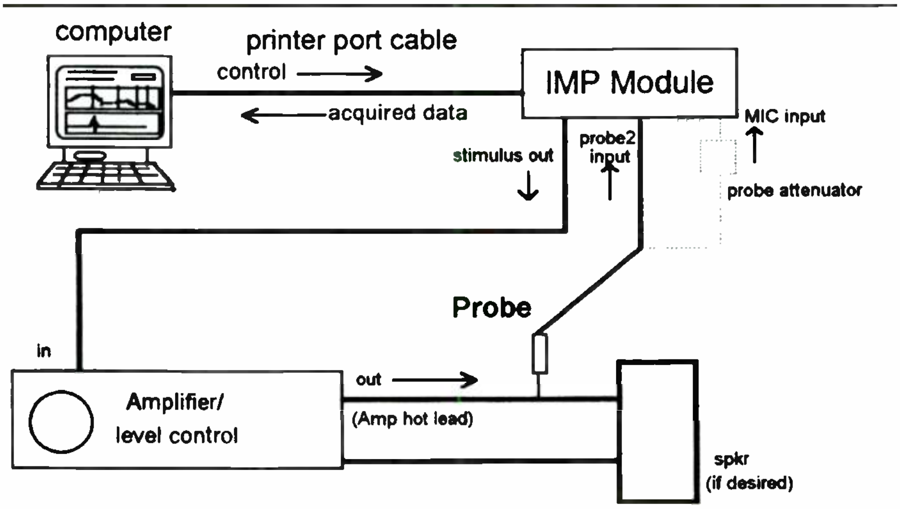

FIGURE 1: Setup for knob calibration.

Set_cal]. Select [* Transform Cal] to achieve a line at 0dB, which corresponds to the "0 dB" full-clockwise position of the probe level knob.

Put one of the markers near 1.5kHz. You will be adjusting the probe level knob for a series of specific decibel level readings, which you will read using the marker. You can do this most easily with the "cycling" feature in the auto_Measure menu, enabled by the menu sequence [* auto_Menu Setup cYcling On]. Then press [Esc] once and select [Cal_source]. Enter a "1" and press [Enter] to indicate that the existing "cal" data should be used.

TUNE-UP

Then press [Esc] once more and select [Elec]. The program creates a continuous series of probe measurements and compares them to the original, which you declared as "cal." Do not adjust your amp's volume control.

You will now see the line at 0dB being redrawn every few seconds. As you turn the probe level knob counterclockwise, the line being redrawn on the screen will move downwards. Near the position where the line is at –6dB you will also likely notice that the line shows some curvature near 14kHz, although it was dead straight at 0dB.

This is why you must use a marker near 1.5kHz to set up your knobs. The curvature is due to stray capacitive coupling on the IMP board (to the trace connecting Pin #3 of

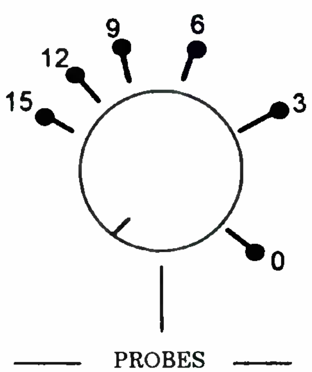

FIGURE 2: Knob with calibration markings.

U17, a high-impedance input to the op amp). The curvature will reach a maximum variation of about 0.8dB, the effect of which will be largely cancelled in typical usage when you calibrate your responses.

Adjust the knob so that the line drops to –3dB (you can read the marker to pinpoint it) and stays there. Indicate this knob position on your front panel using a pen or piece of tape to mark the reference point so you can return to it. Label the point "3dB" (leave off the negative sign) to denote the attenuation of that point relative to full clockwise.

Then repeat for readings of –6dB, –9dB, –12dB, and –15dB, again omitting the negative sign when you label the points. The region of your front panel around the probe level knob should resemble *Fig. 2*.

To calibrate the IMP's mike level knob, you can be lazy and just copy as well as possible the positions off the probe level knob (the controls should be nearly identical), or you can make a probe attenuator as shown in *Fig. 3* so that your probe can feed the mike input without overloading it.

If you choose the second method, then position the attenuator between the probe cable and the microphone input as shown and follow essentially the same procedure for the mike input and level knob as for the probe input and knob. Select the MIC option as INPUT, and instead of selecting [Auto_Measure Elec] for the cycling traces, choose [auto_Measure Acous], which uses the microphone input.

After you complete these steps, remember to turn off the "cyling" option under [auto_Measure Setup cYcling] and select "2" (get new cal data) under [auto_Measure Setup Cal_source].

REDUCING MIKE SENSITIVITY

Your IMP now features calibrated level controls. You'll notice that the spacing of the markings is very uneven; this is because IMP uses linear taper controls (so that time-domain trace amplitude tracks knob rotation), while your calibration marks are in decibels or logarithmic format. The markings become more crowded as the attenuation increases.

This could cause some problems if you operate the microphone input with the mike level turned down to more than 15dB attenuation, since your calibration marks would be too difficult to read. That might happen if you use a microphone, such as Mitey Mike, which includes a gain stage to boost the level from the bare capsule (IMP is designed for adequate sensitivity using its mike capsule alone). In such a case, it would be a good idea to make a microphone attenuator to reduce the microphone output so that the mike input knob will be operated toward the middle to upper end of its rotation, where the attenuation can be more accurately determined.

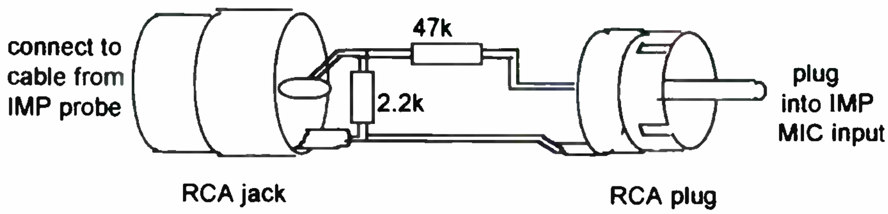

FIGURE 3: Attenuator to allow probe to feed microphone input.

You can make a suitable attenuator in the same way as the probe attenuator shown in *Fig. 3*. In fact, you can reuse the connectors from your probe attenuator, since it has already served its purpose. You only need a series resistor for the mike attenuator, but be sure to connect the grounds of the two connectors. Omit the shunt resistor (2.2k in *Fig. 3*). You can calculate the value of the required series resistor if you know the output impedance of your amplified microphone and how much you want to attenuate it, using the following formula:

$$R_A = 2210 \cdot (10^{\frac{dB}{20}} - 1) - R_{out}$$

Where R_A = value of series resistor to install in the attenuator
(Use the nearest 1 or 2% resistor value)
R_{OUT} = microphone output resistance
dB = attenuation (positive number) desired relative to the microphone's normal unloaded output level

For example, to attenuate Mitey Mike (output resistance = 150Ω) by 15dB to bring it close to the sensitivity of an IMP mike capsule, use a 10k resistor. Be sure to mark on your attenuator the attenuation value for future reference.

SENSITIVITY MEASUREMENT

Finally, the equipment is ready for sensitivity measurements. IMP/M calculates the sound pressure level which the microphone would detect if the speaker being measured was driven with 2.828V RMS (i.e., 1W into 8Ω).

IMP doesn't actually put sine waves of this voltage into the speaker; the sensitivity is calculated from the relative size of the signals picked up by the cal probe (input voltage) and the microphone (output level), with knowledge of the mike input attenua-

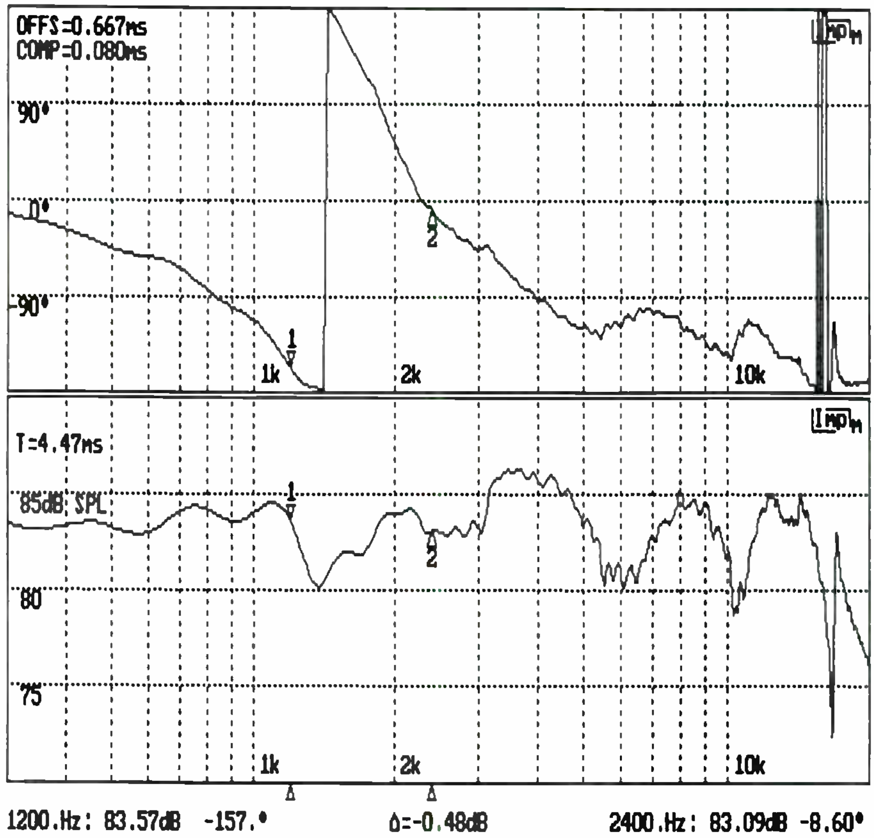

FIGURE 4: SPL sensitivity at 1m for a rather inefficient two-way system.

tion, probe input attenuation, and microphone sensitivity (from the mike correction file). Sensitivity decreases as the microphone is moved away from the speaker; the distance should therefore be controlled, usually to 1m.

Set up the test as you would normally conduct a frequency response measurement, using the 47.5kΩ Cal (#1) probe on the speaker "hot" input terminal and the microphone at the specified distance from the speaker's acoustic center. Set the input levels to avoid overload (the computer will beep if an input goes over range), also arranging to put the IMP level knobs on one of your knob calibration markings, preferably near the middle of the knob's travel. If you are using MLS, set up the [*setup Averaging Impulse_mode] parameters to the same number for probe1, probe2, and MIC.

Now, you need to communicate with IMP/M. You wish to change the frequency response display format to SPL mode, so execute [* Display Format Scale Spl oN]. Another menu of settings appears. Choose [Mic_attenuation] and enter the attenuation value of the mike level knob setting plus the attenuation of any fixed microphone attenuator you may be using (both positive numbers). For example, if the mike level knob is at 6dB attenuation and you have a 15dB fixed attenuator on the microphone, you would enter "21."

Then choose [Probe_attenuation] and enter the knob attenuation setting of the probe level knob. The value [Nominal] is where you can enter the SPL dB value you wish to assign to the first grid line from the top in frequency response displays (the 0dB line in non-SPL format). When you are done, choose [Ok] to activate the SPL format.

After that, it's easy. Just make the measurement, getting a new cal, and the display will be in absolute SPL format (see *Fig. 4*). If you alter the settings of IMP's level knobs, be sure to enter the changed values into the Spl menu. You can get there quicker with the shortcut keys [Shift-F4]. Frequency response data saved in SPL format will also be displayed in this format when it is retrieved from disk.

Another shortcut, useful sometimes if you want to avoid the knob calibration process described above: Only the *difference* between the knob readings actually matters. If both knobs are at the same position, you can just enter "0" for both attenuations in the Spl menu (or only the value of the fixed mike attenuator, if used, for the Mic_attenuation value).

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THE DAMPING FACTOR: ONE MORE TIME

By Donald Jenkins

In determining the damping factor (DF), which is usually defined as the ratio of the loudspeaker's impedance to the impedance of the source, some suggest adding the resistance (impedance) of the voice coil, while others even recommend adding the resistance of the speaker cables to the total source value. This article examines how realistic these assumptions may be, and what the value means, if anything, as a mathematical definition.

A COMPLEX SITUATION

In the classical treatment of damped harmonic motion, the damping factor is the coefficient relating to the curve that follows the peaks of the damped oscillation. In *Fig. 1*, the DF is the value e^{-at} where a is the damping coefficient, or constant, and t is the time for the value of the peak amplitude. In this article, I will use the value of a, the damping constant, as the comparative parameter to the loudspeaker/amplifier damping factor.

To adequately define the DF using the impedance method, you must at least know the speaker and the amplifier impedance. This presents two problems: 1) both of these quantities are complex variables, that is, there is both a resistive and reactive component to the impedance value; and 2) this value is first-order dependent on the frequency, and the amplifier produces a second-order effect with delivered power. The quotient of complex numbers, while having a mathematical solution, has a less than useful significance when used for the subject value.

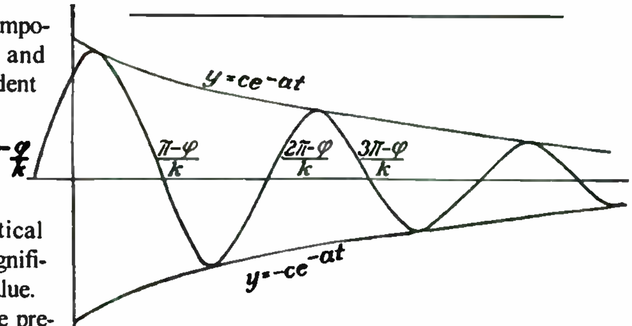


FIGURE 1: Damped harmonic motion.

I have never seen the DF value presented as a complex variable. Generally, the value is the ratio of the magnitude of the impedance, and, on occasion, the resistance of the voice coil is added to the amplifier's impedance and used in the denominator. This is an incorrect addition, since the two values are not the same type of number. Therefore, this value lacks a rigorous technical meaning.

STRICTLY EXPERIMENTAL DATA

The damping of the driver cone has three major contributors; 1) the acoustic damping, i.e., the air resistance on the moving cone, 2) the mechanical damping, i.e., the structural spring constant of the cone material and its attachment structure, and 3) the electrical damping due to the current generated by the moving coil whose energy is to be dissipated in the resistance of the circuit elements.

To evaluate the various effects, you should know at least the approximate value of each factor's contribution. Without a vacuum tank, 1 and 2 are difficult to separate, but they can be evaluated together at the local ambient pressure by putting tension on the cone and then releasing it to an open circuit. *Figure 2* is the result of such a test.

I extended the free-air cone with a direct current and then opened the switch. The curves shown are from an accelerometer mounted on the cone and the open circuit

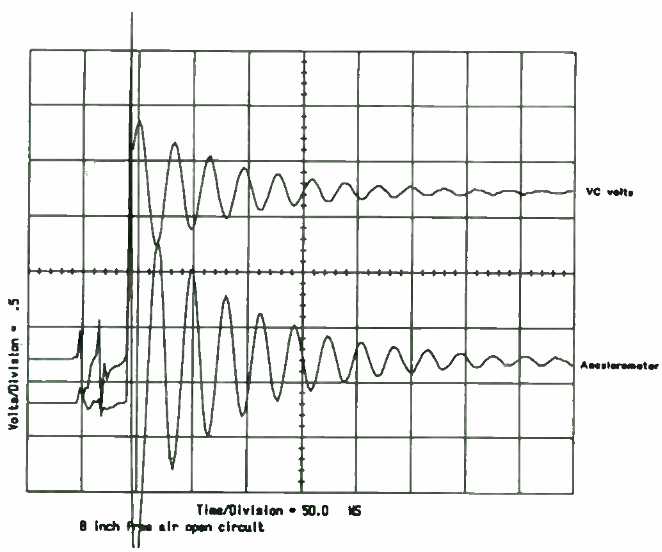


FIGURE 2: Free-air open circuit test.

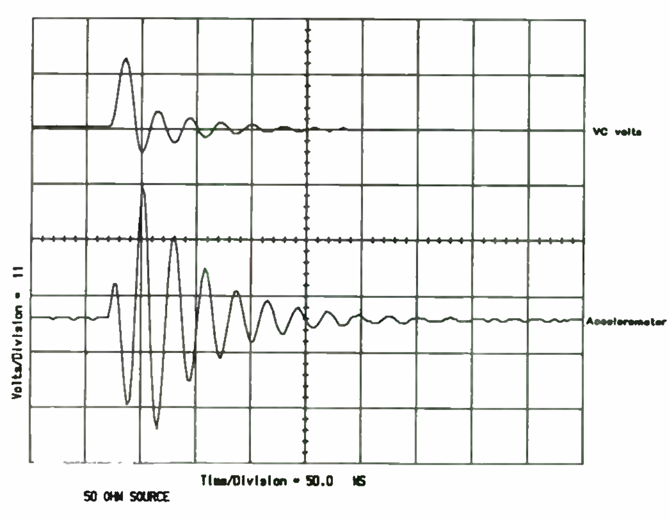


FIGURE 3: 50Ω impedance test.

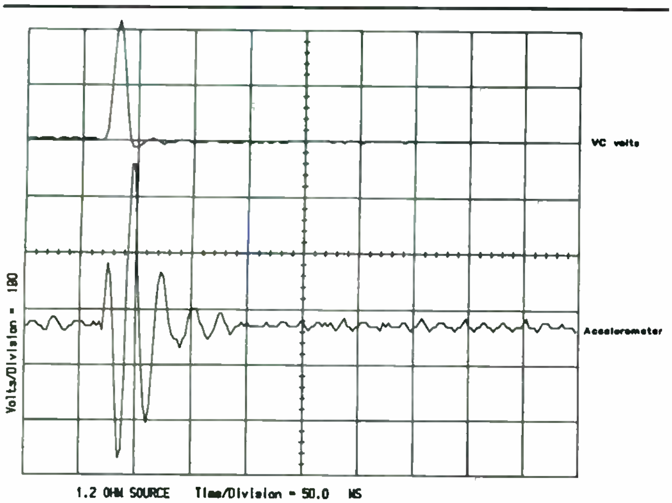


FIGURE 4: 1.2Ω impedance test.

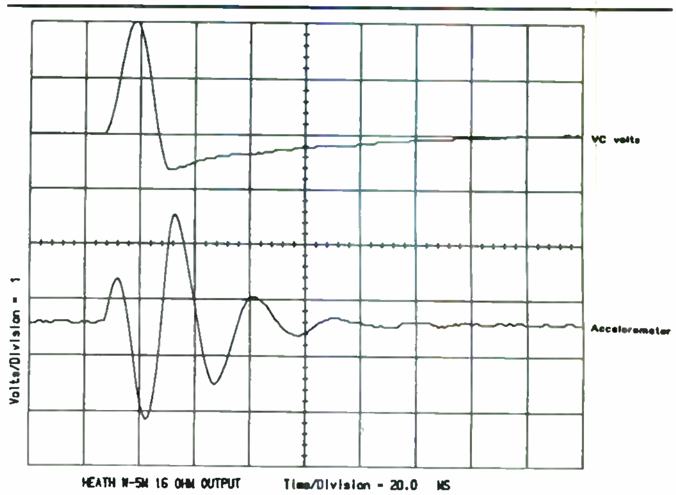


FIGURE 5: Heath W-5M 16Ω impedance test.

voice coil voltage. I calculated a 9.5 damping constant for this driver (an 8" diameter unit).

Figures 3, 4, and *5* are time records of a half cycle sine wave pulse, using the same 8" free-air mounted driver, but different source impedance values. *Figure 3* features a 50Ω generator resistance, *Fig. 4* has a 1.2Ω generator resistance, and *Fig. 5* shows the results of the driver connected to the 16Ω output of a Heath W-5M amplifier (transformer-coupled push-pull output).

These curves are for the voltage input to the voice coil—with the continuing record after the input pulse has terminated—and from an accelerometer mounted to the driver cone. As you'll see later, the cone motion is the only valid way to measure damping under all conditions. Both the voltage and current characteristics both have major deficiencies under certain conditions.

MORE VALUES

From these records you can calculate the values of peak amplitudes and time. Use the equation of the damped asymptotic curve $y = ce^{-at}$ of *Fig. 1*, together with a little arithmetic, to determine the value of a (damping constant) which equals ln (last peak amplitude/first peak amplitude)/time between peaks. *Table 1* lists the results from these measurements and calculations.

The DF value in the Heath manual for this amplifier is 40, which closely agrees with the calculated value of 51.4. How Heath calculated this value from the other published amplifier data, however, remains a mystery.

I determined the free-air resonance of this 8" driver to be about 36.4Hz. If we calculate a DF from the formula of impedance ratios, we should determine the driver and amplifier impedance values at this frequency.

Figure 6 shows a fairly flat driver impedance over the range of 30–40Hz. The value at 36Hz is (7 + j6.6) for a magnitude of 9.6Ω. *Figure 7* shows an amplifier value of (0.89 + j2.95) for a magnitude of 3.08Ω at 36.4Hz. You arrive at this value by holding the amplifier input constant and varying the load resistance (nonreactive) over a range to achieve a maximum power to the load. This maximum power point is defined as the amplifier output impedance for those conditions.

Using these values determined by the driver and amplifier measurements, you would get a DF of 0.92 if $DF = Zls/(Z_{OUT} + R_{VC})$. If you used the complex values for the same parameters, the DF would be (1.04 + j0.41). Obviously, neither of these values is correct or very useful.

For this amplifier the 16Ω output winding has a resistance of 0.89Ω. Using this value as the Z_{OUT}, not including the voice coil resistance, and using the 9.6 impedance magnitude for the driver, you can still only calculate a DF of 10.8. This value, too, is obviously incorrect and meaningless.

OUT, OUT DAMP SPOT

How then to determine the damping constant? *Figures 8* and *9* are open circuit voice coil voltage records for the Sony and Heath

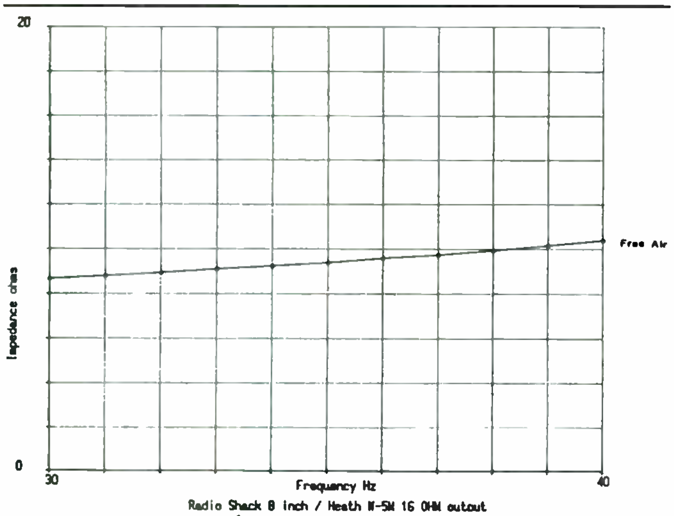


FIGURE 6: Driver impedance.

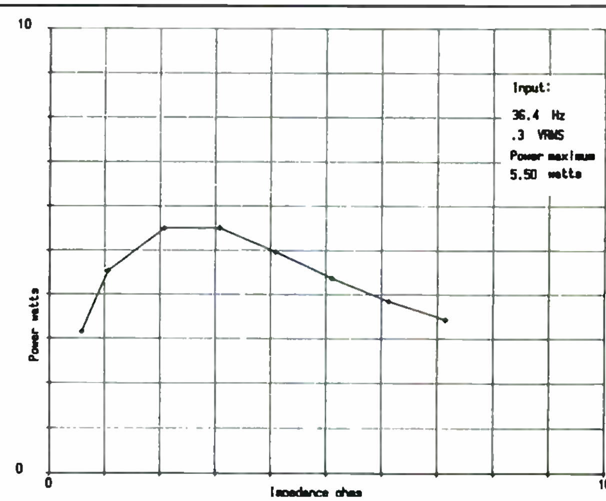


FIGURE 7: Amplifier output impedance.

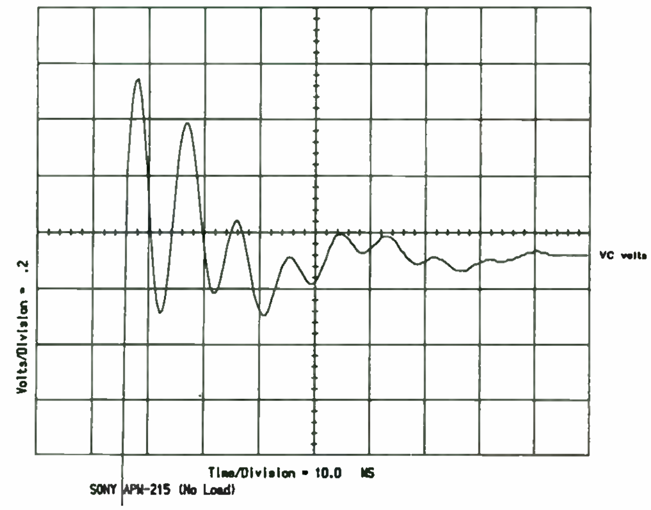


FIGURE 8: Open circuit voice coil measurement for Sony speakers.

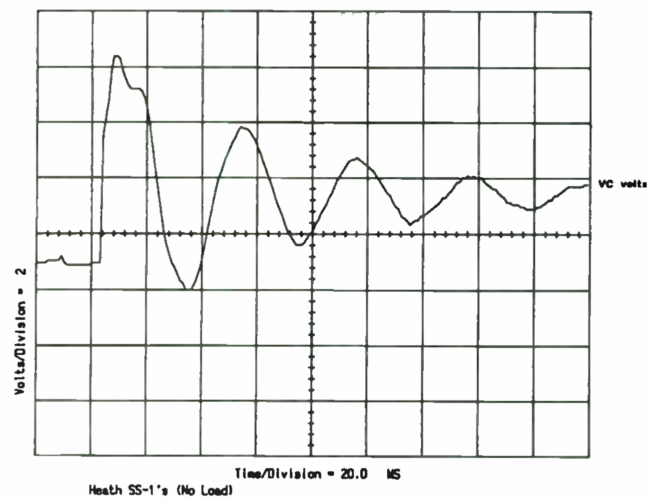


FIGURE 9: Open circuit voice coil results for Heath speakers.

speakers, respectively, and both systems contain crossover networks. The Heath system has four drivers in two cabinets, while the Sony system has three drivers in a single cabinet. From this data you can determine the acoustic and mechanical damping constants—21.4 for the Heath system and 147 for the Sony system.

Whatever the electrical damping contribution in the Sony system turns out to be, it will not make much difference in the final sound, because any system that damps to less than 25% amplitude in the first cycle has, in effect, "no free oscillation." The Heath unit, however, does have room for improvement.

Figures 10 and *11* are the same Heath and Sony systems pulsed using a Heath W-5M amplifier, and an attempt to evaluate the effect of the amplifier impedance on damping. The curves are voice coil voltage and current (no accelerometer mount possible). One fact becomes clear: it is not possible to determine the value of the damping constant by using circuit voltage and current relationships.

TABLE 1

PEAK AMPLITUDE/TIME CALCULATIONS

Figure No.	peak1	peak 2	time	a	source Z
2	44	3.0	0.2813	9.5	OPEN
3	30	3.5	0.1388	15.5	50
4	18.5	4.0	0.0525	29.2	1.2
5	39	9.5	0.0275	51.4	W-5M

THE ELUSIVE DF

Compare *Fig. 12*, which graphs the free-air 8" speaker with the accelerometer, with *Figs. 10* and *11*. The current trace does not provide enough data, nor is it the type needed to make a damping constant calculation. The current records are similar for the three speakers, but none shows the damped sinusoidal characteristic.

A damping constant is useful only for a transformer-driven speaker when the evaluated system is complete, fully connected, and installed. You must also use a pulse input and calculate a damping constant using the classical damped harmonic motion definitions. Voltage and current records do not provide adequate data to calculate a meaningful damping constant. Driver cone displacement is the only valid parameter you can use when calculating the damping constant for fully connected systems. You can evaluate drivers mounted in enclosures

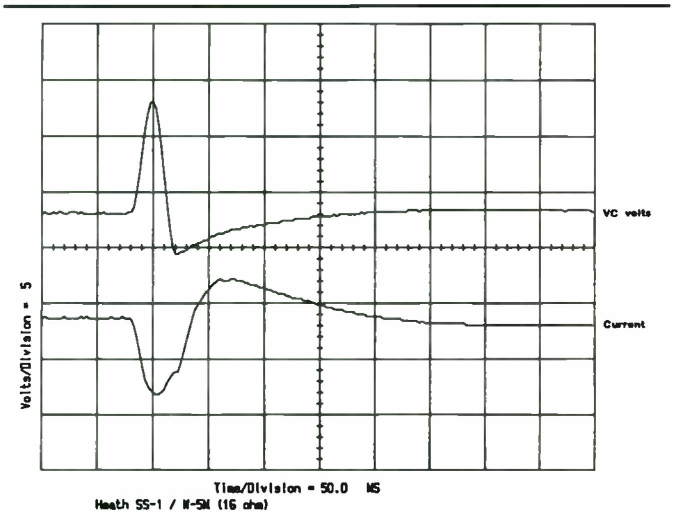


FIGURE 10: Amplifier impedance effect on Heath speaker.

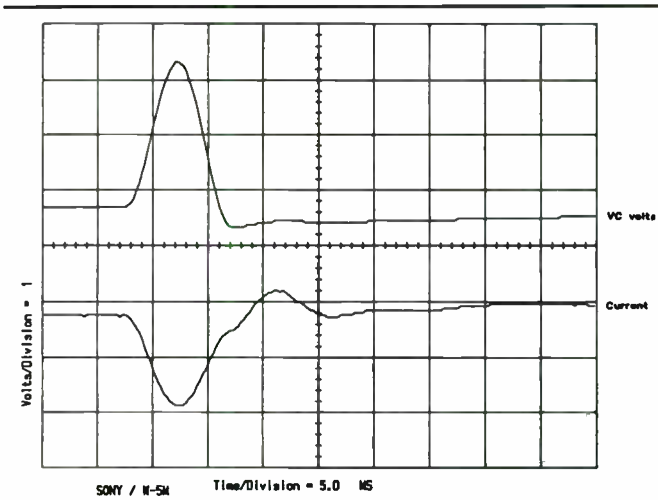


FIGURE 11: Effect of W-5M on Sony speaker.

for the acoustic and mechanical damping constant by using the preload and release technique and voice coil voltage.

To calculate the DF of the integrated system, you must use some technique to measure the cone displacement. For multidriver enclosures, or for speakers driven from the same amplifier/output transformer, each speaker may, in fact, have its own DF.

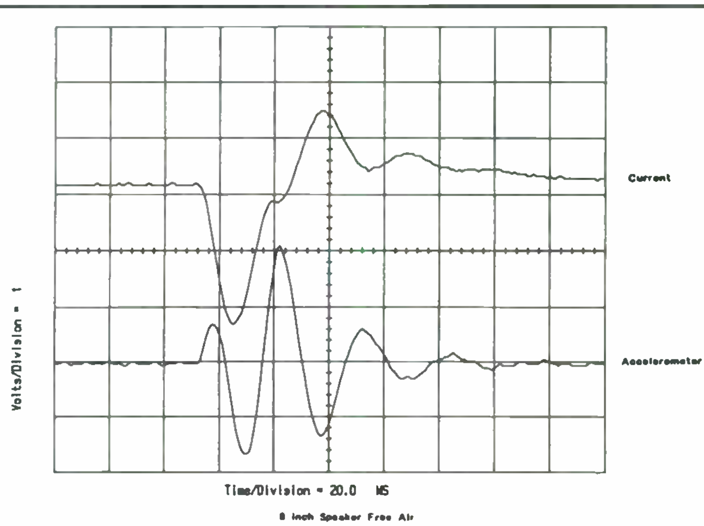


FIGURE 12: Eight-inch driver open air test.

A MIKE TECHNIQUE

The problems in determining cone displacement for enclosure-mounted drivers are obvious. One of the easier applied methods—other than cone-mounted accelerometers or displacement transducers—is to measure the direct acoustic output from the speaker using an acoustic-coupled transducer. A low frequency response microphone is the obvious choice.

Using a good-quality sound level meter (SLM)—one that has an AC output for the transducer—is also an option. The problem is that most SLM units perform poorly or unreliably below about 30Hz. My EXTECH Model 40770—one of the better meters of its kind—registers 3dB down at 31.5Hz and very distorted AC output at about 20Hz. Since most of the speaker enclosures of interest will be at resonance below 30Hz, you can rule out SLMs.

I have had somewhat better results using a microphone (in this case a Sony Model F-99S). With this microphone the response below 30Hz is still not too good, but with careful input pulsing you can obtain usable data.

Using the microphone technique with the 8" free-air driver and the Sony APM-215 system—both driven by the Heath W-5M amplifier 16Ω output—I determined overall system DFs to be 198 to 229 for the Sony unit and 23 to 29 for the 8" free-air driver.

Using cone displacement, I measured the DF for the 8" speaker at 51.4. The Sony unit's DF was 147, as measured with the open circuit voice coil method. For the 8" driver, the resonant frequency was 36Hz. The Sony microphone response at this frequency is also distorted, and you'll need some interpolation to "eyeball" the peak amplitudes for input to the DF equation. On the other hand, the Sony speaker resonant frequency is 110Hz, at which frequency the microphone output is very repeatable, and the peak amplitudes are much easier to determine.

With a well-matched and damped amplifier, the system's DF should always be greater than the open circuit voice coil value. This holds true for the Sony unit (147 to >200).

High-performance speaker designers, however, aspire to improve the response at the lower frequencies. For this environment (the low frequency enclosures) the open circuit test is better than no test, but the cone displacement method is really the only valid way to determine the true DF. If you are fortunate enough to have a good low frequency microphone, or some other type of a low frequency response acoustically coupled transducer, then the acoustic coupling method should also give good results.

J MARKS THE SPOT

From the above damping analysis, I propose a new driver and enclosure standard for a

Continued on page 64

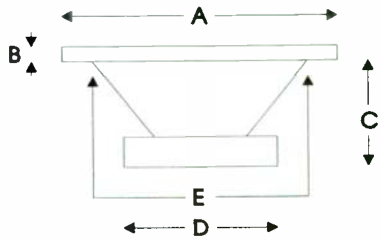
seas

Audiophile Price List
June 1994

Madisound Speaker Components
(8608 University Green)
P.O. Box 44283
Madison, WI 53744-4283 U.S.A
Voice: 608-831-3433
Fax: 608-831-3771

Model	Size Description	Imp. Ω	Fs Hz	Qts	Vas Liters	X-Max mm Peak	Power Watts	Rating db @Freq	Sens. 1W/1M db	Cost Per Unit
Tweeters										
19TFF (H586)	19mm textile dome tweeter	8	1700				80	12@5K	87	$15.50
25TAF/G (H398)	25mm aluminum dome tweeter with grill	8	1400				100	12@3.5K	90	$19.60
25TAC/G (H400)	25mm aluminum dome tweeter with chambered back & grill	6	660				55	12@3K	91	$23.50
Midranges										
MCA11RC (H143)	4.5" treated paper cone mid with cast magnesium basket	8	140	.72	1.3	.9	110	6@800	89	$30.85
K2852	Chamber for MCA11RC	-	-	-	-	-	-	-	-	$1.50
CA11RCY (H149)	4.5" treated paper cone bass/mid with cast magnesium and rubber surround	8	58	.24	5.4	3	60		86	$34.80
Woofers										
P11RC (H454)	4.5" poly cone woofer with cast magnesium basket and rubber surround	8	55	.34	5.3	3	60		84.5	$30.25
P14RC (H395)	5" poly cone woofer with cast magnesium frame and rubber surround	8	40	.28	18.9	3	60		89	$29.65
P17REX (H416)	6.5" poly cone woofer with cast magnesium frame, rubber surround, large VC	8	34	.24	30.5	3	80		89	$42.20
P17RE (H419)	6.5" poly cone woofer with cast magnesium frame and rubber surround.	8	34	.33	30.5	3	80		87.5	$39.00
P17RC (H353)	6.5" poly cone rubber surround, cast magnesium frame for sealed enclosures.	8	35	.32	40.8	3	60		89	$32.50
P21REX (H282)	8" poly cone rubber surround, cast magnesium frame woofer with 1.5" VC	8	33	.37	68.9	3	80		91	$48.50
P21RF/P (H511)	As above with phase plug and 2" diameter voice coil	8	34	.34	48.3	4	125		88	$54.40
P21RE4X/DC (H442)	Dual VC 8" woofer with poly cone, rubber surround, cast magnesium.	8/8	31	.30	66.4	3	90		90	$55.30
P25REX (H283)	10" poly cone woofer with rubber surround and cast magnesium frame.	8	27	.44	156.8	3	80		93	$54.30
CA25RE4X/DC (H372)	10" treated paper cone woofer with rubber surround, cast magnesium frame	8/8	25	.31	187.9	4	90		91	$59.00
Coincidental Coaxial Driver										
P17REX COAX/F (H489)	6.5" coaxial P17REX/25TFFN/G in a coincidental configuration.	T6 W8	T1.8K W35	W .25	W 26.9	W 3	T90 W100		T89 W89	$82.00

Unit	A mm	B mm	C mm	D mm	E mm	Vented Box QB3 alignment					Sealed Box	
						V_b liter	F_b Hz	F_3 Hz	Vent Ø"	Vent L"	V_b liter	F_3 Hz
19TFF (H586)	93.8	3.1	18.4	66.5	66.5							
25TAF/G (H398)	103.8	3.6	24.4	74.8	74.8							
25TAC/G (H400)	103.8	3.6	37.9	74.8	74.8	No Rg values are used in these alignments.						
MCA11RC (H143)	109.4	4.4	49.1	72	95.8						3 to 5	145
CA11RCY (H149)	109.4	4.4	55.1	93	95.8	1	59	135	1	5.7	1 to 2	175
P11RC (H454)	109.4	4.4	49.1	72	95.8	3	64.4	75	1	4.6	1.7	114
P14RC (H395)	133.2	3.7	62.3	72	112.8	5.5	56	70	1	2.7	3.7	100
P17REX (H416)	170.4	3.8	67.7	110	145.3	6	55	70	1.5	6.2	4.3	97
P17RE (H419)	170.4	3.8	67.7	93	145.3	13	42	50	1.5	4.8	8.6	73
P17RC (H353)	170.4	3.8	64.7	72	145.3	15.7	44.6	52	2	6.2	11	78
P21REX (H282)	215.4	4.7	75.3	110	186.8	46	36	39	2	2.6	27	63
P21RF/P (H511)	215.4	4.7	75.3	110	186.8	22.4	41	45	2	4.9	14.5	71
P21RE4X/DC (H442)	215.4	4.7	75.3	110	186.8	25.6	39.5	45	2	4.5	16	70
P25REX (H283)	261.2	4.2	82.8	110	229						70	44
CA25RE4X/DC (H372)	261.2	4.2	82.8	110	229	72	32	38	3	5.2	46	57
P17REX COAX/F (H489)	170.4	3.8	67.7	110	145.3	5.5	57	70	1.5	6.7	4	100



Call for more specific alignments and assistance with your application. Our staff is very experienced and most helpful.

Reader Service #1

CLIFFNOTES FOR LOUDSPEAKER UNIVERSITY

By Richard A. Honeycutt

I remember well when I first fell in love with speakers. It was about 1960, and my dad bought a 1940s-vintage Arvin radio at an auction. A couple of dollars' worth of tubes put it back into operation, but the speaker cone was torn, so it buzzed badly. We applied the standard technology of the day—Duco cement—and the radio soon played beautifully, or at least several grades above our much newer GE table radio. Encouraged by this success (and perhaps even more by lack of funds to buy decent speakers), I followed the path so common to all *SB* readers: I built a cabinet from plans in *Popular Electronics*, then designed my own, and eventually began experimenting with the speaker driver units themselves.

SCHOOL FOR PROS

Ten years after my first encounter with speakers, I had a degree in physics, and I knew that a scientist's approach is to locate textbooks and/or papers which discuss theory and practice, plug the desired parameters into equations, juggle a little, and then apply the results that pop out. After another four years, I concluded that my collection of those classic papers and some of the standard textbooks in acoustics and electroacoustics was considerable, but I was nowhere near having the speaker builder recipes that I had expected to find. I have since concluded that they do not exist.

What does exist are practitioners of the art with much very specialized knowledge about particular aspects of speaker technology. Very few can claim to know much outside their specialty. And no college program (in this country, at least) exists to bring these diverse bits of lore together, or provide initiation into the cult for new devotees.

For these reasons, Mike Klasco of Menlo Scientific organized the Loudspeaker University, a three-day intensive symposium on distortion reduction and improved power handling in loudspeakers. Designed for professionals in all areas of the speaker industry, the event was held in Lowell, MA, on May 12–15.

ABOUT THE AUTHOR

Richard Honeycutt teaches electronics and consults in acoustics, speaker design, and church sound system design. He has written for various magazines on electronics, speakers, and guitars, and has authored textbooks on op amps and electromechanical devices. Mr. Honeycutt lives with his wife Betty Jane and three children in Lexington, NC.

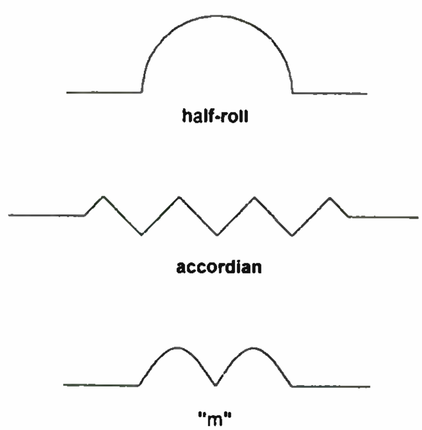


FIGURE 1: Surround profiles.

As you'd expect from a symposium designed for industry, many presentations held little interest for hobbyists. For example, the difference between Taiwanese and American methods for forming steel probably is not an issue of burning interest for most *SB* readers. But the discussions presented much information that bears upon likely near-future improvements in the speakers we will buy for our projects.

HATS OFF!

Speaker cones have been studied off and on since about 1925, when Rice and Kellogg invented the modern cone speaker. Traditionally, cones were made of either paper sheet stock, which was steam-and-pressure formed and seamed, or of felted paper. Seamed cones, the oldest technology, are still used in low-cost speakers and in some musical-instrument speakers. Today most cones used in stereo, pro-sound, and autosound speakers are formed by felting.

The felting process begins when a soup of water and paper pulp is passed through a screen basket shaped like the desired cone, and a layer of wet pulp is caught in the bowl of the screen. A heated mold shaped like the inside of the cone is pressed into the basket, squishing some of the water out and partially drying the pulp, which adheres to the mold. Another mold is now used to form the outside while applying more heat and pressure.

By carefully controlling the amount of pulp, the temperature, and the pressure, the manufacturer can make a variety of cone types, from very thin, hard ones used for high-frequency drivers to soft, thick, heavy ones for subwoofers. These techniques were well-known before speakers were invented; in fact, the first manufacturers of speaker cones were actually felt-hat manufacturers!

IMPROVE YOUR SURROUNDINGS

So, what's new in paper cones? Mainly surround materials. In the early seamed cones the paper was simply formed at the outside edge to give some flexibility. To reduce cone flexing, a technique was developed for grinding the edge after forming to increase its flexibility. Felted cones could simply be molded with an appropriate edge.

However, paper surrounds have limited compliance, making low resonance frequencies difficult to attain. Even worse, they do not absorb sound energy that travels through the cone, so it reflects from the point of attachment to the basket, setting up standing waves in the cone, and resulting in peaks and dips in the frequency response. An ideal surround allows very free axial (in-and-out) cone movement and almost no radial (side-to-side) movement. It would not be so floppy as to flap on its own, and would have enough internal friction to absorb sound waves entering from the cone.

Cloth surrounds have been used for many years, and are available in three basic shapes (*Fig. 1*). The half-roll works fairly well as long as the speaker excursion is limited. But for large excursions, the cloth tends to wrinkle and create "cricketing" clicks or even pull the cone off-center.

The accordion surround permits free motion and absorbs sound well, but sometimes flaps too freely, especially at frequencies between about 500Hz and 1.5kHz (depending upon speaker size), where the primary surround resonance occurs. When this happens, the surround tends to move out of phase with the cone, producing a serious dip in response at that frequency. Also, at low frequencies large back pressures in the speaker cabinet can make the surround flex so freely as to cause buzzing as the surround reaches its elastic limit. The "m" surround shape avoids both problems fairly well.

To further improve surround performance, both butyl rubber and plastic foam surrounds have been used. Butyl rubber types, though difficult to glue, perform quite well. However, they are relatively heavy, reducing speaker efficiency, and they are also expensive, reducing wallet size. Foam is lighter and cheaper; however, it deteriorates. Treating foam with a plastic skin such as polyvinyl acetate (PVA) helps to slow down the rotting process.

Two new cone materials deal with these problems. Santoprene® is basically rubber encased in a polypropylene skin and treated with an adhesive primer for easy gluing. It looks much like butyl rubber, with many of the same advantages for somewhat lower cost. The Hawley edge (named after the cone maker who pioneered it) is a molded-in-place, self-skinning polyether foam resistant to ozone and chemical attack. Because of the skin, it avoids the air-leakage problem common with standard foam surrounds, and is only slightly more expensive than ordinary foam.

Of course, speaker manufacturers are unlikely to use Hawley's proprietary name in their advertising. But if we carefully read speaker descriptions in ads, maybe we can detect when one of these improved surrounds is being used. Most of us would be willing to pay a little extra for improved performance and longer speaker life.

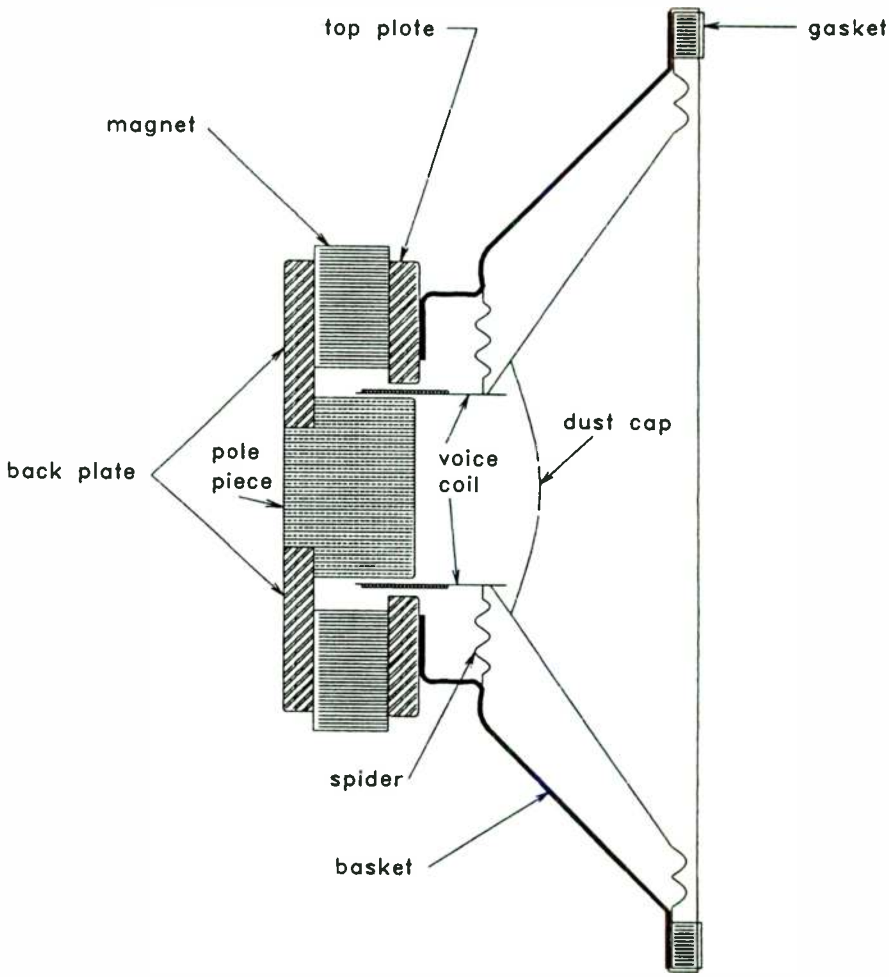


FIGURE 2: Cross section of a speaker.

PAPER OR PLASTIC?

Not all cones are made of paper. Mike Klasco related an incident in which paper cones, subjected to extreme humidity, experienced a 10% resonance decrease. The effects are less dramatic under normal humidity levels. And paper cones are treated with materials that reduce their sensitivity to humidity. Also, cones that are stiffer than paper have flat response to higher frequencies, and cones with more internal friction will suffer less from break-up modes that cause peaky response.

Numerous efforts have been made over the years to improve upon paper. The earliest non-paper commercial cones used a honeycomb aluminum shell skinned with very thin aluminum. The response flatness was excellent, but the efficiency was as low as the cost was high. Paper/plastic composites, including the current "wet-look" cone, have higher internal friction, but are still not entirely immune from humidity, and are no stiffer than paper cones.

Pure plastic cones, the most common of which are polypropylene, have been tried. Although it has a high internal resistance, polypropylene is not very stiff, and can exhibit undesirable flexing under certain conditions. Also, it has a low melting temperature, so it will sometimes literally melt itself loose from the voice coil under sustained high-power operation. Finally, it is very difficult to bond, and some commercially available poly-coned speakers will literally pull the cone from the voice coil or surround with a powerful impulse signal. At least one speaker manufacturer actually sews the surround to his poly cones.

Details show flux in the gap. The curved lines represent fringing.

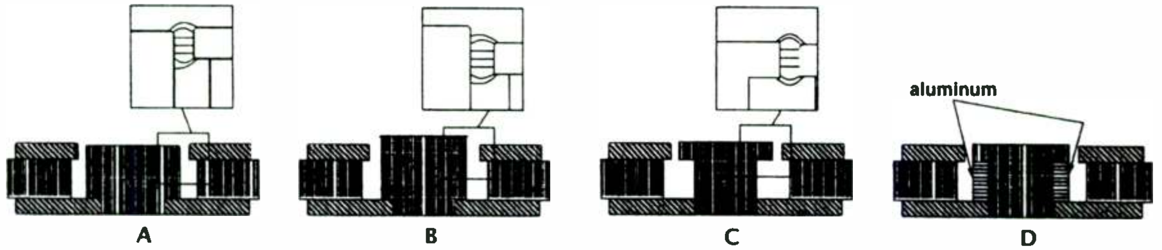


FIGURE 3: Magnetic system geometry.

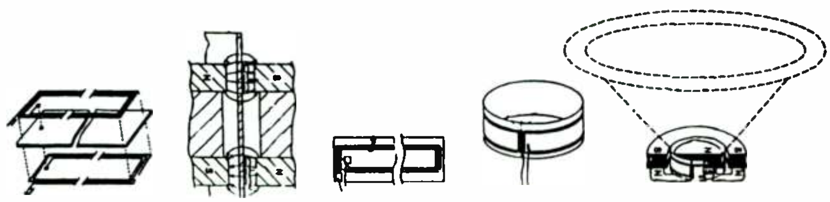


FIGURE 4: Linaeum printed-circuit voice coil in double-gap structure.

So the search continues. DuPont is experimenting with cones made of Nomex® paper and Kapton® plastic. In addition to excellent high-temperature resistance, Nomex, which is used in firemen's gloves, has a high internal friction and a very good stiffness-to-mass ratio. Also, it can be formed into cones more cheaply than paper pulp. However, the initial samples shown at the symposium need further development in strength and consistency before they will even be ready to test in actual speakers.

Kapton is a very high-temperature plastic with the same advantages as Nomex. Though no response curves were shown for speakers with Kapton cones, samples seemed quite promising, and in fact some of them are currently being manufactured. At present, their main use is in weather- and chemical-resistant speakers, but they may find their way into home sound systems and auto speakers before long. Because of its superior strength, Kapton can be made into thinner and lighter cones than can polypropylene.

The other major development area in cone technology involves composite cones, usually made of a super-fiber (fiberglass, Kevlar® [an aramid fiber related to Nomex], and carbon fibers) bonded by plastic resin. Paper cones reinforced with carbon fibers have been available for some time, and perform very well. Carbon fiber has one of the highest stiffness-to-weight ratios of any material known, so carbon cones perform much like ideal pistons (i.e., have a flat response) over a considerably broader frequency range than do other cones. Fiberglass and Kevlar composite cones offer similar advantages, and several companies already manufacture speakers using each of these materials. For a given design, the fiber you choose depends upon several factors.

Bonding plastic resins to the reinforcing fibers is one difficulty in manufacturing composite cones. This problem is being addressed by electrostatically powder-coating the resin onto the fibers, then baking the resulting product. As this technique becomes better known, we will likely see significant improvements in fiber-reinforced cones.

ARACHNOPHOBIA

The other suspension member (besides the surround) in most speakers is called the spider. *Figure 2* shows a cross section of a speaker, with the various components identified. Some of the early acoustics texts referred to the spider as the rear suspension. (I often wondered why it was dubbed the spider, but when I saw my first late-'20s-vintage Atwater-Kent speaker, I understood. Its rear suspension was corrugated paperboard cut in a shape that would likely send the squeamish screaming and scurrying!)

Almost all spiders are some combination of cotton, polyester, and linen cloth, treated with phenolic and heat-pressed into shape. Spiders can be either flat or cupped in profile. The flat ones are preferred for low distortion, since the cupped ones flex asymmetrically, introducing even-order distortion products.

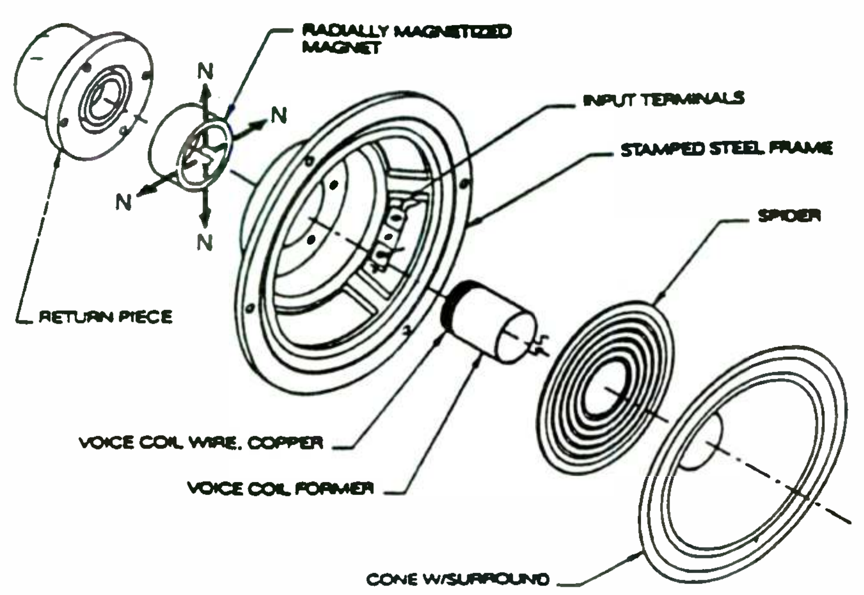


FIGURE 5: Aurasound structure.

Spiders contribute to the background noise produced by a speaker in a way not often appreciated. As the cone moves back and forth, air is pumped through the tiny openings between the spider's phenolic-coated fibers. The resulting air-rush noise does not produce what we would usually term distortion, but it does add to the background noise, and so reduces speaker clarity.

At the 1988 AES convention Dean Jensen and Gary Sokolich of Jensen Transformers introduced a method of measuring the undesired sounds (technically called spectral contamination) produced by any audio component. This method has only recently been applied to speaker testing, but has identified air-flow noise through spiders as one of the culprits. Punching holes in the spider reduces this noise, but can allow dirt particles to enter the voice-coil gap.

Some thermoformed Kapton spiders were shown at Loudspeaker U. (Thermoforming is a manufacturing process used to heat-press plastic materials into a final shape.) While easier to manufacture to close tolerances, and less prone than cloth spiders to change characteristics over time, these spiders require vents, since they permit no air to flow through.

DIFFERENT MAGNETS

The magnetic system shown in *Fig. 2* (and repeated in *Fig. 3A*) is by far the most common used in speakers today. However, several variations on the theme exist, as shown in *Fig. 3*. In *Fig. 3B*, the pole piece extends above the edge of the top plate so that the magnetic fringing flux is more symmetrical, reducing even-order harmonic distortion.

The undercut pole piece shown in *Fig. 3C* achieves the same results by different means. This method, however, moves the iron far-

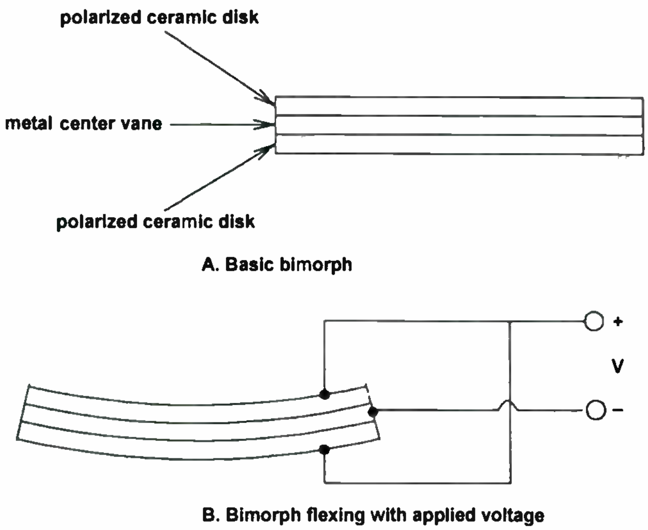


FIGURE 6: Piezoelectric bimorphs.

ther away from the coil, making heat transfer from the coil to the iron more difficult. The scheme shown in *Fig. 3D* uses aluminum to fill the opening created by undercutting the pole piece, restoring a thermal path, and keeping the coil cooler.

In addition to these variations, some engineers at the symposium admitted that they are beginning to use neodymium magnets even for woofers. Neodymium, which can support a very dense magnetic field, was first used in microphones, and soon after, to compression drivers. Only now is it beginning to be used in woofers.

For example, the magnet on a typical pro-sound woofer with a 2″ or 2½″ voice coil might have a 54 oz barium-ferrite (ceramic) magnet measuring about 5 or 6″ square by 5/8″ thick. This can be replaced by a piece of neodymium about the size of a silver dollar. Needless to say, this makes portable speaker systems much easier to carry around.

Two companies have introduced significantly different magnetic systems that reduce distortion and increase efficiency when compared to the best traditional designs. Linaeum uses a voice coil made of flexible printed-circuit material, suspended in a double-gap structure as shown in *Fig. 4*. This design gives about a 3–5dB increase in efficiency, and reduces distortion to as little as one-tenth that of the more common systems.

Aurasound uses a "pot" structure reminiscent of the old Alnico structures, but with a radially charged neodymium magnet instead (*Fig. 5*). This structure spreads a very uniform magnetic field over a large area, allowing the voice coil winding to be shorter than the magnetic field (underhung coil) rather than longer (overhung coil) as is usually done. If the voice coil winding equalled the height of the gap, when any cone motion occurred, part of the coil would be out of the gap, changing the effective force on the cone. The result would be serious distortion.

The overhung coil avoids this problem by having part of the coil—ideally the same percentage—out of the gap at all times. The underhung coil avoids the problem by never letting any of the coil out of the gap. Since speaker efficiency depends upon the square of the Bl product (magnetic flux density (B) times the length of coil wire in the gap), speakers with overhung coils waste that proportion of electrical energy that goes into creating alternating magnetic fields outside the gap. For a long-throw speaker, this may be half or more of the energy input.

Historically, creating a very tall magnetic gap while maintaining a high flux density was so difficult that using an underhung coil meant using very short coils, which cut down the length of wire that could be in the gap. The Aurasound structure provides a very tall, very uniform, very intense gap which overcomes this problem. As a result, very long-throw, low-distortion, efficient speakers are possible.

Aurasound does not manufacture the speakers, but licenses the technology to other companies. When speakers using this technology become commonly available (and after the initial high prices have somewhat moderated), we can expect very significant improvements in both speaker efficiency and distortion performance.

PIEZO INTRO

Piezoelectric drivers have been hailed with both cheers and jeers for well over a decade. During that time, piezos made by Motorola and others have been applied and misapplied to countless systems, mostly in the pro-sound and medium-fi markets. Yet certain piezo types, correctly applied, perform as well as their electromagnetic counterparts.

Pierre and Marie Curie discovered in the late 1880s that certain crystals, when stressed or flexed, produce an electrical potential difference between opposite faces. The effect is

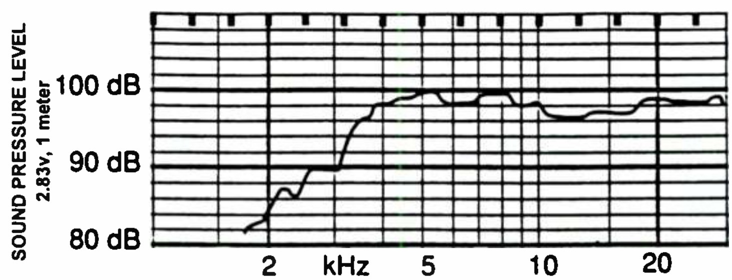


FIGURE 7: Response of Motorola KSN1177A Piezo Dual Horn (from published data sheet).

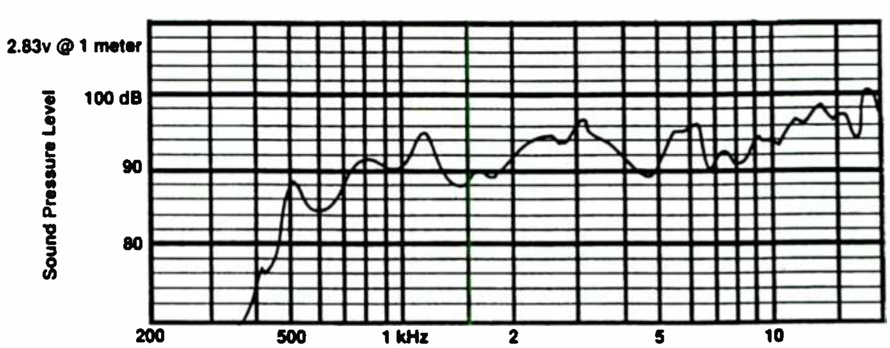


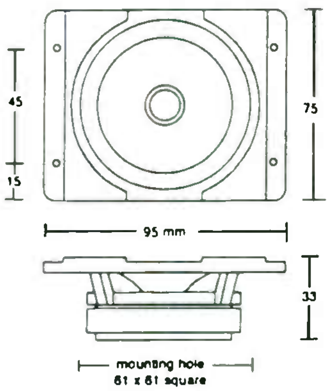
FIGURE 8: Response of Motorola KSN1188A Compression Driver on EV 8HD Horn (from published data sheet).

Reader Service #31 →

A&S is pleased to welcome **E.J. Jordan Designs** back to the U.S. market. As a pioneer in the development of high-end, wide bandwidth drivers that eliminate the need for crossovers in the critical regions, **Ted Jordan** has refined his low mass **Controflex (metal-foil) cone technology** and designed three drivers that have no equal. The **JX53,** with higher power handling and sensitivity, is the proud successor to the legendary **Jordan Module.** The **JX125** is the 170 mm bass driver designed to complement the **JX53** or any other high performance high frequency driver (Finally the perfect bass-mid for your Accuton drivers). And the **JX92** is a 140 mm wide range with a usable response from 60 to 15,000 Hz making it the perfect choice for a midrange in a system using first order filters.

Jordan drivers will be available exclusively through A&S in the Fall of '94. Call or write for pricing, and Ted Jordan's own cabinet and crossover designs.

E.J. JORDAN DESIGNS



weight: 0.32 kg

Features

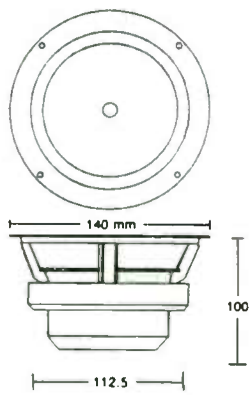
53 mm Controflex metal foil cone
Fluid suspension and cooling
Precision cast rectangular chassis

Thiele-Small parameters

Zn	8 Ω
fs	120 Hz
Qms	1.1
Qes	1.1
Qts	0.57
Vas	1.7 l
Bxl	1.7 N/A
Re	4.7 Ω
Le	10 μH
Cms	2.0 mm/N
Mms	1.0 g
Sd	25 cm²
SPL	86.5 dB
Power handling	50 watts
Minimum loading	1.5 litres, IB

JX53

The **JX53** embraces the musical spectrum from **300Hz-20kHz,** avoiding crossover anomalies in this most critical band of frequencies. Its exceptional transparency and detail is complemented by the **JX125** bass unit which in conjunction with a simple, first order crossover, provides seamless phase-linear performance throughout the entire audio range. The rectangular format of the **JX53** allows close stacking in multiple arrays, allowing a virtually unlimited range of options.



fixing holes: 4 x 5 mm on 112 PCD
weight: 1.1 kg

Features

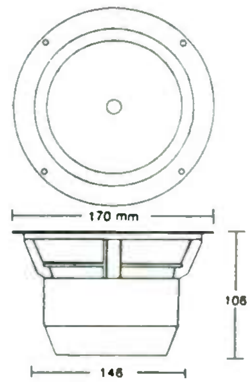
92 mm Controflex metal foil cone
Jordan Axiline rear suspension & fluid cooling
Aluminium die cast chassis

Thiele-Small parameters

Zn	8 Ω
fs	41 Hz
Qms	1.6
Qes	0.75
Qts	0.51
Vas	23 l
Bxl	3.5 N/A
Re	5.8 Ω
Le	200 μH
Cms	2.5 mm/N
Mms	6.0 g
Sd	80 cm²
SPL	85 dB
Power handling	70 watts
Minimum loading	6 litres, IB

JX92

The **JX92** is a full range driver achieving an exceptional bandwidth, **60Hz-15kHz,** from a single **Controflex foil cone.** Mounted in a simple, sealed box enclosure, the unit offers a superb, crossover-free sound quality and is a high performance alternative to conventional, small, nearfield monitors. This makes it ideal for both hi-fi and surround sound/home theater use. Additionally, the **JX92's** compact size makes it an excellent building block for multi-room and interior design applications, where the loudspeakers may need to be concealed from view or built into the structure of a room.



fixing holes: 4 x 5 mm on 162 PCD
weight: 1.2 kg

Featrues

125 mm Controflex metal foil cone
Jordan Axiline rear suspension
Magnesium die cast chassis

Thiele-Small parameters

Zn	8 Ω
fs	33 Hz
Qms	2.0
Qes	0.58
Qts	0.45
Vas	49 l
Bxl	5.4 N/A
Re	6.5 Ω
Le	12 μH
Cms	1.9 mm/N
Mms	12.5 g
Sd	137 cm²
SPL	86.5 dB
Power handling	100 watts
Minimum loading	25 litres, IB

JX125

The **JX125** is a bass/midrange drive unit, designed to complement the **JX53.** Its wide bandwidth permits the use of a first order crossover to provide a seamless, step-free sound quality throughout the entire audio spectrum. Straightforward sealed box loading enables the **JX125** to operate down to 30Hz and below and its low mass cone and unique rear suspension provide tremendous speed and control.

A&S speakers

3170 23rd Street; San Francisco, CA 94110

Telephone: (415) 641-4573 Fax: (415) 648-5306

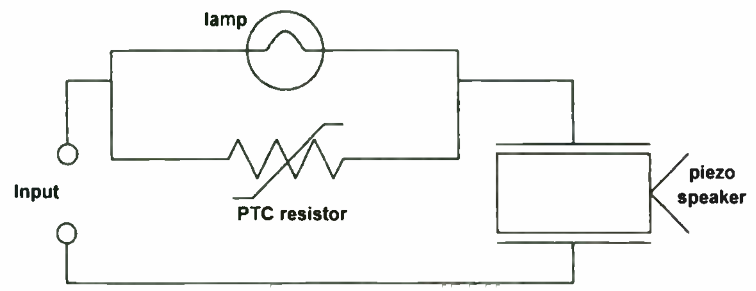


FIGURE 9: Power limiter for piezo speakers.

FIGURE 10: RAINBOW piezo actuator.

reversible; when a potential difference is applied, they flex. This phenomenon was named piezoelectricity, from the Greek verb *piezon*, meaning to press. [The correct pronunciation is "pie" (as in apple), eeee (as in "I just saw a mouse."), zoe (as in "so," pronounced by a German).]

Virtually all of today's piezo speakers use a "bender" made in bimorph form, as shown in *Fig. 6*. Not only does this provide greater maximum flexure than would a single layer of ceramic, but also it is inherently balanced, producing very low levels of even-order harmonic distortion. Piezo bimorphs are massive devices which can exert a significant force. Therefore, they are usually used in conjunction with a horn, which acts as an impedance transformer between the high acoustic impedance of the bimorph and the low acoustic impedance of the air into which it radiates. A stiff paper cone provides the acoustic coupling between the bimorph and the horn throat.

Electrically, the piezo acts as a capacitor with a dissipation factor of about 3%. Its capacitance is typically around 1μF, depending upon the specific model. Because of this capacitive impedance, piezos draw almost no current at low frequencies, increasing at the rate of 6dB per octave as frequency increases.

At 20kHz, most piezos still have an impedance over 20Ω. This means that no crossover is required for a piezo tweeter used in a multiway system, except to limit radiation of high-frequency energy from the low-frequency drivers. It also means that at high frequencies a resistance must be wired in series with the piezo to prevent amplifier instability from an overly capacitive load.

Usually this resistance is about 20Ω at 10W. However, with the combination of a high impedance and a low dissipation factor, the in-phase current in the piezo is so small that one engineer reported successfully using 2W resistors even for arrays of piezos operated at high power.

A PAIN IN THE PIEZO

Motorola has long led the field in producing high-quality piezo speakers. Their early horns produced a sizzling top end with almost ideal efficiency, and quickly became favorites of pro-sound designers of the '70s. Unfortunately, these simple horns had a rather peaky response with little output below 5kHz. While this might not be much of a problem if used correctly, most weren't.

The standard configuration asked the piezos to provide all the top end from about 4kHz on up (if indeed the musical-instrument-grade woofers used in the sound columns of the day could squeak out 4kHz!). Since there was still a hole in the top, the tendency was to overcompensate by using too much drive to the piezos. Often the excruciating peak was also bathed in distortion caused by supersonic amplifier oscillation due to omission of the series resistors. Many engineers still shudder at the mention of piezos for this reason.

Even in the well-designed systems, there was a general lack of understanding of the piezos' failure modes. Power consumption is extremely low, but a maximum voltage can safely be applied without causing the bimorph to crack. When this voltage is exceeded, the speaker may go silent, but more often, it just buzzes. Perhaps as many engineers were turned off by the sound of damaged piezos in the early days as by incorrectly used ones.

Motorola has not been idle in piezo technology for the last 20 years. Two relatively new products discussed at the symposium were the KSN1177A dual horn and the new KSN1188A compression driver. The dual horn contains two elements connected in parallel to provide 6dB greater sensitivity than the standard piezo horns: 99dB at 1W/1m rather than 93dB. This sensitivity very nearly matches that of many pro-sound woofers. Also, the elements in the dual horn are carefully manufactured so that their resonances are complementary, giving a relatively smooth wide-range response on axis, as shown in *Fig. 7*.

The new KSN1188A compression driver is the only high-fidelity piezo product with response below 1kHz. The response of this driver on a good horn is shown in *Fig. 8*. While lacking sufficient sensitivity for pro-sound work, the driver could be used in high-efficiency home systems.

Because of the addition of a new input-voltage-limiting circuit, this driver can be connected to systems delivering up to 400W at 8Ω without damage. The limiting circuit uses an incandescent bulb connected as shown in *Fig. 9*. The onset of limiting is quick but smooth, making the operation of the limiting circuit almost inaudible. For inputs below 28V RMS (100W into 8Ω) the limiter is not active.

Aura Ceramics (formerly Honeywell Ceramics) is developing a new ceramic technology called RAINBOW (Reduced And INternally Biased Oxide Wafer) tech-

Continued on page 41

The next Loudspeaker University will take place on Nov. 15-16 (introductory workshop) and Nov. 17-19 (advanced workshop) at the Airtel Plaza Hotel in Northridge, CA. For further information contact Mike Klasco at (510) 526-8087.

SOURCES

Motorola
4800 Alameda Blvd., N.E., Albuquerque, NM 873113
(505) 822-8801, FAX (505) 822-8812
KSN1177A dual horn
KSN1188A compression driver

Aura Systems
2335 Alaska Ave., El Segundo, CA 90245
(310) 643-5300, FAX (310) 643-8719
AuraSound Speaker

Aura (Honeywell) Ceramics
(An Aura Systems Company)
5121 Wimmetka Ave. North,
New Hope, MN 55428
(612) 535-9660
RAINBOW Benders

Linaeum
1238 N.W. Glisan, Ste. 404
Portland, OR 97209
(503) 222-5138

Continued from page 38
nology. *Figure 10* shows a bender made using this technology, which chemically forms electrode layers on each side of the piezoceramic. Note that the bender has no voltage applied; it is prestressed due to a step in the manufacturing process. This prestressing provides significantly greater flexure than a bimorph, and can withstand greater maximum voltages.

Aura is not interested in manufacturing piezo tweeters using this technology, but in manufacturing the RAINBOW benders and/or licensing their use to a speaker manufacturer. At this point, the technology shows promise, but it has not yet been developed into a specific speaker product.

MAGIC FLUID

Almost anyone seriously interested in speakers in the last ten years is familiar with ferrofluid. Much time at the symposium was devoted to discussion of the applications of this rather marvelous liquid. Essentially, ferrofluid is a colloid whose base is an oil (hydrocarbon or synthetic ester), in which tiny magnetic particles (about 10^{15}–10^{16} per cm^3) are suspended. A small amount of surfactant is added to prevent clumping of the particles.

The primary purpose of ferrofluid is to help remove heat from the voice coil, increasing short-term maximum power handling capability. (Long-term power handling is ultimately limited by how rapidly heat energy can be removed from the speaker frame and magnetic structure.) The magnetic particles make the fluid remain in the gap, despite the coil's pumping action and the effects of gravity. Aside from simply enabling a speaker to handle larger short-term power bursts, the heat conductivity of ferrofluid reduces power compression, which is a temporary reduction in a speaker's sensitivity due to a temperature-induced increase in voice-coil resistance.

In addition to the heat-dissipation improvements available from ferrofluid, the substance provides other performance advantages, including the ability to hold the voice coil centered in the gap. Several design factors can cause the cone to rock somewhat as it moves, which sometimes causes the coil to scrape the top plate or pole piece. As you might imagine, this produces gross distortion and can ultimately damage the coil. The moving ferrofluid generates a restoring force that helps prevent this from happening. In fact, the reduction in rocking reduces harmonic distortion by about 5–10dB, even when the rocking is not so severe as to cause scraping. Spectral contamination measurements also show improvement with the addition of ferrofluid.

To reap the full benefits of ferrofluid, its properties must be taken into account when designing speakers. Retrofitting existing speakers with the fluid often causes more problems than it solves.

FOR GOOD MEASURE

Both designers and hobbyists share the need for good measurements. Among the presenters at Loudspeaker University was Dr. Sunil Puria of AT&T Bell Laboratories, who demonstrated the SYSid measurement system. The system is an FFT-based analyzer whose operation requires a digital signal-processor board in a computer. (*SB* readers will find an excellent synopsis of FFT analyzer operation in the series of articles describing the IMP [Part I, *SB* 1/93, p. 10; Part II, *SB* 2/93, p. 30; and Part III, *SB* 3/93, p. 36]. Discussions of the relative performance merits of several analyzers can be found in the *SB* Letters column of the last year or two.)

The SYSid system is very sophisticated, permitting not only measurements of frequency and phase response, but also single-frequency and swept-frequency distortion measurements and spectral contamination measurements. Assuming you already have a computer, you can buy your own SYSid package for about $3,000. A measurement microphone can cost anywhere from $100 for a Mitey Mike up to several thousand dollars for a lab-quality measuring microphone. Naturally, individual calibration of any microphone used for measurement is a must.

Of course, it is not possible to completely cover a three-day symposium in a single article. Much of the presentation and discussions concerned matters of little interest to most hobbyists, since the purpose of the event was to enhance quality and communication among industry professionals. My aim here is to present important developments that will likely affect amateur and professional alike in the near future. Readers interested in more in-depth articles on any topics mentioned here should contact *SB*.

PREVIEW
Glass Audio

Issue 3, 1994

- A Purist Power Supply
- An Amplifier Soft-Starting Control
- Differential Series Feedback Pair
- More on Output Transformers

Wayland's Wood World

SPLINE CORNER JOINTS

By Bob Wayland

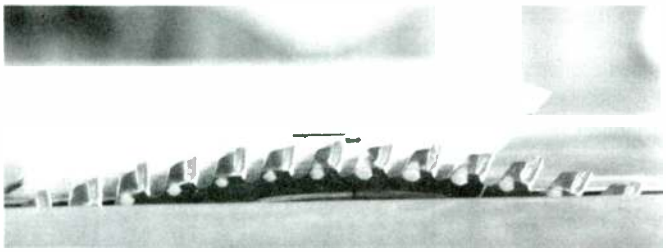


PHOTO 1: Measure and mark the midpoint of your board and set the depth of your blade accordingly. Note the void in this piece of furniture-grade plywood.

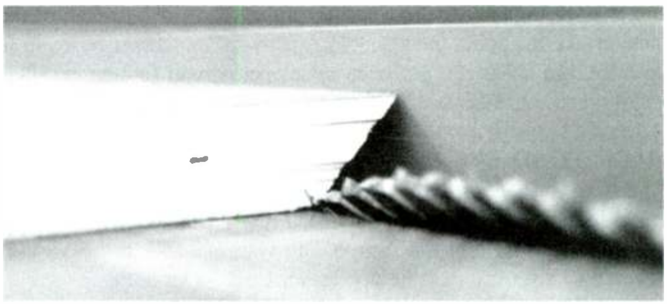


PHOTO 2: Setting the saw fence for the spline cut.

Last issue (*SB* 5/94) I demonstrated how to make a corner joint using a specially constructed biscuit jointer jig. But those who don't own a biscuit jointer, or prefer a more traditional approach, should consider the spline corner joint, which is equally strong and good-looking. You can make this joint with a table saw and careful attention to detail.

As with the biscuit jig, you must first be sure your saw is accurate. The most critical setting is at 45°. To check, make two saw cuts, then hold the pieces together and test with a try square to see if the resulting corner is a right angle. You can also verify the cut with a 45° try square, although they are notoriously imprecise. With your saw set to 45°, cut the sides of your enclosure that form the corner joints.

JOINT CONSTRUCTION

The strong spline corner joint can be used with any material, but is especially suited for plywood and medium density fiberboard (MDF). It consists of a rectangular stick of wood (the spline), which fits into mating slots cut at right angles to the 45° faces. Because it extends the entire length of the joint, it forms an airtight seal for your speaker.

Most woodworking references advise you to locate the spline about 1/16″ from the inside edge. With speaker enclosures you can move the spline cut out to the very inside edge of your 45° face. The depth of the spline cut should provide adequate gluing surface without weakening the corner's structure. Most woodworkers set the saw blade depth half the distance of the board's thickness (*Photo 1*).

Next, mark where the spline will be relative to the inside edge of the joint face. With this as a guide, set the fence of your saw to correctly position the spline cut (*Photo 2*). Keeping the edge of the joint firmly against the fence, make the cut as shown in *Photo 3*. You may discover that you are more comfortable cutting with a miter guide, which not only helps keep the board against the fence, but also provides even pressure for guiding the board during cutting.

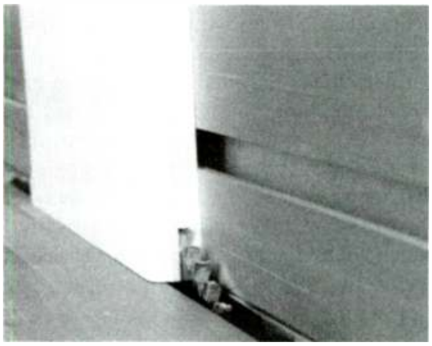


PHOTO 4: Making the first cut to create the spline inserts.

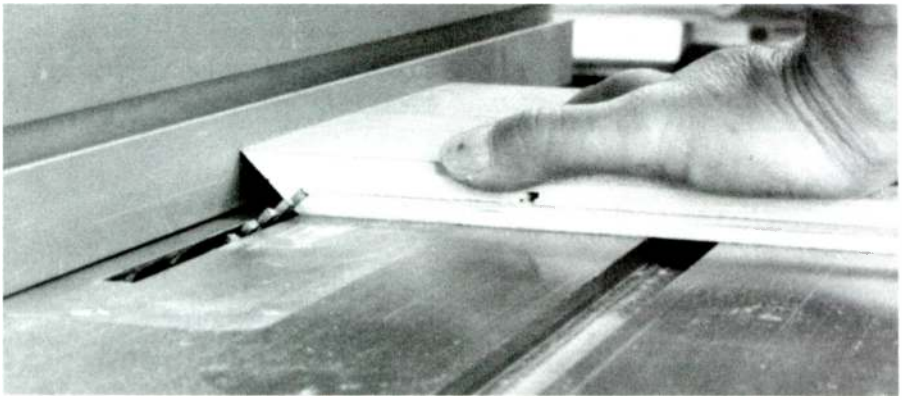


PHOTO 3: With the board firmly against the fence, make the spline cut.

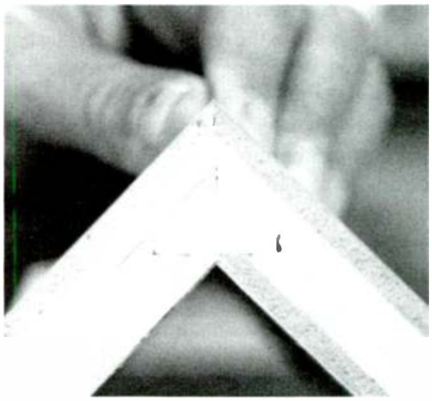


PHOTO 5: Test-fitting a spline corner joint.

PHOTO 6: Note the smooth, clean joint along the outside edge of this corner spline joint.

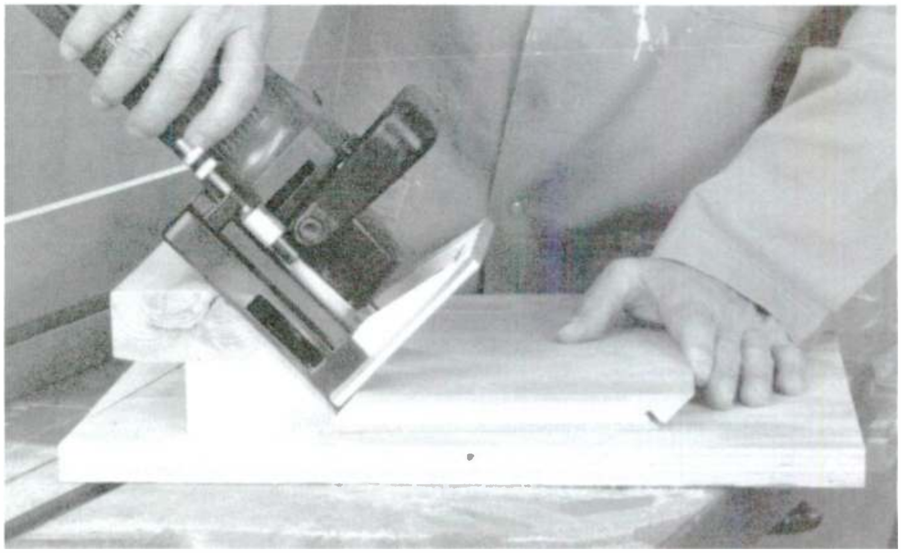


PHOTO 7: Using the biscuit corner jig and the biscuit joiner to make a spline cut.

Rotate the board 180° and cut the other end. Follow this procedure for all of the boards. Because you are working with uniform material, you will be surprised how quickly you make all the necessary spline cuts. Although not absolutely necessary, it is a good idea to use a blade that makes a flat-bottomed cut (see below).

MATERIAL

Next, you need the spline itself. Some experts maintain that the spline should be the same material as boards being joined; others disagree. In our case—working mostly with MDF or plywood—MDF is not suitable, and the plywood we use is not the same thickness as the kerf of our saw blade. I recommend you use solid wood, preferably a strong hardwood. The grain of the spline must be perpendicular to the long dimension of the spline for maximum strength.

After setting the blade depth to the width of the spline (easily measured when two of the pieces are held together), space the fence to the thickness of the spline cut. Using scrap wood, cut the spline so that the grain is as shown in *Photo 4*. You can produce two splines by rotating the board 180° and making another cut. Reset the blade depth to the thickness of the spline and spaced from the fence the width of the spline. Cut the splines.

Continued on page 66

Software Report

THE LISTENING ROOM FOR MACINTOSH

Reviewed by Victor Staggs

The Listening Room for Macintosh, Sitting Duck Software, PO Box 130, Veneta, OR 97487 (503) 935-3982. Requires 512K RAM. Hard-drive installable. Introductory price $47.50.

Ralph Gonzalez—audio-related software developer and frequent contributor to Audio Amateur Publications—has released The Listening Room (TLR) for the Macintosh computer. The availability of this room acoustics program, formerly only for IBM-compatibles, signals a significant event for computer-aided sound system design.

The percentage of Mac computers used for commercial loudspeaker design is small. Digital measurement boards and accompanying processing and speaker optimization software exist almost exclusively for PC-compatibles. So, the author has expended much effort to reach an as-yet-untapped market.

In the audio amateur marketplace, however, Mac use—although hidden due to a dearth of software—is likely to be more significant. In fact, in the professional media marketplace, such as image- and sound-processing houses in Hollywood, the Macintosh reigns supreme by virtue of its built-in graphics and sound capabilities.

A HIT IN HOLLYWOOD

The Macintosh is noted for its user-friendly graphical interface, which makes programs easy to install and use. For a Hollywood production team, this ease of use is a big plus when performing complex tasks on a tight schedule. Amateur users will also wish to take advantage of this ease, and TLR mostly delivers on its promise to make the Macintosh behave like a friendly machine. Because of my exposure to the collection of Macintosh system facilities and its documentation, I respect the amount of system knowledge and programming expertise a product of this type requires, and Ralph Gonzalez has my appreciation for his efforts.

The TLR system comes on a single 3.5" floppy disk, and the manual describes the installation procedure. The printed instructions are terse, and startup requires some Macintosh-user savvy, so if you are new to the Mac have an experienced friend sit in on this one. The manual could be more helpful in this area with expanded installation and startup screen displays.

HOW'S YOUR FPU?

You'll have to determine whether your Mac has a built-in math coprocessor—also called a floating-point unit (FPU)—the Motorola MC68881 or MC68882 chip. This chip potentially makes a number-crunching-intensive program run faster by a factor of 40 to 700, according to Macintosh literature. You're more likely to experience a speed increase by a factor of 20. With the FPU chip, your system has near real-time interactive speed, and I wouldn't be without the FPU in my Mac.

DO YOU READ ME?

You must also determine the version of your Mac operating system. The Read Me First file in the newly installed application folder was created in TeachText, a basic Mac word processing application bundled with System 7 for file interchange. If you click on the Read Me First document icon, TeachText will obligingly open and display the latest release notes.

Older Mac computers with System 6 won't have TeachText, and you won't be able to install it because it is compatible only with System 7. To read the Read Me First document, you will need to open it from your own word processing application, such as WriteNow© or MacWrite©, using its file translation capability. Then you can read and save the translated version. This is the kind of detail that drives program developers nuts—there is no universal file-opening utility.

DROP 'N DRAG

Upon entering the application, you'll see a menu bar at the top of the screen and a stack of one or two windows. One, labeled Calculations, is a permanent fixture even though it may be covered by a second window. Calculations is the display window for the output graphics, which consist of a frequency response plot of the direct sound with the early room boundary reflections, plus markers showing the frequencies and relative strengths of the room resonances.

The second, unnamed window contains a small plan view of the listening room with markers indicating the positions of the listener and the speakers. The screen also displays a vertical slider to the right of the room plan with similar markers for height. You can position the speakers and listener anywhere within the room by clicking on the respective cursor, and dragging it to the desired location.

This is an intuitively appealing way to enter data. You can drag the marker on the plan view to set the (x,y) position, and also use the vertical slider to fix the z position. A numerical readout of the (x,y,z) speakers and listener positions lets you work from plans and measured values, and match them on the screen interface.

The user window forms a document that you can save by naming the file. *Figure 1* shows a sample design window. Using mouse clicks or keyboard equivalents, you can close or open this file at any time, in case you work on several rooms, or if you wish, to save several approaches to the same room. Pull down the upper menu bar items with the mouse to reveal choices and preferences for representing and optimizing the room.

HEARING IS BELIEVING

TLR calculates the frequencies and approximate relative strengths of acoustic standing waves along the height, width, and length of a room. It ignores standing waves with oblique incidence. A point isotropic radiator in space represents the sound source. You can modify the acoustics by including a carpet or by eliminating the vertical or horizontal resonances.

You can model frequency response of the source itself by considering it as ideally flat, a sealed box, or a vented (fourth-order) alignment. You determine the -3dB frequency, but the Q, or alignment, is an internally fixed, unknown value.

Calculation of the direct sound is straightforward. The program handles early reflections by assuming the surfaces reflect specularly. Gonzalez refers to this as a

ABOUT THE AUTHOR

Victor Staggs received a PhD in Communication Theory and Systems recently, after a previous career in predicting nuclear radiation transport and shielding for use in defense applications. Now he is a consultant in communication theory, detection theory, physics, and electroacoustics. He has recently turned his attention to a computer program for solving the acoustic wave equation.

"bouncing-ball" model: The angle of reflection equals the angle of incidence, relative to a surface normal. Early reflections also take into account the presence or absence of a ceiling. According to my understanding of the documentation, the carpet does not affect the standing waves, only the early reflections.

The room-reflection model is complete for a single generation of reflections. That is, the model considers all ways of sound bouncing off one, two, or three walls and returning to

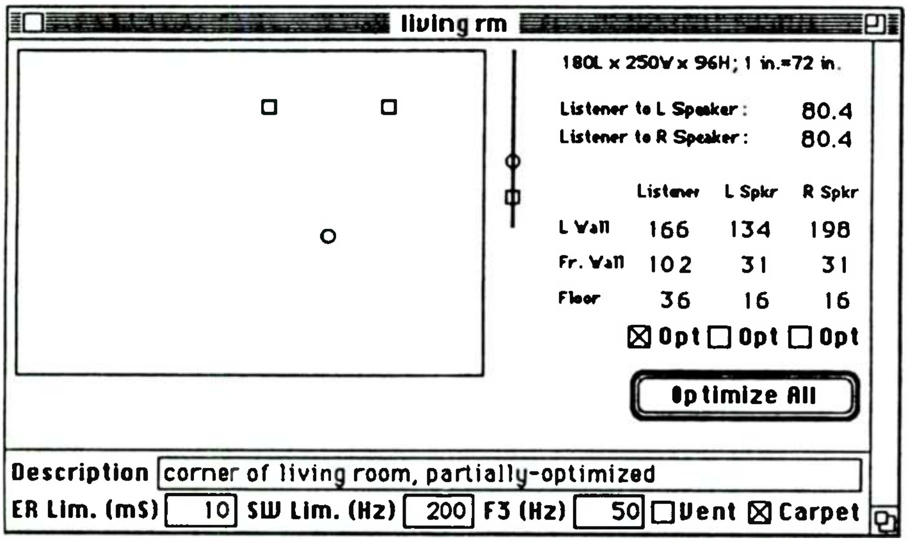


FIGURE 1: Sample design window.

the listener, up to (but not including) the point of bouncing off a wall for a second time.

TLR is restricted to the low-frequency range of the loudspeaker-room combination, which makes sense when you consider room resonances and speaker placement. The resonances are restricted to below 200Hz, but you can change this limit. The display window goes up to 500Hz, as does the direct-plus early-reflected sound curve.

Gonzalez explains further details of the acoustical model, and also includes a useful list of references for those who thirst to know more. This information is useful to interpret the program and develop a seat-of-the-pants feel for its applicability.

OPTIMUM OPTION

You can drag the speaker and listener symbols on the user window, and the display window will update itself (unless you indicate otherwise). If you had two screens, or a very large single screen with room for both windows, you could see the resonance and early sound display change in real time, and optimize the listener setup via the controls. But on single screens, the user window covers the display window, and you have to click on the display window to see the responses, so this eliminates visual real-time user optimization.

However, if you click on the Optimize All button on the user window, the program will optimize the setup for you. As the program runs, the little speaker and listener markers will move on the floor plan and the height slider, which is both informative and entertaining. When the optimization is complete, you will notice a smoother resonance distribution, plus the new associated early sound response. Fortunately, the early sound response is somewhat related to the evenness of the resonance distribution, so when tuning for late-time response, the early response tends to follow.

Gonzalez explains that the optima are local, meaning that you can start with different initial conditions to arrive at different optima. This encourages you to experiment with different initial setups to find the best local optima. To help you document all of this, the program lets you save more than one input window in the stack, and then access—and print—them at will.

BOUQUETS & BRICKS

Generally, TLR is user-friendly, and the new self-optimizing feature is much appreciated. The author also does a pretty good job explaining the program uses and the applicability of the physical models it embodies.

I agree with his treatment of the early reflections and their addition to the direct sound. At early times, this is what you perceive, and it has a great bearing on your perception of transient response. You can tell the difference in bass quality between a large speaker and a minimonitor by the early time signature, even though the measurements, including the late reverberation, are the same.

The physical model limits the representation of the speaker and listener to points in space. The program could give the listener a larger size by calculating the response at more than one point and averaging the results.

Extending the speaker model to a finite size would require a vastly different program from the present one, so you'll have to accept its limitations and use it with care. Very large or directional speakers will not have early sound reflections that match the computer model, although the resonance distribution will probably be similar.

The program addresses only rectangular rooms. I wonder how much effort would be involved to dust off the Fourier transform and make a version that treats more realistic-looking floor plans. I have witnessed few rooms with purely rectangular floor plans. This is not so much a criticism of the present program as it is a suggestion for future development.

MAC INTERFACE

The manual fails to explain the options of printing either the problem input data or the output frequency response data. These print commands are under different menu headings, but should both be placed under the File heading with explanatory names, such as "Print Input" and "Print Output." Ideally, there should be only one Print command, and both the input data and the response plot should appear on a single page, so they can't be separated.

The user input window includes check boxes for choosing whether the listener, left speaker, or right speaker will be included in the optimization. When you check one of these boxes, the program immediately swings into action and optimizes that item by moving it slightly in three dimensions. This violates the Macintosh behavior for check boxes, which are supposedly only used for setting option switches.

This feature should be changed so that optimization occurs only after you've set all of the check boxes and then clicked on a large Optimize button. You should also be able to choose to Optimize All by setting all of the switches on.

WHERE SHALL I SIT?

In addition, optimizing the speakers and listener moves them vertically, as well as horizontally. Many users cannot adjust their chair height nor their speaker height, so these variables should be fixed or free at your option. In the program's defense, the vertical movement allowed is slight, but I still don't wish to slouch nor sit bolt upright on a sofa.

Optimal room setups resulting from program automation often have unequal listener-to-speaker distances between the left and right speakers. In fact, they can vary by inches. Beware of this and move the final setup around to equalize distances, recalculating to see how much quality is lost. The optimum is computed on the basis of the room resonance distribution, and if you wish to preserve imaging and early reflection symmetry, then you must hand-adjust the computer setup.

Of course, you'd expect to readjust the speaker and listening chair positions in the actual listening room. This is a great feature of reality: it is not always user-friendly, but it is always available.

A RESONANT ROOM

With a program like this in my hands, I could not resist applying it to real-world cases. My first test involved the IEC room designed under the stewardship of Dr. Floyd E. Toole, then at the National Research Council of Canada. Dr. Toole used this room for his famous research into the ability of listeners to rate loudspeaker quality via listening tests under tightly controlled conditions. Designed for pleasing distribution of acoustic resonances, this room simulates conditions in a large home listening room.

Its dimensions, which I had to convert into inches for program input, are 4.1m wide, 6.7m long, and 2.8m high. I started with the speakers 1.0m from the side walls and 1.3m from the front wall (behind the speakers). The listener was on the room's center line, 3.7m from the front wall. The speakers were originally placed 18" from the floor, and the listener was 38" from the floor.

Because this room is large, the TLR program drew it to a small scale to fit into the user window. It was difficult to set the positioning dimensions, since the cursor did not resolve distances accurately at this scale. With two-inch resolution, it was impossible to center the listener exactly.

I opted for the carpet on the floor and a sealed box source with a 3dB frequency of 42Hz, to match the IEC room and the majority of speakers tested. All room axes were allowed to create resonances, and all listener and speaker dimensions were allowed to vary during optimization. The time window for early reflections was set to 13ms to conform with D. B. Keele's practice in his speaker reviews for *Audio* magazine.

LISTEN UP

Without optimization, the direct- plus early-reflection plot showed a dip at 180Hz, as the handy cursor utility indicated, and the room resonances were not well distributed. After optimization, the speakers were the same distance from the rear wall, but 0.89m (35") from the side walls. The speaker height was lowered to 14", and the listener stayed at 38". However, the listener's chair was now 2.43m from the left wall, putting it 6.1" off-center to his/her right.

This put the listener 116.2" from the left speaker and 103.7" from the right speaker, a difference of 12.5", or roughly 1 ms of sound travel, or one whole wavelength (360°) at 1kHz. The resonances were very well distributed, but the early sounds were much different between speakers. The left response curve dipped at 128Hz, and the right response curve dipped at 202Hz. Both curves showed a minor dip at about 62Hz.

Clearly, from an imaging standpoint, this would not be acceptable, so I manually layout and plotted the responses. The resonances were not nearly as smooth as those in the IEC listening room, both before and after optimization. Furthermore, the 40Hz region showed a hump in the early-time response, perhaps due to the speakers having a fourth-order response down to 32Hz.

One speaker is near the right corner, and the left speaker is roughly centered along the long front wall. Therefore, their early-time responses have dips at different frequencies above 100Hz, affecting their imaging. I know from experience that they image much better if I lean forward on the sofa. Also, the bass is

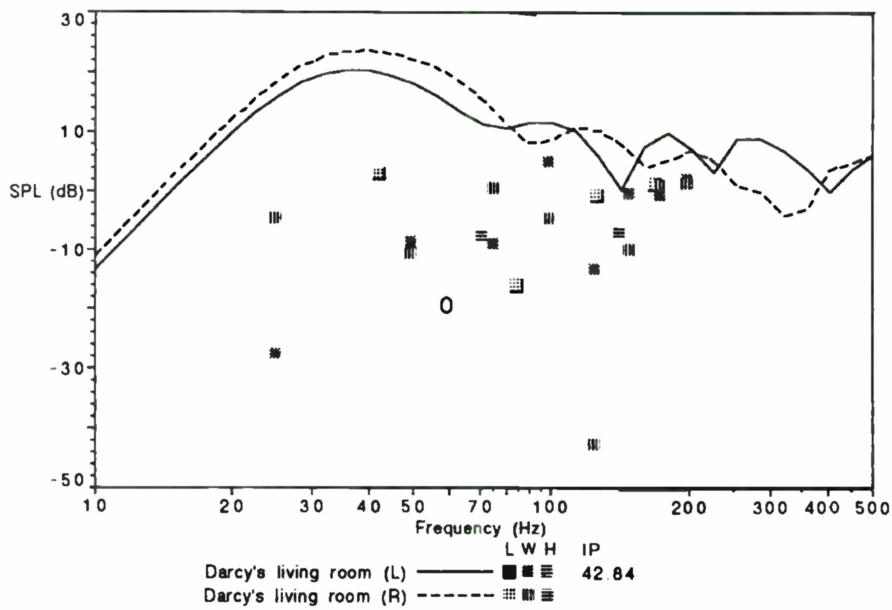


FIGURE 2: TLR-predicted resonances (symbols) and early-time response (curves) for Darcy Staggs' listening room.

recentered the listener. Now the resonances had more variance, but the two early sound curves matched. Toole's published speaker response curves were either anechoic, or they were averaged over many listener positions, so I could not make a comparison. The bass was extended to below 30Hz by the early reflections, if the point source model is believed.

AT DARCY'S

I have access to the second computer model, the listening room of my brother, Darcy Staggs. It is a tract home living room, designed without any thought of acoustics. It is much wider than it is long (deep), and the left half of the room behind the listener has no wall, since it opens into the entryway and the dining area. Nevertheless, I treated it as a whole room in the program, because there is a wall behind the listener's sofa.

I entered the dimensions and listening smoother if I lean forward, and more exaggerated if I lean back toward the wall.

It was no surprise, then, that TLR positioned the listener forward 4", and it might be practical to move the sofa forward by this amount. The program moved the listener 1" closer to the left wall. The program did not save the new listener-to-speaker distances, but I calculated them at 119.1" and 122.1", for a difference of 3", or a wavelength at about 4kHz.

Darcy's speakers image well from far off-axis, so this is not a problem. Imaging due to phase differences is most effective at lower frequencies anyway. The room resonances were slightly better distributed after optimization, but still not wonderful. The early-time hump in the 40Hz region probably exists in real life, but real bass below 100Hz doesn't come along very often in classical music. Besides, we don't use the maximally flat alignment, so the program is probably over-predicting the bass at 40Hz.

GETTING CRITICAL

Figure 2 shows the TLR prediction for the early-time response in Darcy's listening room, plus the distribution of room modes. If I measured the response with a sound level meter and warble tones, I would not expect the total response to look like the early-time response. Darcy's speakers are much larger than point sources, and also many room reflections would contribute to the measurement. In addition, the room resonances would be averaged over by the 1/3-octave warble tones. A string bass played in the vicinity of 40Hz sounds fairly even in amplitude.

My subjective evaluation is that the real room resonances do affect the sound pretty much as does the prediction, but one learns to listen through them and accept the sound as-is. Darcy's speakers were positioned with practicality in mind as much as perfectionism, probably a typical case.

You'll have to make your own decision regarding the relative importance of early-time and late reverberant sound. In acoustics, you must consider the "critical distance," which is the distance from the loudspeaker where the reverberant field equals the direct field. You must think of the reverberant field as having random direction and as being uniform in space. The direct field (ignoring the early reflections) decays as 1/R, where R is the distance to the sound source.

You can measure this distance with a sound level meter while playing a typical signal, such as pink noise. Just start near the speaker and walk away from it while holding the meter. The level will decay to a constant value and then stay relatively fixed. The critical distance occurs at about 3dB above the far-field reverberant level.

In most home listening rooms I have measured, the critical distance is not much more than 1m. Some listeners position their speakers at the opposite end of the room from their chair, which puts them purely in the reverberant field. Naturally, they will be less concerned with the early sound than with room resonances.

To excite the room resonances to their maximum strength with a loudspeaker, you would have to play steady tones or make a very slow sweep through the bass range. This is because it takes some time to excite the high Q resonances. Only occasionally does real music act like a test tone and produce peaky sound in a room. However, the late-time decay of broad-band sounds will probably hang on longer at the room mode frequencies.

THE VERDICT

Despite my criticisms, I do have praise for this program. It worked flawlessly, and never caused the system to hang or crash, nor did I get any warnings about low memory. This is better performance than some highly vaunted software released through mass distribution. Furthermore, it did everything its manual said it would.

I used this program with a Macintosh IIvx running System 7.1 with 8MB RAM. In addition, I tested it on a Macintosh SE/30 with 2MB RAM under System 6.0.5. This should have cramped TLR a bit and challenged it with an older environment, but it passed with flying colors. Ralph Gonzalez includes hints for adapting his program to even earlier Macs so it will run successfully on them, too.

Small criticisms notwithstanding, if you are serious about making the most of your speakers and listening room, this is a versatile and convenient tool to help you do it. If you buy it now, you will get much use from it while a more perfect (but not necessarily more productive) version shows up eventually. Then you can upgrade if you think it is worth getting the last inch of accuracy from a computer.

Macintosh owners will be pleased at the easy and intuitive graphical interface that Ralph Gonzalez has provided, and now they need not feel left in the cold outside the acoustical prediction world anymore.

The Listening Room for Mac is available as #SOF-TLR3M3G for $47.50 plus $3 S/H from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458; (603) 924-6526; FAX (603) 924-9467.

DEVELOPER RESPONSE:

Thanks to Victor Staggs for his informed and Mac-literate review of The Listening Room for Macintosh. I agree that there is a dearth of audio-related Macintosh software; perhaps now that many Macs have built-in CD-quality sound acquisition capability this will change.

Just a couple of items which Dr. Staggs commented on: (1) When using the program on a machine with a small screen, you can reduce the size of the room layout window(s) to make the graph visible. (2) If you don't want speaker height to be affected by the optimization routine, deselect the Floor-Ceiling dimension in the graph window. (3) You can ensure that optimization preserves

continued on page 66

Book Report

THE THEORY AND DESIGN OF LOUDSPEAKER ENCLOSURES

Reviewed by Contributing Editor Robert M. Bullock

The Theory and Design of Loudspeaker Enclosures, by J. Emest Benson. Synergetic Audio Concepts, 12370 West County Rd. 100N, Norman, IN 47264. Available from Old Colony Sound Lab (PO Box 243, Peterborough, NH 03458; [603] 924-6526, FAX [603] 924-9467) as item #BKSA1 for $24.95 plus $3 shipping in the USA.

This book is a compilation of the following three papers by Benson that first appeared in *Amalgamated Wireless Australasis Technical Review* in 1968, 1971, and 1972:

"Theory and Design of Loudspeaker Enclosures," Part 1—Electro-Acoustical Relations and Generalized Analysis.

"Theory and Design of Loudspeaker Enclosures," Part 2—Response Relationships for Infinite-Baffle and Closed-Box Systems.

"Theory and Design of Loudspeaker Enclosures," Part 3—Introduction to Synthesis of Vented Systems.

These papers offer a comprehensive treatment of loudspeaker systems that can be modeled by lumped element methods. The best known examples of such systems are infinite baffles, closed boxes, vented boxes, and passive radiators.

A GENERAL MODEL

In Part 1 the author presents a general frequency response model of a passive-radiator system with box, port, and leakage losses. The transfer function coefficients are cataloged in all their complexity. By choosing the system parameters in this model appropriately, you can use them to model many types of loudspeaker systems. In particular, Benson shows in detail how to choose parameters to obtain each of the system types, but one, named in the last paragraph, as well as one less well-known type. He also explains how the model formulas simplify in each case.

In addition to frequency response formulas, Benson provides transient response formulas from both impulse and step inputs to assist in performance analysis.

I have also independently obtained response formulas from exactly the same general model Benson uses.[1] I wanted a uniform set of formulas for the calculations in my program BOXMODEL. I was aware of Benson's work through citations in Small's papers, but I could not acquire reprints. Except for the model itself, there is little overlap between his papers and my work. My objective was simply to cast the model formulas into the most useful form for programming purposes, while Benson fashioned the model to best assist him in performance analysis.

You should note a discrepancy between his formulas and mine for the most general configuration, i.e., the passive radiator with losses. It appears that one or both of us made mistakes in the general model derivation. However, when the model is reduced to a vented box, closed box, or infinite baffle, both versions are identical. The discrepancies between our passive-radiator versions involve the passive-radiator compliance ratio. This parameter occurs in few enough terms of the transfer function coefficients that I doubt significant differences between the two versions would be apparent.

BAFFLED & CLOSED BOXES

In Part 2 Benson studies infinite-baffle and closed-box systems in detail. He supplies the requisite transfer functions from the general model described in Part 1. Starting with a discussion of driver parameters, Benson includes appendices in which he derives the procedures for determining driver total Q, compliance, and resonance frequency described by Thiele in earlier works. He also provides a procedure for finding voice-coil inductance.

Transfer functions are presented for system diaphragm velocity and displacement, electroacoustic efficiency, and system response. He also determines formulas for phase response, group delay, and transient response.

From these formulas Benson launches into an exhaustive discussion of the model-predicted behavior of closed boxes and infinite baffles. He closes this paper with his observations on relevant criteria for the synthesis of systems for specified performance.

VENTED BOXES

In his Part 3 paper Benson investigates vented boxes with losses in much the same format as he did for closed boxes in Part 2. However, the problem of parameter measurement and synthesis is much more complicated for vented boxes, so he covers both topics extensively.

In particular, he pays much attention to measuring absorption and leakage Qs for vented boxes and offers the most thorough treatment of this topic I have seen. He uses both the closed- and vented-box methods for measuring α and f_S and discusses the effect of voice-coil inductance on system performance. The derivations of the measurement techniques are more thorough and detailed here than in other sources.

This paper also contains a complete discussion of synthesizing vented boxes by graphical interpolation and by fitting responses to the classical QB3 and C4 filter function forms. Again, the coverage of these topics in this paper is much more thorough than in other sources. If you want the hard detail of the necessary formulas to produce these alignments, this is the place to look. I have reproduced many of these calculations for my own work, and, after a cursory comparison, I have uncovered no disagreements between my calculations and his.

TERRIFIC TRIO

In summary, these three papers contain a wealth of material on loudspeaker modeling and model analysis. There is more detail and completeness here than in any other single source. By carefully studying these papers, you should achieve a good understanding of the performance capabilities and limitations of vented boxes, closed boxes, and infinite baffles. I highly recommend this book to anyone who is interested in loudspeaker system design, who is somewhat familiar with circuit analysis techniques and tools, and who is comfortable with mathematical formulas. Even if you have trouble with things mathematical, Benson's discourses on performance matters are informative and worth picking out to read.

The papers in this book are not Benson's only contribution to loudspeaker design. He also wrote an equally comprehensive series of papers on filter-assisted alignments for closed and vented boxes. All these papers were reprinted in *Loudspeakers, Volume 2*, and I commend them to your attention.[2]

REFERENCES

1. Bullock R., "A Unified Model for Closed-Boxes, Vented-Boxes and Passive-Radiators with Losses," AES 87th Convention, preprint 2841, 1989.

2. *Loudspeakers, Volume 2: An Anthology*, Raymond E. Cooke, ed. AES, 1984. Available from Old Colony Sound Lab (PO Box 243, Peterborough, NH 03458, [603] 924-6526, FAX [603] 924-9467) as item #BKAS1/2 for $29.95.

Available from Old Colony is SOF-BEN1BXG ($9.95 plus $3), a unique software package by G.R. Koonce, which includes the program BENSON.EXE for use with the Benson book. Further information about the software, as well as a book and software special for $29.95, is available in Old Colony advertising in this issue.

VERNIER TUNING FOR A FUNCTION GENERATOR

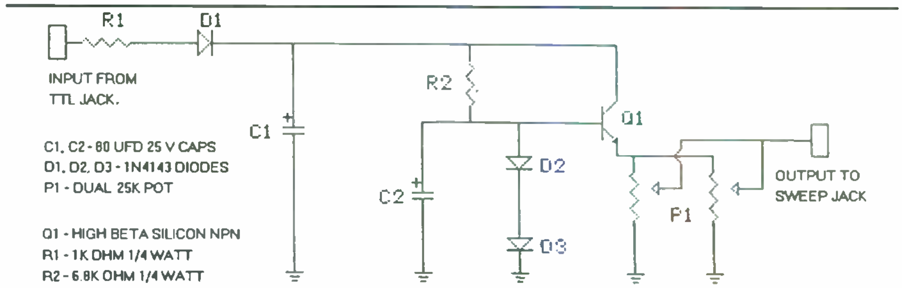


FIGURE 1: Schematic for vernier tuning control.

I recently purchased a new function generator for my impedance meter setup. I selected a model with a large tuning knob to make locating impedance peaks and dips as easy as possible. After I tried the new unit, I soon realized that the tuning was too touchy to allow easy peak/dip location. It needed a vernier tuning knob, but I was reluctant to open and modify the unit, and thus void the warranty.

The function generator featured a BNC jack input for frequency sweep. A voltage applied here would allow vernier tuning. After testing, I determined that about 0.5V would provide a reasonable vernier range at the top end of the frequency dial, but only about one-tenth of this voltage is required at the bottom end.

To accomplish this I needed to take over the tuning remotely, providing both the coarse and vernier controls. I soon envisioned a remote box with power supplies and two pots to accomplish the task. But the last thing I needed was another piece of test gear with an AC line cord in the test setup.

A BETTER APPROACH

A battery with a pot—although a little coarse at the bottom end of the dial—would be a simpler modification and provide vernier tuning. But I anticipated that I would forget to turn off the battery, and battery life would become an issue. I then noticed a TTL output BNC on the function generator, and wondered whether it could produce the needed voltage to replace the battery.

Testing showed the TTL output was a square wave at output frequency covering about 0 to +4V into light loads. Averaging this with an RC network produced about 1.9V DC. A peak network could produce a much higher voltage, but the drivers that produce the TTL output cannot be expected to deliver much current in the pull to +4V mode.

I ended up using a diode with 1kΩ in series to a capacitor to produce about 2.6V DC. This was sufficient to allow simple regulation to produce 0.5V DC across a pot for vernier tuning.

Figure 1 shows the schematic for the circuit that works with my function generator. I did not have a 10kΩ pot with a good feel, so I used the dual 25kΩ pot with the two sections in parallel. The pot is wired so that the sweep input sees 0V with the vernier full clockwise, thus the function generator frequency knob is calibrated.

Turning the vernier pot counterclockwise applies positive voltage to the sweep input and reduces frequency on the generator. The capacitors are sized to provide ripple-free voltage across the pot down to 10Hz. If you let ripple get to the sweep input that is synchronous with the output frequency, you will distort the function generator output waveform.

The result is a "passive" box that attaches to the function generator via two BNC cables and provides a minimum range of 5% vernier tuning. Many function generators provide both the TTL output and the sweep input jacks, so this approach may be helpful for others. Just remember not to attempt to pull too much current from the TTL output BNC.

G. R. Koonce
Liverpool, NY

Vintage Designs

Wharfedale Designs

Courtesy of Paul Penk, Santa Maria, CA 93454

The following diagrams show the essential features of enclosures which give optimum results (with relation to size) with Wharfedale free suspension speakers. Other units can be used in these cabinets provided the cone resonance is not higher than the equivalent Wharfedale speaker. (From Wharfedale's Cabinet Construction Sheet, Oct. 1958)

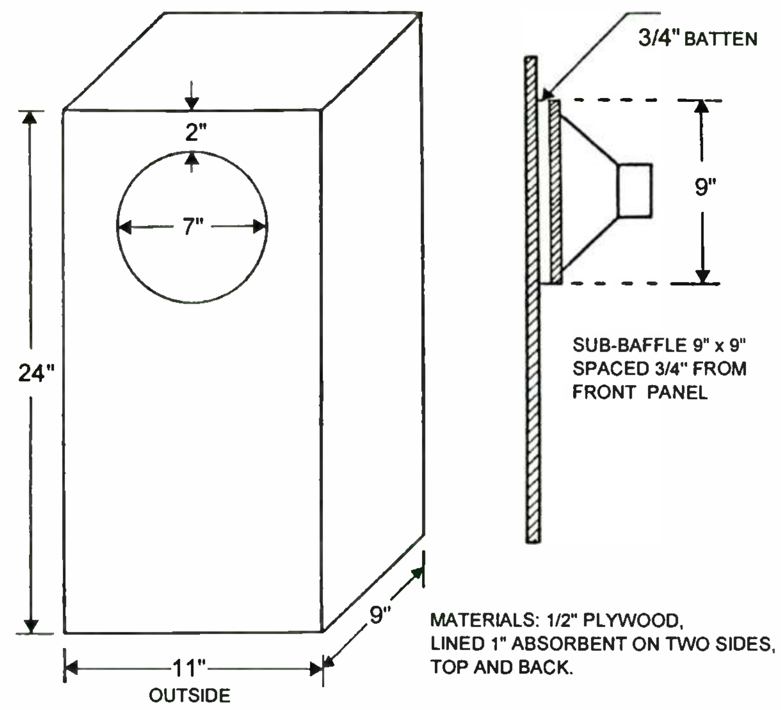


FIGURE 1: Totally enclosed cabinet for 8″ unit with spacing to replace tuned vent. Volume = 1 ft^3. This small enclosure is simple in design and gives optimum results with the Super 8/FS/AL, the new Column 8/145 unit, or the 8″ Bronze FS/AL. The sub-baffle is spaced ¾″ away from the front panel, and the rest of the cabinet is totally enclosed as advocated in the first edition of *Loudspeakers* ten years ago.

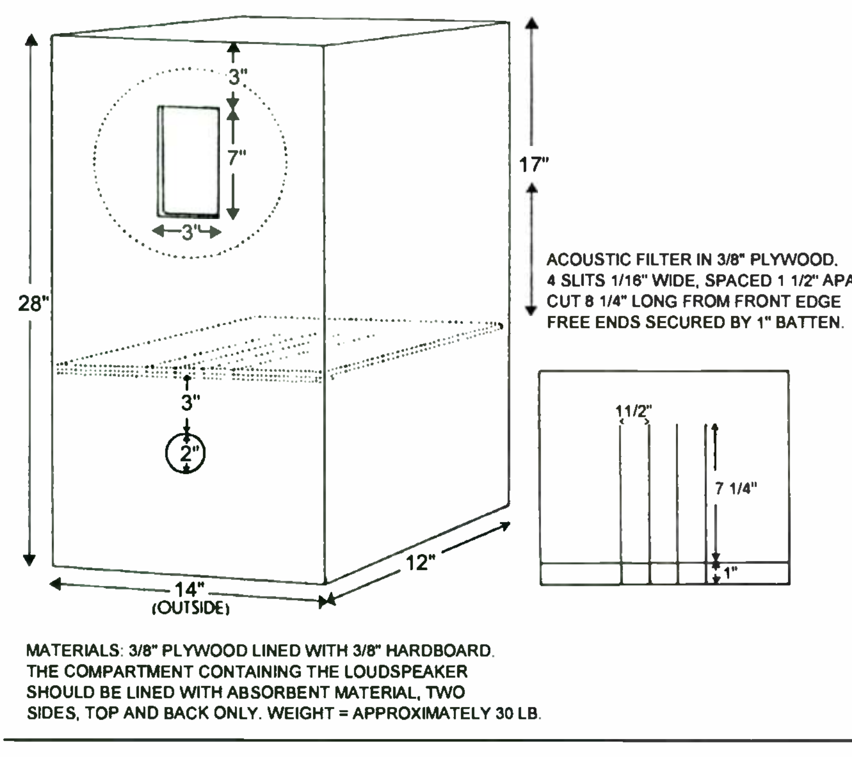


FIGURE 2: Ten-inch reflex cabinet with Wharfedale acoustic filter (patent applied for). Excellent results with only 2 ft^3 of volume. The rectangular opening improves diffusion in the upper register. The vent is tuned to about 40Hz and will blow out a lighted match at this frequency with about 3W input to the 10″ Bronze/FSB or about 2W to the Golden/FSB or W10 FSB. The Wharfedale acoustic filter improves the power handling capacity at low frequencies and reduces standing wave effects.

SB *Mailbox*

CAPACITOR CORRECTION

Figure 1 of my article ("Converting Radio Shack's SLM to Millivolt Use," *SB* 4/94, p. 12) incorrectly shows a 15pF capacitor in series with the input attenuator. The input is directly to the top of the attenuator, and the capacitor, which is connected between the top and the wiper, providing HF compensation, should be 150pF, as explained in the article text.

Without the HF compensation (i.e., the 150pF), the two meters I modified read 3dB down at 20kHz and 1dB down to 10kHz. Without the LF correction (the 47μF), the meter was 4dB down at 20Hz.

C.L.P. Carrington
Barbados

A CHALLENGE

Since *SB* readers are very learned, sophisticated, and clever, I'm asking them to share their expertise in building a servo motor subwoofer that resembles an umbrella with an electric motor mounted at a 90° angle to the stem. I believe the Intersonics aerospace company originated the idea, and supposedly just used a cheap stamped steel frame (basket), a rocker assembly, and a DC external motor to achieve 1½" X_{MAX} and 1% linear distortion; the standard magnet and voice coil are omitted.

Otherwise, the unit does not appear as complex as motional feedback subwoofers with accelerometers and perhaps could be used with 5¼" models where space is of prime consideration. I encourage readers to write to *SB* and share their expertise. The units are now being touted by such car stereo firms as Phoenix Gold for $3,000, while rock musicians have used them for years.

Edward Endicott
San Francisco, CA 94122

THE RIGHT EQUATIONS

In SB Mailbox, *SB* 5/92, p. 51, Mr. Hansen asks which of two cited equations is correct. In the same issue, Mr. Bullock states that both versions are incorrect, and gives the correct equation.

In SB Mailbox, *SB* 1/93, Mr. Katz notes ambiguities in the expression offered by

Mr. Bullock; in the same issue, Mr. Bullock gives a correction to the "correct" equation.

It seems, however, that when the latest version of the correct equation is converted to the cone displacement to voltage ratio form, it exactly matches the second equation cited by Mr. Hansen (line 420 of "Box Response"), which Mr. Bullock originally rejected as incorrect. In view of this, it is necessary to ask all over again: Is the expression in line 420 correct or incorrect?

David J. Meraner
Scotia, NY 12302

Contributing Editor Robert Bullock responds:

The formula in the box on p. 17 of SB *1/83 is incorrect. The formula in line 420 on p. 15 of* SB *1/84 is correct; i.e., the formula*

$$Y1 = Y * SQR\left[(X^2/H-1)^2+(X/Q7)^2/H\right]/X^4$$

is correct. The formula in Mr. Katz's letter as published on p. 67 of SB *1/93 is incorrect, as is the formula on p. 51 of* SB *5/92 given in response to Mr. Hansen.*

I turned the equations in your "Stalking f_3" (*SB* 2/93, p. 24) article into a C program and checked the graph against other "box" programs. (Why the world needs another program to graph speaker responses is another question altogether.) The results didn't match. After checking the calculations, I compared equations in a BASIC program I downloaded from the Madisound BBS.

An apparent anomaly in the equation for α_2 led me to dig up the original articles that you referenced. The equation, taken from Small (in "Vented Box Loudspeaker Systems—Part I: Small-Signal Analysis," JAES, vol. 21, #5, June 1973), equation 23, with your adaptations seems to be:

$$\alpha_2 = \frac{h+\left[(a+1+h^2)Q_L Q_{TS}\right]}{(h^{\frac{1}{2}}d)}$$

Is this correct?

Also, do you know a convenient source for the obligatory Thiele and Small references from JAES beginning in 1971?

Lee Meador
Arlington, TX 76007

Manning Redhill responds:

The equation for α_2 in SB is incorrect. The version in your letter is right. The right equation, using my notation, is

$$\alpha_2 = \frac{h+(a+1+h^2)h^{\frac{-1}{2}}d}{h^{\frac{1}{2}}d}$$

All the T/S references are in AES Loudspeakers, *Vol. 1–25, 1953–1977, which is available in some libraries, from the Audio Engineering Society (60 E. 42nd St., New York, NY 10165–2520), and from Old Colony Sound Lab (PO Box 243, Peterborough, NH 03458, (603) 924-6526, FAX (603) 924-9467).*

JUST THE FACTS

I am disturbed over Marc Bacon's recent article, "The Danielle II" in *SB* 2/94. As a four-year speaker builder on a small budget, I have assembled a toolkit consisting of a variable audio oscillator, *Stereophile* Test CD II, Radio Shack SPL meter and DVM, and software aids CALSOD and Speaker Designer. I read *SB* to learn how other designers, such as Marc Bacon, develop their crossovers, not just the final product. Understanding the thought process will make me a better speaker designer, too.

Marc's comments about the "smear" of polypropylene drivers is a first for me and sounds like marketing hype for FOCAL rather than substantiated scientific fact. For those of us who can't afford FOCAL, does that mean a well-executed design using poly drivers is doomed to smear? I doubt it.

Could you answer the following:

1. Where did you get the driver SPL and impedance data for your CALSOD modeling?

2. How did you know to depart from the D'Appolito crossover recommendations and change to even-order, higher crossover frequency?

3. Why did you include a trap filter in your woofer circuit to reduce the "effect of the resonance peak"? Is that effect audible? Is it worth paying for a 24mH inductor to remove it?

Jerry Exner
Taylor, MI 48180

Marc Bacon responds:

I wrote the articles on the Danielle and Danielle II specifically to describe the design process in preference to that of building speaker boxes. Unfortunately, in the limited space available, it was impossible to answer every reader's question.

Refer to the Loudspeaker Design Cookbook *for the pros and cons of standard transfer functions for crossovers. Although its standard formulas apply only to resistive loads, and are not practical for actual drivers unless zobelled heavily, the discussion surrounding crossover types, such as Butterworth, Linkwitz-Riley, and Chebyshev, is about the best you can find for the price.*

Timbre preferences affect how people regard certain speakers. I prefer good Kevlar® cones to polypropylene. Refer to designer-engineer Keith Johnson's comments regarding polypropylene cones (SB 6/88, p. 9). Kevlar sandwich cones, in particular, combine an excellent stiffness-to-weight ratio to avoid cone breakup with internal damping. However, some excellent polypropylene drivers on the market, especially those with thick, mineral-loaded cones, outperform certain thin-coned, peaky Kevlar drivers. You are not doomed to failure with polypropylene.

Due to market pressures, the price of successful drivers is determined by cost of design, tooling, materials and labor, units sold, number of middlemen, and currency exchange. I would suggest that you fix your budget and size constraints, then divide your budget wisely among drivers, crossover components, damping materials, cabinets, and accessories.

To answer your specific questions:

1. Since I did not, at the time, have access to a gated measuring system that works outside an anechoic chamber, I used driver SPL and impedance gathered from the manufacturer's curves.

2. The Danielle II uses a fourth-order L-R crossover for its excellent transient behavior and tweeter protection. Using a dipole, the radiation pattern is a figure eight with a subjective response dependent on room reflections to provide a sense of ambiance. D'Appolito's criteria for vertical imaging became, therefore, somewhat less important.

3. The woofer crossover is at 400Hz. Using passive circuits, this is much too close to the bass resonance to provide a smooth crossover, since the rate of change of woofer impedance is greater than that of the crossover impedance. Had the crossover been 1kHz, this would not have been a problem. Many well-intentioned amateurs build poor subwoofers by assuming that a textbook low-pass filter will suffice. As a good rule of thumb, either use an active crossover or equalize the bass resonance for woofer low-pass crossovers under 800Hz. You can model this on your CALSOD quite easily by following the excellent instructions packaged with the software.

Thank you for your comments regarding the need for instruction for speaker builders. Not everyone is well-versed in physics, electronics, acoustics, and materials engineering, and those who think they are often have much to learn from others. To make matters worse, marketing hype is rampant in the audio industry. In other less subjective fields, such as pharmaceutics, people would be sent to jail for making claims as widespread as we regularly read in leading audiophile publications, and much of what is passed off as "golden ear" listening is nothing more than savvy milking of an audiophile's pocketbook by appealing to his/her self-esteem. I can therefore understand your skepticism regarding my subjective remarks about Kevlar, and your expressed wish for an honest means of acquiring unbiased knowledge in our common avocation.

Good, honest speaker building is one of the most rewarding hobbies around, but can become very frustrating if you make a few wrong choices and end up with poor results after investing heavily in top-quality components. Probably the most daunting, yet potentially rewarding, part of speaker building is crossover design. I receive more pleas for assistance from people redesigning a textbook crossover gone amok than any other single problem.

To address this need, I plan to provide a hands-on course to answer specifically the kinds of questions you raised, and am marketing it as The Loudspeaker Design Information Series *through Technologie MDB. Since writing the Danielle II article almost two years ago, I've worked part-time to develop their kits, but have tried to maintain objectivity in developing this course to avoid turning it into yet another forum for marketing hyperbole. This, together with the widespread availability of low-cost software, books such as Dickason's* Loudspeaker Design Cookbook *and Colloms'* High Performance Loudspeakers, *and regular reading of periodicals to see how professional designers succeed, should go a long way towards helping the average speaker builder.*

A GRAPHICAL SOLUTION

Over the years I've become increasingly skeptical about using cookbook equations for problem-solving or in simulations, especially in the acoustics field where the original constraints put on expressions are sometimes forgotten or ignored. Also, the intuitive feel about what's going on can be lost by simply plugging numbers into an equation.

In the article "Measuring Speaker Impedance Without a Bridge" (*SB* 2/94, p. 18), Larry Markwalter states that COS ≤ $(A^2 + B^2 - C^2)/2*A*B$, and later that "although accurate, does not provide identification as inductive or capacitive." Since Larry is somewhat of a legend around these parts, I was sure the method was sound. But curiosity got the best of me, and I'd like to share my findings on why this clever method works.

Since we are trying to find the speaker impedance (Z_{SP}), we'll work with impedance, although as stated, voltages will also work.

From the three measured voltages you can find the following:

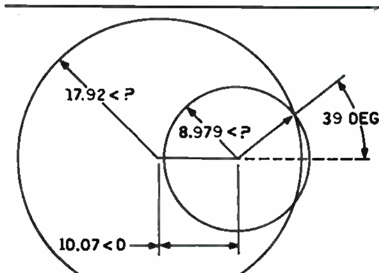


FIGURE 1: Reader Marchese's graphical solution.

(1) magnitude of the total current: $I = V_{RESISTOR} / R$

(2) magnitude of the speaker impedance: $Z_{SP} = V_{SP} / I$

(3) magnitude of the total impedance: $Z_{TOTAL} = V_T / I$.

So we now have the total impedance (Z_T) and the speaker impedance (Z_{SP}), but this is in magnitude only, and no information about the phase angles is known. The total impedance is the vector sum of the resistor and the speaker impedance.

Drawing the impedance graph (see *Fig. 1*), the total impedance (Z_T) can be made with a circle (a magnitude at all angles) whose magnitude is 17.92. The components of this total impedance are a resistor—10.07Ω at a 0° angle—and another magnitude of unknown angle (circle) of the speaker impedance magnitude. The two circles intersect at the points where the total impedance is equal to the vector sum of the resistance and the speaker impedance.

If voltages are used in the analysis, the points satisfy Kirchhoff's loop theorem (the sum of the voltages around a loop is equal to zero). A straight line can be drawn for both ZT and ZSP, and the angles measured. The circles will intersect at two places, except in a purely resistive network. This explains why the impedance can be inductive or capacitive.

This graphic analysis agrees with the equation in the article and gives a basic understanding about why the law of the cosines works and also the reasoning for the erroneous answer.

Paul Marchese
Severna Park, MD 21146

J.L. Markwalter replies:

Paul Marchese's graphical solution to the voltage impedance vector diagram is valid and might well be used when no calculator capable of handling trigonometric functions is available. The mathematical solution is preferable, however, since it avoids the inaccuracies inherent in graphical solutions in the construction of the vector magnitudes and angle measurements. On the other hand, if the calculations are carried out significantly beyond the accuracy of the measured data, no additional error would be introduced in the reduced data.

The mathematical solution of the speaker measurement problem would lend itself nicely to a simple computer program in which the reference resistance value might be entered once, and the three measured voltages entered at each frequency where the impedance value is desired.

POWER SEEKER

I am mainly interested in speaker systems for musical instrument and PA applications. I have read several books on speaker design, but haven't found much information on power handling. One book stated: "More than half the power will go to the woofer."

In a 200W RMS power amplifier, for example, how much output will go to each component in a three-way system? Does a general rule about power output exist? I need to know the power-handling requirements of the midrange and tweeter components in relation to the amp output.

Edwin H. Myers Jr.
Kingston, NY 12401-2729

Bill Fitzmaurice responds:

A moderately loud sound reinforcement system for a rock band—generically, although incorrectly, known as a "PA system"—should produce at least 100dB average output at 8m distance, with at least 3dB additional headroom available, with a bandwidth flat from 100Hz–14kHz. This is equivalent to a component output of 118dB at 1m. While the average home system's efficiency of 90dB/1W/1m *would need over 1kW to produce this kind of output, professional systems routinely enjoy 100dB efficiency ratings, which makes your 200W amplifier adequate.*

Start your search for suitable components with a woofer with 100dB efficiency, 100W or more rating, and extended frequency response (to eliminate the need for a dedicated midrange driver). A 15" size—such as EV force, EVM 15 L or 15B, or JBL E 130 or E 140—is the most common choice.

The wattage requirements of the high-frequency unit you choose is primarily dependent on driver efficiency. If you use a 100dB efficient tweeter with a 100dB efficient woofer, then both drivers should have the same power rating. However, pro PA tweeters are usually horn-loaded and can often have efficiencies above 110dB. The simple rule is that for every 3dB additional efficiency your tweeter has compared to your woofer, you may cut the tweeter's power rating in half (for example, a 200W tweeter with 106dB efficiency, assuming proper level matching of components).

Crossover also affects tweeter power handling. While a 12dB rolloff is adequate for home systems, 18dB is the minimum and 24dB preferred for high power systems. In fact, passive crossovers are rarely used in pro gear of over 100W/channel; active crossovers with separate power amps for each frequency range are the norm. In your example you could use a 200W amp for the woofer and 50W amp for the tweeter.

Regarding amplifier headroom, the cost and size of 400W and larger amps make it difficult to enjoy much, if any, extra amp headroom. With woofers especially, it is not unusual that speaker wattage exceeds amp wattage.

To eliminate clipping which could damage drivers and amps, you need compressor-limiters and output meters to ensure that power amps are not driven into clipping. Ideally, you'd like to have power amps at twice the rated speaker wattage cruising along at half-power; realistically, you'll probably have a smaller amp running at 99% power most of the time, so use a limiter to protect your investment.

I'd recommend you try my Electric Bass Tri-Horn (described in SB *4/93, p. 10). Replace the 5" mid driver with a good horn-loaded tweeter of at least 106dB efficiency and 100W power crossed over at 800Hz–1.2kHz. For superior high-frequency dispersion, mount the tweeter in a separate enclosure on a speaker tripod stand about 7' high; this will let the highs reach the back of the audience without blowing away anyone close to the stage. A similar pair of units formed the basis of my PA and was able to drive a full band even outdoors with audiences of over 500.*

SPICE FACTS OF LIFE

Thanks for your good work and interesting articles. I would appreciate more computer-based articles, such as "Exploring the BUF 124 with PSpice," which I found current and valuable. If possible, please publish some examples of SPICE modeling unconventional enclosures other than closed and vented designs.

Speaker Builder is becoming a serious speaker reference and a great value. While the majority of your readers probably request good audio projects to construct, I think your current balance is excellent.

M. Barabasz
Dandenong, Victoria
Australia 3175

What do you mean "becoming"?—Ed.

I found Richard Campbell's recent article, "Exploring the BUF 124 With PSpice" in *SB*

Continued on page 61

STRUCTURAL HEARING: TONAL COHERENCE IN MUSIC — BKD17 — $12.95

Felix Salzer

In this two-volumes-bound-as-one book written by a pupil of the late Heinrich Schenker, the phenomenon of tonal organization in Western music is explored by means of a detailed analysis and discussion of more than 500 musical examples ranging in time from the Middle Ages to such modems as Bartók, Hindemith, Prokofieff, and Stravinsky. Used in all important music schools, this book starts with basic definitions and simple examples, and trains the reader not only to hear successions of tones, melodic lines, and progressions of chords, but also to understand their structural coherence and significance. It is invaluable for musicians and all who are seriously interested in music, whether as a student, critic, performer, or conductor. 1952, 1982, 672pp., 5⅜ × 8½, softbound.

HARMONICS: A FIELD HANDBOOK FOR THE PROFESSIONAL AND NOVICE — BKS49 — $24.95

Mark Waller

Many operational problems can be solved through an understanding of power system harmonics. As life/safety issues become more and more important in the world of electrical power systems, the need for harmonic analysis becomes ever greater. This book is the essential guide to understanding all of the issues and areas of concern surrounding harmonics and the recognized methods for dealing with them. Covering nonlinear loads, multiple PCs, notching, K-factor transformers, and much more, this volume is a useful tool for system and building engineers, electricians, maintenance personnel, and all others concerned about protecting and maintaining the quality of electrical power systems. 1994, 132pp., 7⅜ × 9¼, softbound.

WHAT'S A SEQUENCER? — BKHL5 — $5.95

Greg R. Starr

Sequencers are becoming a fact of life in today's music scene. These wonderful tools are now available to you at home, where you can use them to produce exhilarating and inspiring music without alot of fuss and without having a big-budget recording contract or large family inheritance. If you own a sequencer, this book will cover both the basics of using it and the more advanced goodies that are available on some models. If you don't have one and are thinking of buying one (or a keyboard with one built-in), you'll find enough info here to ask the right questions to decide which model is right for you. 1990, 62pp., 6 × 9, softbound.

THE PENGUIN GUIDE TO JAZZ ON CD, LP & CASSETTE — BKP9 — $22.95

Richard Cook, Brian Morton

You'll find the answers to all your jazz recording questions in this volume, the first comprehensive, critical listing covering all jazz recording currently available. From the earliest days of the Original Dixieland Jazz Band to the most recent recordings by today's young stars, each entry has been carefully rated by leading jazz critics Richard Cook and Brian Morton. Included are musical and biographical info; full lineup details; recording information; accurate label/number details; authoritative critical ratings; full index; and special sections for "Anthologies" and "Various Artists" collections. United Kingdom, 1992, 1287pp., 6 × 9¼, softbound.

HOW TO TEST ALMOST EVERYTHING ELECTRONIC — BKMH10 — $14.95

Delton T. Horn

This third edition has all the practical, hands-on guidance you need to troubleshoot efficiently with today's electronic test equipment. Filled with helpful examples of typical circuits and easy-to-follow instructions, the book shows you how to pinpoint problems in all types of electronic systems, from TVs to VCRs to computers. In addition to learning how to use multimeters, oscilloscopes, signal generators, logic probes, voltmeters, and ohmmeters, you'll be taught how to repair mechanical components and how to use flowcharts when troubleshooting. 1993, 326pp., 7⅜ × 9¼, softbound.

LOUDSPEAKER ANALYSIS AND DESIGN PROGRAM (LADP) SOFTWARE FOR WINDOWS — SOF-LAD1B3G — $134.95

NEW THIS ISSUE

G&J Technologies

This new loudspeaker engineering tool for Windows combines visual interface and electroacoustics into a flexible loudspeaker designer and analyzer. Modeling, small signal analysis, and design for ported (standard or isobarik), sealed (standard or isobarik), bandpass, tubed, and horn-loaded enclosures are provided. For ported enclosures, the design can be changed to see the effects of different port frequencies and/or box volumes. Bandpass enclosures of orders four through seven can be designed and effects of changing the rear and/or front box tuning, or changing the rear and/or front volume, can be tested. Tubed enclosures, ideal designs for truck speakers, can be designed for any diameter. Large signal analysis is provided for ported (standard, series isobarik, or parallel isobarik) and sealed (same) enclosures. An active high pass filter (orders one through four) can be used to act as a subsonic or rumble filter. Large signal analysis shows the relative on-axis response (SPL) at one meter, as well as the maximum input wattage per frequency. Port dimension calculation for all designs and one to four ports is also available, as is enclosure dimension calculation for all types covered.

LADP also can design high- and low-pass filters of order one through four. Individual types include Linkwitz-Riley, Chebyshev, Legendre-Papoulis, Butterworth, and Bessel. Bandpass filters of order one or two can also be constructed for all the above types except Bessel. Small signal analysis can show up to three filters at a time and include the dB response and phase angle. Further, the polarity of the high-pass filters can be reversed to show the effects of in-phase and out-of-phase filter designs.

Loudspeaker directional quality can be seen graphically as a polar plot. The off-axis response is shown from -90 degrees to +90 degrees. This can be a useful tool in determining the upper cutoff frequency of a low-pass or bandpass filter, and the high frequency "beaming" effects of the speaker can also be tested. If you have a sinewave generator, voltmeter, frequency counter, and 1000-Ohm and 10-Ohm resistors, you can also measure Thiele-Small parameters. A graph of the voice coil impedance in free air can be shown, as well as the output parameters: Vas, Qts, Qes, Qms, Ras, Mas, and Cas.

Some basic architectural acoustics such as room reverberation, impulse response, and room modal frequency distribution can also be explored with LADP. A graphical room modal frequency response will help the user to determine how the room reacts to certain frequencies, as will the decay response to a tone burst for frequencies at 125, 250, 500, 1000, 2000, and 4000Hz.

On-line help provides a supplement of information for each dialog box. It also contains helpful hints for designing, building, and testing enclosures and crossovers, as well as general car stereo information. Background information on acoustics, electrical/stereo components, and speaker designs is also available, as are a glossary, program overview, program utilization info, graph interpretation guide, and definition of every prompted input.

NEW IN LATEST VERSION: addition of network drive search at installation time; replacement of vented enclosures with eq by sixth-order active-filtered enclosures, schematic and print capability; replacement of sealed enclosures with eq by fourth-order active-filtered enclosures, schematic and print capability; addition of cone excursion and port air exit velocity calculations to the Loudspeaker Sound Pressure option; addition of summed loudspeaker(s)' acoustical response and loudspeaker enclosure system phase modeling into the Crossover/Driver Interaction option; improvement of bandpass enclosure modeling transfer functions, with more custom designing options for bandpass analysis; improvement of Windows system resource usage in the repaint routines; and addition of passive radiator design and analysis.

All output can be printed to the Windows default printer in either portrait or landscape mode. System requirements: 2Mb RAM, 3.3Mb hard disk space, MSDOS5.0 or later, Microsoft Windows 3.1 or later, math coprocessor. 2 x 3-1/2" DS/HD.

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$100.00–199.99	5.00	20.00	15.00	20.00	30.00	40.00
> $200.00	6.00	30.00	25.00	30.00	40.00	50.00

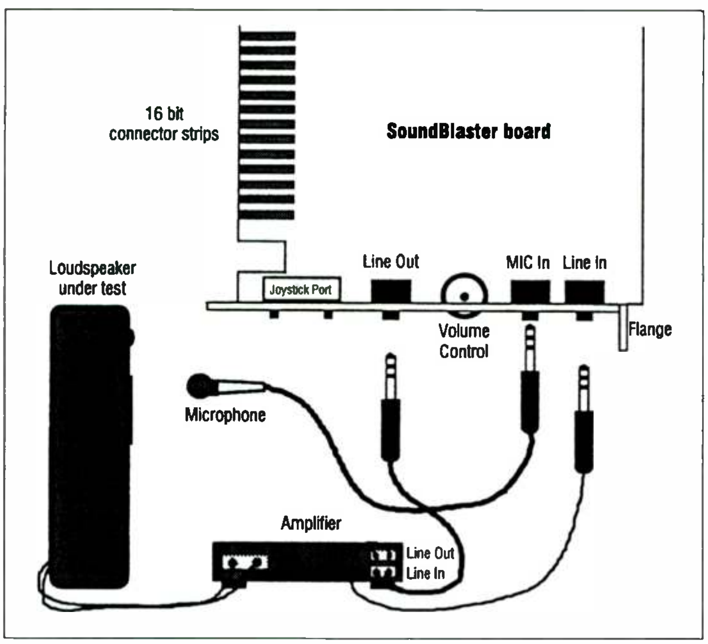
NEW THIS ISSUE

AIRR (ANECHOIC IN ROOM RESPONSE) SOUNDBLASTER SOUND CARD SPEAKER RESPONSE SOFTWARE — SOF-AIR1B3G — $49.95

Julian J. Bunn

NEW FROM SWITZERLAND! AIRR is a program that offers a cheap alternative to loudspeaker measurement for owners of PCs equipped with a SoundBlaster or compatible sound card. Based on the principle of pulse mode excitation of the loudspeaker followed by Fast Fourier Analysis, AIRR features full color real-time plotting of both the digitized pulse received by the microphone and the frequency response of the loudspeaker system. AIRR is controlled by a single key stroke and data entry at the keyboard, and full on- line help is supported. You can toggle between several modes of operation, which include Loudspeaker/Amplifier response measurement, Time/Frequency averaging, Square/Sin(x)/x pulse types, and Pulse On/Off. For anechoic measurements, cut markers allow a window to be placed over the time data in order to remove the part due to room reflections. Frequency response spectra obtained can be stored as ASCII files on disk, which then can be subsequently read back in by AIRR and subtracted from and/or overlaid on the current spectrum. Using the subtraction feature allows corrections to be applied for amplifier or microphone response. The AIRR display itself can be "captured" into a PC Paintbrush (PCX) file, which may then be included as graphics in a document.

The AIRR software includes operating instructions combined with on-line help, last minute changes and instructions, the AIRR article itself from *Speaker Builder* 8/94 (in Word), and a sample file containing microphone response data. AIRR requires a PC equipped with a SoundBlaster 16 sound card (or compatible, such as the MediVision Pro 16); VGA screen with at least 640 x 480 resolution, preferably color; microphone (usually supplied with the sound card, although a calibrated microphone such as Old Colony's Mitey Mike is often desired); and loudspeaker to test! Math coprocessor not required. 1 x 3-1/2" DS/DD.



THE COMPLETE GUIDE TO HIGH-END AUDIO — BKAP1 — $29.95

Robert Harley

Written for beginners and experienced listeners alike, this is the first book that tells you everything you need to know about buying and enjoying high-quality hi-fi. In 13 easy-to-read and fact-filled chapters, the author, a Consulting Technical Editor for *Stereophile*, shares secrets on how to get the maximum enjoyment out of audio. This book has it all, from nuts and bolts tips on hi-fi equipment to suggestions on becoming a better listener.

With this book, you'll find out how to get high-end sound at low prices; identify the weak links in your present system and upgrade where it will do the most good; discover how to tweak your system to get maximum performance from equipment you already own; learn how to read equipment reviews and determine why the components that sounded so good in the showroom don't sound so good at home; identify the qualities that will make you a more appreciative and perceptive listener; and best of all, learn how to keep hi-fi fun! 1994, 450pp., 7 x 10, softbound.

CHRISTMAS IN EARLY AMERICA CD: 18TH CENTURY CAROLS AND ANTHEMS — CDAL1 — $16.95

The Columbus Consort

This unique CD will make a great stocking stuffer for any audiophile, including you! Recorded in The Netherlands and produced in Germany by Channel Classics, *Christmas in Early America* brings a new sound to the holiday season, as well as an interesting and superbly crafted conversation piece. The Columbus Consort consists of the delightful mix of two violins, cello, bass, organ, and vocal quartet. Thirty-page booklet in English/French/German includes historical background as well as lyrics. 1993, 55:40.

Contents: [1] "How Beauteous Are Their Feet" (Supply Belcher). [2] "How Beauteous Are Their Feet" (Samuel Holyoke). [3] "Anthem for Christmas" (Benjamin Carr). [4] "If Angels Sung a Savior's Birth" (Joseph Stevenson). [5] "Th'Almighty Spake, and Gabriel sped" (Samuel Holyoke). [6] "Comfort Ye My People" (Samuel Holyoke). [7] "Hark! The Glad Sound" (John Jenkins Husband). [8] "While Shepherds Watched Their Flocks by Night" (Musgrave Heighington). [9] "While Shepherds Watched Their Flocks by Night" (Jacob French). [10] "While Shepherds Watched Their Flocks by Night" (William Billings). [11] "Unto Us A Child Is Born" (Johann Friedrich Peter). [12] "Prince of Peace, Immanuel" (John Antes). [13] "Hail Infant Newborn" (David Moritz Michael). [14] "To Us A Child Is Born" (Johannes Herbst). [15] "Thou Child Divine" (Johann A.P. Schulz). [16] "Meine Seele erhebet den Herrn" (Jeremiah Dencke). [17] "Hosanna! Blessed He That Comes in the Name of the Lord" (Christian George). [18] "Magnificat anima mea Dominum" (Charles Theodore Pachelbel).

SOUND SYSTEMS FOR YOUR AUTOMOBILE — BKS48 — $16.95

Alvis J. Evans, Eric J. Evans

Whether you're starting from scratch or upgrading, this book will show you in easy-to-follow steps how to plan your system, choose components and speakers, and install and interconnect them to achieve the best sound quality possible. Installations on specific types of vehicles are discussed, including separate chapters on coupes and sedans, hatchbacks, pick-up trucks, sport utility vehicles, and vans, so you're sure to find the information you need for your vehicle. Also included are a complete glossary and an entire chapter devoted to speakers. 1993, 124pp., 6 × 9, softbound.

AMAZING KOONCE UTILITY DISK #1 — SOF-KUD1BXG — $7.95

G.R. Koonce

This valuable disk contains two programs to benefit just about anyone! CONVERT1.EXE displays instantaneous, simultaneous English and metric values for linear, area, and volume measurements in inches, feet, yards, millimeters, centimeters, and meters. The volume screen also includes liters. On-screen help (hardly needed) is available. PRECRES2.EXE requires VGA to display its amazing wares: instantaneous interpretation of metal film resistor color bands! You can either enter a value and see the band colors appear on the screen, or enter the colors of your mystery resistor and have the program tell you its value. Good for up to 10 megohms, and includes three significant digits, multiplier, and tolerance. Printer option. IBM, requires VGA, 1 × DS/DD—PLEASE SPECIFY DISK SIZE.

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$50.00–99.99	4.00	15.00	7.50	15.00	20.00	30.00
$100.00–199.99	5.00	20.00	15.00	20.00	30.00	40.00
> $200.00	6.00	30.00	25.00	30.00	40.00	50.00

Continued from page 58
3/94, quite interesting from a surface-only point of view. Practical hands-on discussions showing how SPICE can help analyze audio circuits are basically a good idea, and over time have the potential of greatly increasing an audiophile's bag of tricks. Yours had the possibility of this, but was caught short by some unrealistic premises.

Some cardinal rules of SPICE simulation have not been satisfied in the article. SPICE (which, of course, includes PSpice) gives results only as good as the basic premises of an analysis. If the analysis is based on incomplete or incorrect data/conditions, then the results may simply not be applicable as desired. More specifically, if inaccurate circuit models are used for an analysis, then the results can be meaningless. They could be perfectly valid for the conditions as modeled, but unlike what was actually intended. Thus erroneous conclusions can be drawn.

In the article, models for transistors other than those used in the original circuit have been substituted for the specified units. Although this serious caveat is mentioned almost in passing towards the end of the article, no mention is made of its implications. In fact, the JFETs (as just one example) are in substantial error insofar as emulating the original types.

For example, the real BUF 124 input JFET transistors are high Gm, relatively high capacitance types, whereas the 2N5457/59 and 2N5460/62 are medium-to-low Gm units, with <10pF of input capacitance. The circuit employs different devices for the cascode transistors, while the circuit model attempts to emulate this by using scaled Gm types from the same (incorrect) model family for J1/J2 and J3/J4.

For situations where specific transistor models are not available, the commercial version of PSpice includes a "Parts" program, which can be used to build a model from data sheet input. In fact, many of the models distributed with PSpice were developed this way. And, the current release (as of May 1994, version 6.0) contains a new library of Japanese JFET transistors, including many used in the BUF 124.

To summarize, the article simulations show a circuit which may operate and function in some regards like a BUF 124, but it does not represent a valid simulation of the actual circuit. While it might be a useful alternative, it is simply not the same as the real BUF 124 until it uses proper models. Readers should understand these points, which amount to "SPICE facts of life."

In the future I suggest you show the buffer operating in an actual filter, as opposed to stand-alone operation. One of the error sources with Sallen-Key filter amplifiers is output impedance, so it would be interesting to explore what effects the finite impedance of the circuit model has on a typical filter circuit. The Laplace function in the commercial PSpice can be used to build an ideal filter transfer function, which can then be compared to the simulated circuit's response for filter accuracy.[1] This powerful analysis tool can be used to quickly zero in on sources of error within filter circuits.

Walt Jung
Fallston, MD 21047

REFERENCE

1. Walt Jung, "SPICE Technique Compares Frequency Responses," *EDN*, Nov. 25, 1993.

Contributing Editor Richard Campbell replies:

Mr. Jung's remarks are quite correct, but I think the article's intent has been masked by a considered reaction from one of the great *experts in the field. The piece's intention was primarily educational and introductory. It was inspired by reading the Borbely-Gaertner article ("Modular Active Crossovers," SB 1/94, p. 20) and realizing that SB readers tempted to try PSpice can do so with this circuit, which actually has a physical embodiment in a useful part which they read about a couple of months back, and could build.*

I never had a real BUF 124 to play with, but I certainly intend to do so, and eventually produce an update relating the real model to PSpice. Obviously, computer simulations are only that—simulations. The real thing will always be full of little surprises because it's just too much trouble to account for every nitty-gritty detail (although integrated circuit designers do remarkably well). I was most concerned with authenticity when I substituted batteries for the LM336s, but if I had tried to use PSpice models for those two parts, the student's version of the program would have failed, as the semiconductor node count would be exceeded.

I selected the substitute JFETs only to establish a proper I_{DSS} ratio between the upper and lower cascode pairs, but, as Mr. Jung points out, they are definitely not like the ones specified in these parts, which Mr.

Borbely says are made by Hitachi. Again, the spirit of the piece was to show reasonable simulation results even with off-the-shelf transistors. The quiescent currents were all very close to the expected value, and the raw DC imbalance was less than 1V at the output prior to adjusting the pot.

I grabbed an off-the-shelf pair of output transistors which I sometimes use. I suspect that any pair with an HFE over 100 and reasonably low collector-to-base capacitance would provide the same audio output current boost in the passband. Phase shift at the -3dB point would be a different matter.

I'll buy Mr. Jung a meal at Bertucci's before the next AES meeting if the important global parameters of the BUF 124 simulation are more than ±12.2018454% different (±1.0dB) between my parts and Borbely's parts. Specifically, I refer to quiescent currents (I get to balance the pot), output distortion as a function of level, output impedance, and phase shift at 20kHz.

I did not wish the reader to resort to the commercial version of PSpice for models. We have one commercial PSpice at Worcester Poly-technic Institute, but it has not been updated to v. 6.0. I am happy to know the Hitachi JFET models are in that release. (Maybe some wealthy benefactor is reading this.)

Mr. Jung's suggestion for the future is a really good one in that a constant (rather high, actually, at 177Ω) output resistance will definitely influence the behavior of the following Sallen-Key stage, particularly in the high-pass configuration. If I take the example of Fig. 5.9(B) *on p. 169 of Mr. Jung's own excellent book (*Audio IC Op-amp Applications, *Sams, 3rd ed.), the input capacitor has a reactance of 177Ω at 45kHz. Since this is a 1kHz filter, I wonder how significant it is. For low-pass configurations it's very much like using 1.5% resistors. I shall explore his suggestion using the commercial version. It is also important to investigate component tolerance, which the commercial version does easily.*

Since I have the floor, here are a couple of notes which may be important to some readers:

1. The low-priced version of Intusoft© "IsSpice 1.41" is a faithful PC implementation of SPICE 2.G with unlimited nodes within the computer's memory capacity. However, like its parent, it knows not of at least half the parameters in the JFET model which I show, illustrating how MicroSim has progressively refined the modeling of semiconductor devices. If some reader is using that software, I can provide working JFET models.

2. The BUF 124 PSpice listing as printed has two problems: It should have been typeset in a monospaced font, and the scattered + signs should only appear as the first character of a statement continuation line.

I am indebted to Mr. Jung for his interesting letter and valid finger-pointing where I have appeared too cocky about SPICE simulations. It is just another software tool providing only an indication of performance. My models do not represent the real devices screened by Mr. Borbely. You get these in your BUF 124 kit of parts you can order from Welborne Labs (683 S. University Blvd. #70, Littleton, CO 80122).

One of my heroes—name forgotten—is the engineer who was at a famous meeting in 1961. The boss said, "Well, you heard the President, what do we do?" This guy pulled out his slide rule, grabbed a pencil and an old envelope, scratched a bit and said, "I figure if we develop a rocket with 1.2 million pounds of thrust with 560,000 gallons of hypergolic fuel, it will lift 83,000 lbs. to Earth orbit and this payload can have a command module, a service module, and a retrievable four-legged lander and we can...." Honest. A guy actually drew that out on an envelope in ten minutes. Nine years and 6.5 billion dollars later, it became a reality; it didn't look quite the same, but it was close.

I'm a back-of-the-envelope kinda guy, at least to start with. When I first saw Mr. Borbely's circuit I recognized it as similar to an old scope vertical input amplifier I used to fight with, and I got sucked in. Between Mr. Jung's letter and this reply, the reader will better appreciate the technical issues surrounding any simulation, especially a simple-minded one like mine.

Loudspeaker History

Continued from page 21

duced another model, the AR-1W, which contained only the woofer from the AR-1, without the 755A tweeter. Those who desired a better tweeter could place the Janszen 130, a four element electrostatic array, on top of the AR-1W enclosure.[11]

In 1957 another Boston firm, KLH, was formed under the guidance of Henry Kloss, who was a co-founder of AR. After his tenure with KLH and its sale to Singer Corp., he created a third Boston loudspeaker firm, Advent Corp. The KLH-1, introduced in 1958, was manufactured under license from AR and contained two 12", high compliance woofers in a floor-standing enclosure, along with a cutout for the same Janszen electrostatic elements used with the AR-1W.[11] Throughout the 1960s the AR and KLH loudspeaker systems were known for their very low sonic coloration and high accuracy, characteristics which became known as the "New England Sound" and the "Cambridge Sound." For many years the AR-3A and the KLH-6 were standards by which other dynamic bookshelf systems were judged.

In 1962 Electro-Voice introduced a bookshelf speaker—which they called an air suspension loudspeaker—using a high compliance woofer. AR filed suit against Electro-Voice for patent infringement.

A series of court battles followed and in early 1963 Villchur's patent was invalidated. EV's lawyers convinced the court that Olson's, not Villchur's, work was the basis for their design. The court failed to consider the fact that Villchur invented the high compliance woofer, which was also used by Electro-Voice. Villchur, however, decided to forego further litigation because of the legal costs involved.[11] Thus, in 1963 the acoustic suspension loudspeaker became a public domain invention.

Tony Hofmann (the "H" of KLH) developed the original formulas, which became known as "Hofmann's Iron Law," for acoustic suspension loudspeaker design.[11,12] In 1972 American Richard Small—in a series of articles appearing in the *Journal* of the Audio Engineering Society—analyzed and described the mathematics of closed-box loudspeaker design.[16] (These were reprints from the IREE Journal of Australia, originally presented in 1969 and published in 1970.)

Small showed that closed-box loudspeakers are second-order (12dB/octave), high-pass filters. His mathematical analysis, derived to some extent from the earlier work of J.R. Benson, let designers accurately predict the system response based on the woofer's electrical and mechanical characteristics and enclosure size.[10] These papers are now considered landmark contributions to the science of loudspeaker design, and most contemporary closed-box designs are based on the work of Small.

In Part 2 we will continue our look at the evolution of speaker enclosures with a focus on port, or vented, models—from the early bass reflex units to today's high-performance multi-way systems.

Damping Factor

Continued from page 31

"figure of merit." The Heath and Sony units have grossly different resonant frequencies (24.2Hz and 109.9Hz, respectively). Since lower resonant values are certainly more noticeable to the listener, systems that have a low damping constant connected with an equally low resonant mode yield a "double whammy."

My proposed standard is the J coefficient, which is the critically determined damping constant divided by the natural logarithm of the resonant frequency. Using this criterion, the Heath system has a J value of 21.4/ln(24.2) or 6.72 and the Sony unit boasts 147/ln(109.9) or 31.3. The free-air 8" speaker rates a 2.64 J coefficient.

This coefficient, as a description of the enclosure, driver, crossover network, and feed cables interface and interconnections, provides a defined and quantitative value for comparing systems with respect to damping and resonance characteristics. The greater the J value, the greater the resistance to input transient distortion.

WRAP-UP

The DF value calculated from the speaker and amplifier impedance ratios has little rigorous significance. If you wish to know the true damping constant of a system, then you must use a method such as the technique I've described. The ultimate value of these measurements, and the system design objectives, is to ensure that the integrated system has a greater DF than the speaker system alone. This ratio—of the integrated speaker

system DF to the open circuit voice coil DF of the driver mounted in its enclosure and connected to the applicable crossover network—is the real figure of merit for these systems, and this ratio should always be greater than unity.

Spline Corner Joints

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ASSEMBLY

If you're using a blade that produces a flat-bottom cut, the spline will fit snugly (*Photo 5*, left). If your blade leaves a ridge at the bottom of the cut, the fit will be looser (*Photo 5*, right), but either one is acceptable. The flat-bottomed cut looks better, if anyone ever sees it. *Photo 6* shows the exterior results.

You can also use the biscuit corner joint jig described last issue to make your spline cut. The better biscuit jointers have retractable positioning spikes that protrude from the face of the fence on either side of the saw blade slot. After retracting these spikes, simply slide the jointer along the face of the boards (*Photo 7*). The setup we discussed last issue applies here as well. Proceed slowly, making certain that you will not lose your balance, while keeping the biscuit joiner firmly against the jig and your board.

Spread a thin coat of glue on the joint faces. Run a small bead of glue along both edges of the spline cut. For best results, be sure to apply the glue within a few hours after making the cuts.

Audio Software

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identical distances between the listener and the two speakers by selecting the Symmetric room option, though this is somewhat restrictive. (4) A useful feature which Dr. Staggs did not mention is the ability to layer up to three graphs, making it easy to compare alternate layouts.

The Listening Room cannot tell you conclusively what positions will sound best, since the perception of reproduced sound is extremely complicated and only partially understood. TLR will help simplify selecting good room positions based on an analysis of standing waves and early reflections, but let your ears make the final decision.

Finally, I must credit Bill Fitzpatrick of Sitting Duck Software with conceptualizing The Listening Room and with writing most of the original PC version.

Ralph Gonzalez
Smiths Parish, Bermuda FL05

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WIDER, DEEPER, MORE HOLOGRAPHIC SOUNDSTAGE: Room Acoustics Handbook $14.95, Monarchy 22a DAC $695, Level 1 Mod $295, Digital Cable $125, Loudspeakers from $995. ***MACH 1 ACOUSTICS***, RR #2, Box 334A, Wilton, NH 03086, Phone/FAX (603) 654-9826. 8/94

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TWEETERS: 32 Philips ribbon AD21600RT8, $50 ea.; 12 Dynaudio D28-AFSQ, $30 ea. ***JOHN ÖTVÖS***, RR# 4, Brighton, ON Canada K0K 1H0, Phone/FAX (613) 475-3633. 6/94

ACT AUDIO GIVES GOOD SOUND! Computer analysis of your listening room. Send SASE for details. ***ACT AUDIO***, 619 Moon Clinton Rd., Corapolis, PA 15108. 8/94

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FOR SALE

Power amplifier: MFA D-75A, 75W/channel, $1,100; Cary SLA-70, $625; preamplifier: Counterpoint SA-3.1, $450; loudspeaker: Acoustat II, $600; Magnepan MG111A, $1,300; Harbeth Ls3/5A, $600; CD player: Loxman D-1056, $450. Andy Halim, 110 East Foster Ave., #204, State College, PA 16801, (814) 861-6519.

Raw speakers, one only of each: B&W TP26 1" dome tweeter, Koss Tweeter, Koss 8", Polydax 8" TPX, B&W MR100 4" 8Ω Bextrene midrange, 10" Philips AD10100 W8AH, Hitachi 10", 5", aluminum 1" titanium, trio horn tweeter, Yamaha midrange dome, reasonable prices, Jerry Feldstein, (718) 365-6343, eves., 7-10 P.M., EST.

Two JBL ProLine 10" 8Ω woofers, $75. Two JBL ProLine 16Ω horn drivers, $150. Kenwood KM107 150W per channel basic stereo amp, $185. All plus shipping. Stan, (216) 288-9480.

Focal Audiom 15VX woofers, $700/pr. (new); Dynaudio 15W75, $180/pr. (new); Power Wedge Model 1, $250; Carver SDA490T tube CD player, $300; 12 foot pair bi-wire Ocos with Kimber Pair Postmaster, $200. Mike, (801) 580-8904 (UT).

Focal 10K515T woofers, used only to generate the (matched) response curves. My wife says a TL will be too big, so.... Call and make an offer. Dan M., (206) 874-8411 Home, (206) 803-6410 message.

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ReVox B77 open reel recorder, $750. Denon DP-2550 turntable, SME 3990-III arm, Shure V15 cartridge, $300. Yamaha DSP-1 soundfield processor, $495. dbx 124 4-channel dbx-II processor, $150. Nakamichi 700-II 3 head cassette recorder, $400. Audio, video tape, CDs, more. Stamp gets list. R. D. Darroch, 1807 Elm Crest Dr., Arlington, TX 76012.

Dyna equipment, PAT-4, all tantalum and poly cap upgrade, gold, $100. Stereo 80, good, $95. AF-6, gold face, very fine, $125. SE-10 Nu Silver, $175. Matching Nu Silver FM-5, good, $95. FM-5, gold, not working, $35. Koss 4DS digital delay, $75. B. Brown, RR 1, Box 50, Valley Springs, SD 57608, (605) 582-6762.

KLH three-way speakers, $50. Four matched 6CA7 and two brand new 12AX7 matched tubes. AudioSource VS-One center channel speaker, $25. Spare drivers, tweeter for VS-One, $15. Tubes are $100 for all 6. Best offer or $125 takes all. You pay shipping. Deric Christensen, 2310 Rock St., Mt. View, CA 94043, (415) 940-5548.

Matching pair of vintage KEF model K-2 Celeste MKII speakers. Made in England, sold out of Hong Kong by Radio People, with serial numbers, rare collector units, cleaned, in good shape, real wood. Like new condition, very desirable for the collector. Price $200, I pay shipping. David, (305) 931-7131.

Audio Research D100, $395; HP 120B, $60; Sony 2251LA turntable, $150; Infinity Black Widow arm, $70 VMPS Sub, $195; Koval QUAD Mod, $100; Heath IG5257, $75. Tom, (415) 851-2779.

Sony TA-E1000ESD DSP preamp: parametric EQ, dynamics, surround all processed digitally Pro-Logic, 10 soundfields, 12 inputs, A&V switched, digital inputs for onboard D/A, S-VHS, five-channel outs plus 80Hz subwoofer out, learning remote, black finish, wood sides, 100dB S/N digital, EQ/effects employable in recording path, $850. Ray, (402) 466-2920.

Eight JBL 117H-1 8" woofers, factory sealed, $45 ea.; two 2404H biradial horns, $250/pr. Electro-Voice: two ST350B, $125/pr. Dynaudio: four 21W54, $65 each; two D-21AF, $20 each; two D-52AF, $50 each; two D-54AF, $60 each; two D-76AF, $45 each. Clay Jordan, (404) 969-8319, 6-9 P.M. EST.

Counterpoint SA-20 power amp, hybrid design, with box, manual, $1,000; pair Yamaha F1030 mono electronic cross-overs, two-, three- way, 12-, 18- dB/octave, balanced or regular, phase switch, variable, rack mount, $600/pr.; Pioneer Spec I preamp, $150; Technics 5550 receiver, $100; Altec 806 drivers, JBL 2370 and 2425 drivers. David, (914) 688-5024.

Gemini cabinets, black oak with stuffing, OFC wiring harness and cups. $160 shipped. Phil, (620) 579-1446.

Audax HD12-9D25, $20/pr.; Audax 1½" dome M1D5, $30/pr.; one new Focal 10V516, $75, EV X-36 cross-overs, $25/pr.; University N-2A, N-2B cross-overs, $25/pr.; University SAHF Comp. driver with H-600 horn, $35; Powervar 13 amp toroidal power line conditioner, 32 lbs., $65. All plus shipping. Jim, (708) 425-6719.

Pair Dynaudio 21W54, never used, $75; pair Dynaudio 54AF, slightly used, $25; both pairs, $90. David, (201) 963-6836.

Four 15" Eminence, $30 ea.; two Altec 415 BiFlex, $50 ea.; two Altec 603, $30 ea.; one Altec 515B, $50; one Altec 416A, $30; two Altec 811B w/ driver, $50; two JBL bass enclosures for 15", $50 ea.; JBL 2070 enclosures, JBL 2220, $700/pr. Ric Mitchell, 65 La Vista Grande, Santa Barbara CA 93103, (805) 966-9446.

Decoursey cross-over, model 120, two-way, 450Hz, 30Hz filter, for $200 or trade for pair of Dynaudio 54. 300B single-ended monoblocks, all regulated DC supplies, MIT caps, $1,500/pr. Eduard, (310) 395-5196.

Dynaudio variovents, $15/4. Four JBL L-40 tweeters, $10 each; four woofers, $25/pair (need new surrounds); four crossovers, $5 each. Klipsch KG-2 tweeter and 10" passive, $5 each. Grenier oak cabinets with clear finish, heavy bracing, cut for D'Appolito M-T-M with 5¼" drivers, $80/pair. Rick, (919) 662-5253 until 11:30 P.M. EST.

Ace Audio two-way electric cross-overs, model 6000-6 6dB with 900, 2000, 2500, 3000, 4000, 5000hZ plug modules, $160. Also Model 6000 12dB 800, 2400, 2800Hz with plug modules, $160. VMPS large subwoofer (cabinet ok), hardly used, must take away $200. Charles Pelosi, 29 Meredith Rd., Piscataway, NJ 08854, (201) 572-0979.

Audax 34mm (HD13D34) voice coils, $10 set; Marchand XM1 24dB crossovers, $25 set; Numark DM1850 mixer, $100; DRAM 256 K-byte SIMMs, $5 each; AMD 386DX-40, $30; HP41C card reader, $40. Dan, (801) 224-8080 (work) or (801) 225-8577 (home), 1768 N. 980 W., Orem, UT 84057.

Dynaco AF6 AM/FM stereo tuner. Neatly built in 1982 from a kit packaged by Stereo Cost Cutters. AM section works fine. FM IF and multiplex sections never aligned, tuner not used. In excellent apparent condition, but offered "as is." With manual, $100 + shipping. John Sehring, Box 373, Baker, MT 59313, (406) 778-2452.

Two DH-200 monoblocks, $250 in mods, $550/offer. (817) 244-3913

Pair Altec 420Y's in Altec cabinets. Drivers in excellent condition. Cabinets show the ravages of time, but are refinishable. $400, you pick up (Decatur, IL). Bill (217) 429-8141, FAX (217) 429-8266 any time.

Two used Morel MDM75 domes, $25. Two used Peerless 8" T0205F, $20. Ditto, new, $12. Four used Peerless 8" TX205F, $120. Two new Vifa 3" D75MX domes, $10. Four new Motorola 2"x6" piezo horns, $5. All in excellent condition. Plus UPS. John, (603) 643-5567.

Tektronix 2213A 60MHz dual trace oscilloscope, 2 x 10 probes and accessory pouch, Tucker price, $895, asking $695. LOFTECH TS-1 audio oscillator with frequency counter and dB meter, asking $100. Burr-Brown PCM63P 20-bit D/A evaluation board system, new $300, asking $100. Crystal Semiconductors CDB5328 A/D evaluation board system, new $300, asking $100. 3-dbx dynamic range expander, asking $50. OTL output tubes, 25 6AS7G, 15 new, $75. Ashby, (305) 475-5225.

ESS AMT-3 speakers, $220/pair. JBL 130A reconed as D130, $75. University N-3 crossover, $25. RTR ESR6, $250/pair. All plus shipping. Mike, (612) 475-2993.

McIntosh speaker manuals and literature. SASE for extensive list. Original owners manuals: Bozak B-302A, $3; B&W 801, $5; B&W DM6, $5; Fisher XP6B, $3. General Radio 1304A audio oscillator (20-20k), exc., $50. Roger Russell, 366 W. Hornbeam Dr., Longwood, FL 32779, (407) 862-5143.

Audio Research LS2 preamp, $850; Forte Model 3 (200 W.P.S.), $600; Dynaudio Esotec D-260, $100/pair; Focal 4VLLL polyglass mid., $60/pair; Legacy Model 1, cost $1,800, sell $850; Peerless CC1727 10" woofer, $100/pair; MIT 330 interconnect, $45. Everything absolutely mint. Walt, (215) 368-8981 EST.

Four JBL LE10 baskets, $40 each or $120 all. McI C-28 preamp, $495. Focal 8K415S, $140/pair, as new. Jerry, (601) 264-6971.

Audio Research L3-2B, mint, $1,900; Boulder 250 AE, $2,500/pair; Pioneer Series 20 Class A amplifier, excellent, $400; JBL 4312 (L 100 studio version with better midrange), $450/pair; Fisher 440T solid state receiver, super sound, recent check-up, $200. Steve, (203) 397-3888 (CT), machine after fourth ring.

Audio Alchemy DTI (Digital Transmission Interface) jitter reduciton unit. Like new, $125 including UPS shipping. Gary Galo, (315) 265-4268 after 6:00 P.M..

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CD player with a silver face. B. Brown, RR 1, Box 50, Valley Springs, SD 57068, (605) 582-6762.

I need four to six AC-10 woofers or similar drivers. Please call with quantity you have and price. Dan M., (206) 874-8411 home, (206) 803-6410 message.

Cartridge for ADC Accutrac +6, Model #3500 RVC. Aaron Shipow, 3408 Ramstad Dr., San Jose, CA 95127, (408) 272-6836.

(Cheap) one Philips 544 speaker, Linn Ittok, Carver TFM 35 or 45. Tom, (415) 851-2779.

Dynaudio Esotar tweeters, Esotec 15W75 mids. (817) 244-3913.

Leak TL12, Marantz 2. Mike, (612) 475-2993.

Information on replacement voice coils for JVC DynaFlat ribbon tweeters. Dan, (801) 224-8080 (work) or (801) 225-8577 (home), 1768 N. 980 W., Orem, UT 84057.

Desperately seeking out of print CD of the England-based Shakatak. Title is "Down on the Street." Also any releases not available in the US welcome. Craig Moore, HC 76, Box 296, Garrison, KY 41141, (606) 757-3477.

Voice Coil back issues. Phil, (602) 579-1446.

Four Focal 8V416J, two D2905 tweeters, two AC-10 rubber surround woofers, one 1252 D.V.C. woofer. Jim, (708) 425-6719.

CLUBS

ALL PERSONS INTERESTED IN STARTING AN AUDIO/SPEAKER BUILDER CLUB IN THE SOUTHWEST MISSOURI-NORTHWEST ARKANSAS AREA, please send your name, address, phone number, and something about yourself to: Greg McKinney, 900 S. Roanoke, Apt. 2, Springfield, MO 65806.

OUTPUT TRANSFORMERS: THE LAST FRONTIER? Hobbyist seeks people interested in OPT design and building. I would like to network a critical mass of people together on this topic for strictly non-commercial interest. Possibly write a *GA* article/create a club newsletter? M.A. Martin, (416) 968-6101, monty@camtwh.eric.on.ca.

THOSE INTERESTED IN AUDIO and speaker building in the Knoxville-East Tennessee area please contact Bob Wright, 7344 Toxaway Dr., Knoxville, TN 37909-2452, (615) 691-1668, after 6 p.m.

WANTED: "CRADLE-TO-GRAVE" AUDIO ENTHUSIASTS. Texas doesn't have an audio club; YET! If you're interested in audio reproduction on stage and in your home, let's get together. (915) 676-7360.

LONDON LIVE DIY HI-FI CIRCLE meets quarterly in London, England. Our agenda is a broad one, including any aspect of audio design and construction. Subscription newsletter. We welcome all, from novice to expert, in free association. Contact Brian Stenning, UK, 081-748-7489.

THE HI-FI CLUB of Cape Town in South Africa sends a monthly newsletter to its members and world-wide subscribers. To receive an evaluation copy of our current newsletter, write to PO Box 18262, Wynberg 7824, South Africa. We'll be very pleased to hear from you.

WASATCH AUDIO, located in Salt Lake City. Our club is interested in construction, modifications, design, and listening to music. We are looking for members and ideas for our new club. Edward Aho, (801) 364-4204.

HI-FI COLLECTOR/HOBBYIST seeks audio penpals to correspond via reel-to-reel tape. If you are into restorations, discarding old recorders, or parting out derelict gear, or have arcane technical secrets you'd like to discuss, or just want an excuse to obsess over mylar, make it so. I'll return all tapes via parcel post. James Addison, 171 Hartford Rd., A-7, New Britain, CT 06053.

SOUTHEASTERN MICHIGAN WOOFER AND TWEETER MARCHING SOCIETY (SMWTMS). Detroit area audio construction club. Meetings every two months featuring serious lectures, design analyses, digital audio, AB listening tests, equipment clinics, and audio fun. Club publication, *LC, The SMWTMS Network*, journals the club's activities and members' thoughts on audio. For information, send name and address: e-mail aa259@detroit.freenet.org; (810) 544-8453 (machine); SMWTMS, PO Box 721464, Berkley, MI 48072-0464.

WANTED: SPEAKER AND AUDIO AMATEURS IN THE BRADENTON/SARASOTA/ST. PETERSBURG/TAMPA, FL AREAS. Would like to form a club and develop a lab for testing speakers/amps/preamps and passive and active crossovers or just to discuss speaker projects and ideas. Angel Rivera, Bradenton, FL 34206, (813) 792-3870.

ARIZONA AUDIOPHILE SOCIETY located in metropolitan Phoenix is a growing and active club in the pursuit and reproduction of recorded music. New members are welcome. Meetings are last Tuesday of each month. Receive monthly newsletter and biannual journal. Club discounts with local high-end audio dealers. Send inquiry to Arizona Audiophile Society, PO Box 13058, Scottsdale, AZ 85267, or call Bob Williams, (602) 944-5929.

THE ATLANTA AUDIO SOCIETY, Inc. – Open to persons that share a common interest in the art and science of audio and music. Activities include construction projects, equipment and recording evaluations, home theater, video, film, live music and recording sessions. Monthly meetings and seminars often include prominent speakers from the industry. Newsletters and bulletins regularly published. John Morrison, Pres., (404) 491-1553, or Chuck Bruce, (404) 876-5659, 4266 Roswell Rd. N.E., K-4, Atlanta, GA 30342-3738.

AUDIO SOCIETY OF MINNESOTA. Meeting the hobbyist needs of audiophiles in the Land of 10,000 Lakes for 16 years! Monthly meetings (Sept.-May), tours, audiophile concerts, special guests, technical assistance, etc. For information and a copy of our latest monthly Bulletin, write ASM, PO Box 32293, Fridley, MN 55432, or call our 24-hour "Audio Hotline," (612) 825-6806.

THE INLAND EMPIRE AUDIO SOCIETY–soon to become **THE SOUTHERN CALIFORNIA AUDIO SOCIETY (SCAS)**–is now inviting audiophiles from all areas of Southern California and abroad to join our serious pursuit for that elusive sonic truth through our meetings and the **IEAS** official speaker, *The Reference* newsletter. For information write or call Frank Manrique, President, 1219 Fulbright Ave., Redlands, CA 92373, (714) 793-9209.

THE AUDIO SYNDROME, Nassau and Suffolk county's oldest group of audiophiles is looking for new members. If you are interested, call Roy Harris, (516) 489-9576.

THE BOSTON AUDIO SOCIETY, the nation's oldest (founded 1972), seeks new members. Dues include the monthly meeting notice and our newsletter, the *BAS Speaker* (6/year). Recent issues cover Carver, a/d/s; the founder of Tech Hi-Fi, Photo CD. Plus visits from famous speaker designers, listening tests, measurement clinics, research investigations, and more. Back volumes available. Membership includes engineers, journalists, consultants, and music-loving audiophiles like yourself. For information write to PO Box 211, Boston, MA 02126-0002.

THE CATSKILL AND ADIRONDACK AUDIO SOCIETY invites you to our informal meeting. Join our friendly group of audio enthusiasts as we discuss life, the universe and everything! Toobers, Tranzzeestors, vinyl canyons, or digital dots. No matter what your level of interest, experience, or preferences, you are welcome. Contact CAAS at (518) 756-9894 (leave message), or write CAAS, PO Box 144, Hannacroix, NY 12087.

THE COLORADO AUDIO SOCIETY (CAS) is a group of audio enthusiasts dedicated to the pursuit of music and audiophile arts in the Rocky Mountain region. We offer a comprehensive annual journal, five newsletters, plus participation in meetings and lectures. For more information, send SASE to CAS, 1941 S. Grant St., Denver, CO 80210, (303) 733-1613.

CONNECTICUT AUDIO SOCIETY is an active and growing club with activities covering many facets of audio—including construction, subjective testing, and tours of local manufacturers. New members are always welcome. For a copy of our current newsletter and an invitation to our next meeting, write to: Richard Thompson, 129 Newgate Rd., E. Granby, CT 06026, (203) 653-7873.

WANTED: SPEAKER/TUBE GEAR BUILDERS AND AUDIO AMATEURS IN SANTIAGO, CHILE. Would like to form a club or just a friendly group of audio enthusiasts to discuss speakers/amps/preamps/crossovers and everything. Anyone interested, contact Christian Bargsted, Los Gomeros 1542, Vitacura, Santiago or call (562) 538-0638, office hours.

MONTREAL SPEAKER BUILDER CLUB. Meets when it can, BYOB, discussions range from speaker design and testing to equipment modification. All welcome. Contact Andrew McCrae, 4701 Jeanne Mance, Montreal, PQ, H2V 4J5 Canada, (514) 281-7954.

ELECTROSTATIC LOUDSPEAKER USERS GROUP is now a world-wide network for those interested in sharing valuable theory, design, construction, and parts source information. If you are interested in building, or have built, your own SOTA ESL we invite you to join our loose-knit organization. For information, send SASE to: Barry Waldron, 1847 Country Club Dr., Placerville, CA 95667.

ESL BUILDERS GROUP is a new address for people who have built or want to build electrostatic loudspeakers and associated (tube) drivers, or are just interested. An answer is ensured, if you include some kind of compensation for postage and handling. Write to: Gunter Roehricht, Bühlerstr. 21, 7030 Böblingen, Germany.

WEST VALLEY AUDIO SOCIETY. We are starting a group interested in all aspects of high performance audio. West San Fernando Valley, CA. Contact Barry, (818) 225-1341.

NEW JERSEY AUDIO SOCIETY meets monthly. Emphasis is on construction and modification of electronics and speakers. Dues include monthly newsletter with high-end news, construction articles, analysis of commercial circuits, etc. Meetings are devoted to listening to records and CDs, comparing and A-B-ing equipment. New members welcome. Contact Frank J. Alles, (908) 424-0463, 209 Second St., Middlesex, NJ 08846; or Bob Young, (908) 381-6269; or Bob Clark, (908) 647-0194.

PACIFICNORTHWEST AUDIO SOCIETY (PAS) consists of 40 audio enthusiasts meeting monthly, second Wednesdays, 7:30 to 9:30 p.m., 4545 Island Crest Way, Mercer Island, WA. Write Box 435, Mercer Island, WA 98040 or call Ed Yang, (206) 232-6466, or Gill Loring, (206) 937-4705.

PIEDMONT AUDIO SOCIETY in the Raleigh/Durham and Chapel Hill area is meeting monthly to listen to music, demonstrate owner-built and modified equipment, and exchange views and ideas on electronics and speaker construction. Tube and solid-state electronics are of interest and all levels of experience are welcome. Kevin Carter, 1004 Olive Chapel Rd., Apex, NC 27502, (919) 387-0911.

THE PRAIRIE STATE AUDIO CONSTRUCTION SOCIETY (PSACS) meets every other month. Meetings feature audio construction, design, and analyses, blind listening tests, equipment clinics, auto sound, lectures from manufacturers and reviewers. PSACS, PO Box 482, Cary, IL, 60013, or call Tom, (708) 248-3377 days, (708) 516-0170 eves.

THE WESTERN NEW YORK AUDIO SOCIETY is an active, long-established club located in the Buffalo area. We issue a newsletter and hold meetings the first Tuesday of every month. Our meetings attract many prominent manufacturers of audio-related equipment. We are involved in all facets of audio from building/modifying to exposure to the newest high-end gear, and the chance to hear more types of music. For information, write to WNY Audio Society, PO Box 312, N. Tonawanda, NY 14120.

TUBE AUDIO ENTHUSIASTS. Northern California club meets every other month. For next meeting announcement send #10 SASE to Tim Eding, PO Box 611662, San Jose, CA 95161.

IF YOU ARE an "Organ Music Lover" and like to test your audio system, **SFORZANDO** has room for a few more members. We have about 3,000 "live," on-the-spot, cassette tapes that are not available in the stores. We are happy to lend them to you via the mail. Just ask EA Rawlings, 5411 Bocage St., Montreal, Canada H4J 1A2.

MEMPHIS AREA AUDIO SOCIETY being formed. Serious audiophiles contact J.J. McBride, 8182 Wind Valley Cove, Memphis, TN 38125, (901) 756-6831.

VINTAGE AUDIO LISTENERS AND VALVE ENTHUSIASTS (VALVE) meets the first Sunday of every month to swap vintage audio gear, audition rare and collectible equipment, and evaluate modifications and scratchbuilt projects. Dues provide a monthly newsletter with current reviews of vintage components and modification information; vintage service data; and access to an active network of serious collectors. For information, call (206) 697-1936 or write to 1127 NW Brite Star Ln., Poulsbo, WA 98370.

WASHINGTON AREA AUDIO SOCIETY meetings are held every two weeks, on Fridays, from 19:00 hours to 21:30 hours at the Charles Barrett Elementary School in Alexandria, VA. Prospective members are welcome but must register in advance in order to be admitted to the meetings. No exceptions please. Call Horace Vignale, (703) 578-4929.

Ad Index

Reader Service #19 →

AUDAX KIT PLANS

Speaker Kit Plans Are Now Available From Parts Express!

"Kit Plans, Build Your Own Loudspeakers" is a 25 page booklet containing plans for 7 speaker kits. The Signature Series has 4 different kits utilizing Prestige and Reference Series components and was designed exclusively by Vance Dickason, author of "The Loudspeaker Design Cookbook". The enclosures, crossovers, and components were carefully selected by Mr. Dickason and computer optimized to ensure the highest level of performance. The Classic Series was designed in house by Polydax and includes 3 easy-to-build kits that offer great sound quality at an affordable price. All of the components and accessories you need to build any of these projects are available and in stock at Parts Express.

To receive the booklet "Kit Plans, Build Your Own Loudspeakers", and a copy of the 188 page Parts Express 1994 Catalog, just send $1.00 to cover postage and handling ($3.00 to Canada).

SIGNATURE
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SERIES

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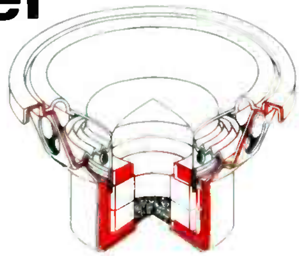
Please send check or money order to receive the kit plans. Be sure to include your name and complete address with zip code. Mail to:

PARTS EXPRESS INT'L.
DEPARTMENT A
340 E. FIRST STREET
DAYTON, OH 45402-1257

LOCAL: 513/222-0173, FAX: 513/222-4644

MW 114-S

Neodymium Magnet DPC Cone 4" Woofer



Specification

Overall Dimensions	Ø118mm (4.64") x 58mm(2.29")
Mounting Baffle Hole Diameter	Ø95mm (3.75")
Magnet System	Pot Type, Vented, Neodymium Magnet
Nominal Power Handling (Din)	150W
Transient Power - 10ms	800W
Voice Coil Diameter	54mm (2.125")
Voice Coil Type/Former	Hexatech Aluminium
Frequency Response	55-7000 Hz
FS - Resonant Frequency	65 Hz
Sensitivity 1W/1m	87 dB
Z - Nominal Impedance	8 ohms
RE - DC Resistance	5.6 ohms
LBM - Voice Coil Inductance @ 1kHz	0.47 mH
Magnetic Gap Width	1.25mm (0.050")
HE - Magnetic Gap Height	6mm (0.236")
Voice Coil Height	12mm (0.472")
X - Max. Linear Excursion	3mm
B - Flux Density	0.88T
BL Product (BXL)	6.75
Qms - Mechanical Q Factor	2.32
Qes - Electrical Q Factor	0.36
Q/T - Total Q Factor	0.31
Vas - Equivalent Cas Air Load	3.18 litres (0.113 cu. ft.)
MMS - Moving Mass	7.00gm
CMS	807µm/n
SD - Effective Cone/Dome Area	53cm² (20.86 sq. in.)
Cone/Dome Material	DPC (Damped Polymer Composite)
Nett Weight	0.500 kg

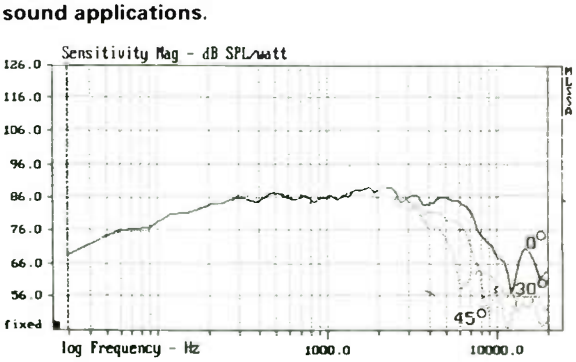
Specifications given are as after at least 45 minutes of high power, low frequency running, or 24 hours normal power operation.

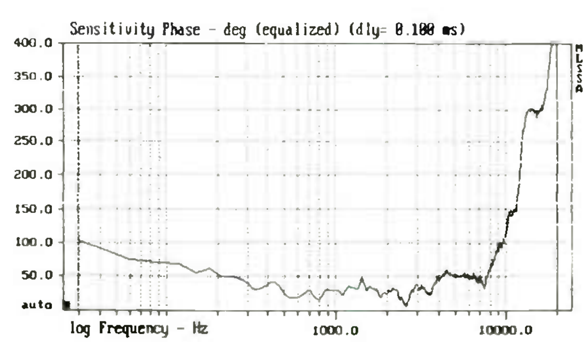
The 114-S is the first of Morel's new generation of woofers, featuring a powerful Neodymium magnet system which provides increased sensitivity, lower Qt and reduced distortion. For a 4" driver it is unique in having a large 54mm (2.125") diameter Hexatech aluminium voice coil.

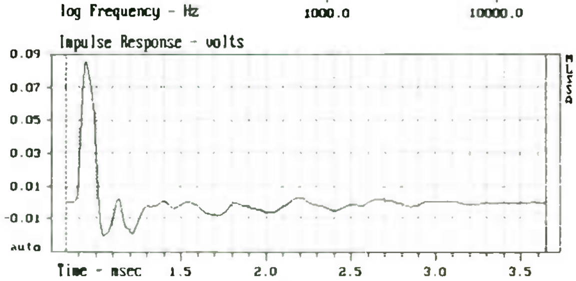
Benefits of this large voice coil diameter include a very high power handling capacity and lack of sound level compression. In addition, it allows the use of a very shallow cone profile. Coupled with the use of Damped Polymer Composite cone material and a rubber surround, this provides excellent dispersion (off-axis response), resistance to cone break-up (even at high sound pressure levels) and lack of colouration.

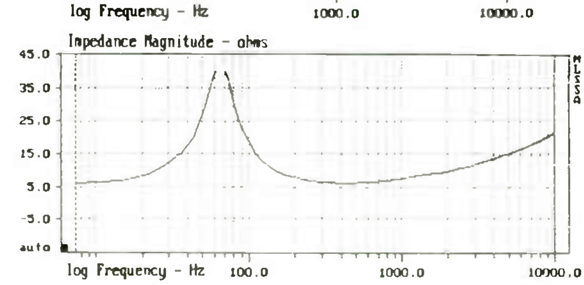
Frequency and phase response are very flat, while the roll-off is very smooth. The MW 114-S may be used either as a bass-mid range in 2-way systems, or as a mid-range in multi-way systems.

The vented magnet system is encased within a steel chassis, which improves efficiency and shields the magnet, virtually eliminating stray magnetic fields. The MW 114-S is ideal not only for high quality hi-fi, but also TV, video and surround-sound applications.









Morel operate a policy of continuous product design improvement, consequently specifications are subject to alteration without prior notice.

morel

high fidelity range

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