FINDING A GOOD, CHEAP SUBWOOFER

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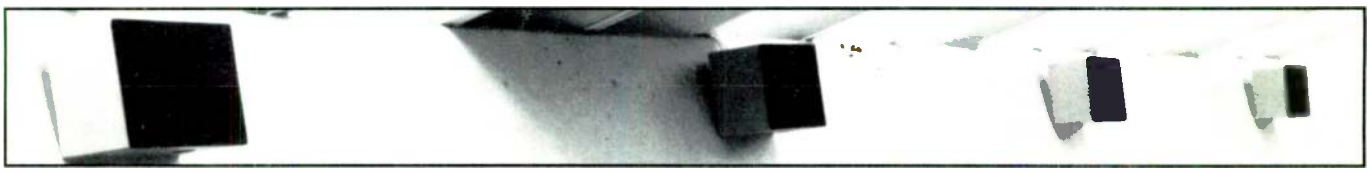
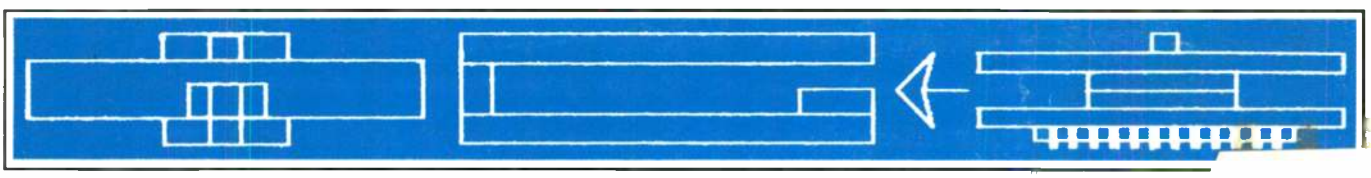
Speaker Builder
THE LOUDSPEAKER JOURNAL

PUBLIC SOUND



MULTI COILS



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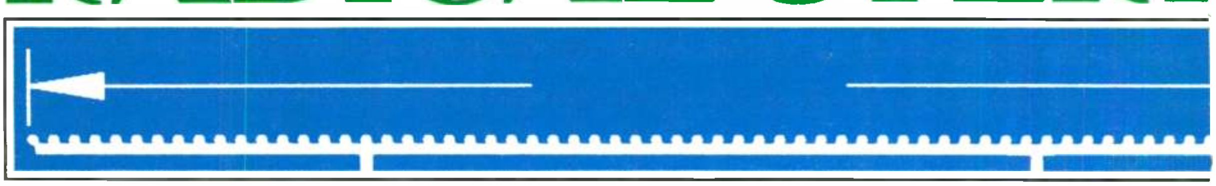

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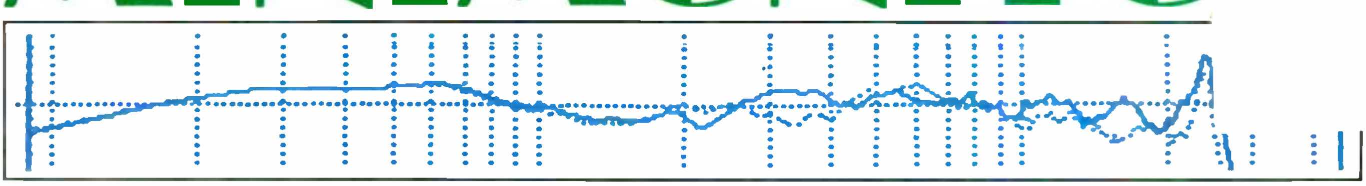

P-SPICING A GREAT CROSSOVER

Good News

■ LOUDSPEAKER UNIVERSITY

On May 13–15, Menlo Scientific, Ltd., will sponsor Loudspeaker University at the Lowell Campus of the University of Massachusetts Center for Recording Arts, Technology & Industry. Hosted by Menlo president Michael Klasco, the three-day intensive workshop will focus on reducing distortion and improving the power handling of loudspeakers. A distinguished roster of speakers will address assorted topics. For further information call Mike Klasco at (510) 528-1277, or (603) 598-7268.

■ SITTING DUCK SOFTWARE

Listening Room, a new program for Macintosh computers, allows users to position listener and speakers interactively and displays the magnitudes of standing waves, plus the effects of 124 early reflections on the direct response. Local optimization of listener and/or speaker positions and other new features. Sitting Duck Software, PO Box 130, Veneta, OR 97487, (503) 935-3982.

Reader Service #102

PARTS EXPRESS

Parts Express announces the release of its 1994 catalog of electronic components, including speakers, audio accessories, video products, and semiconductors. The 188-page catalog is free by calling (800) 338-0531; or write Parts Express, 340 E. First St., Dayton, OH 45402-1257.

Reader Service #124

■ TRUE IMAGE AUDIO

MacSpeakerz version 3.0, newly released by True Image Audio, can now analyze six different box types and their Isobarik variations. Modified box calculators add locks to each of the box dimension fields, and a Bandpass Box Calculator has been added. For existing closed-box systems, MacSpeakerz 3 shows how you can extend the bass response by adding a series capacitor to convert the system to a third-order closed box. True Image Audio, 349 W. Felicita Ave., Suite 122, Escondido, CA 92025, (800) 621-4411, FAX (619) 480-8961.
Outside the US phone/FAX (619) 480-8961.

Reader Service #107

POLYDAX

Polydax presents *Build Your Own Loudspeakers*, a 28-page catalog of loudspeaker kit plans fully optimized, auditioned, and tested by Vance Dickason, author of *The Loudspeaker Design Cookbook* and editor of *Voice Coil* newsletter. Each kit includes a crossover diagram, enclosure plans, performance curves, and construction tips. Booklet includes Dickason's four-system Signature Series using Audax products, as well as three Polydax systems, the Classic Series. Polydax Speaker Corporation, 10 Upton Dr., Wilmington, MA 01887, (508) 658-0700, FAX (508) 658-0703.

Reader Service #101

SYON

Syon Tru-Set® General Purpose Adhesive, a new two-part epoxy, features a Shore Hardness of 79D, and has a flexural strength of 8,000psi. It sets to the touch in 20min; full cure is achieved in four hours. Syon Corporation, 280 Eliot St., Ashland, MA 01721, (508) 881-8852, FAX (508) 881-4703.

Reader Service #112

■ DIABLO ACOUSTICS

Steve and Larry Patzkowski, formerly majority owners of Hales Audio Loudspeakers, have formed Diablo Acoustics. The new company, which will produce ultra-high-quality loudspeakers for the high-end audio market, introduces its Model 2.
Diablo Acoustics, 1944 Windward Point, Discovery Bay, CA 94514, (510) 516-0864, FAX (510) 516-0184.

Reader Service #121

■ GERMAN PHYSIKS

German Physiks announces availability of a new 360° dispersion "Dipolar" driver having a frequency range of 150Hz–16kHz (±2.5dB), being –6dB at 70Hz. Power handling is 120W @ 4Ω; sensitivity is 87dB/W/m. OEM enquiries to FAX (011) 49-69-4940963. German Physiks, Postbox 1026, D-63506 Hainburg, Germany.

Reader Service #106

■ AUDIOCONTROL

With a new factory-installed SPL-170 upgrade, the AudioControl Industrial SA3050 series II realtime audio analyzer can now measure up to 170dB SPL with 0.1dB resolution. Features a high-sound pressure-calibrated microphone. Special introductory "preorder" price through May 31: $650. AudioControl, 22410 70th Ave. West, Mountlake Terrace, WA 98043, (206) 775-8461, FAX (206) 778-3166.

Reader Service #110

← ***Reader Service #36***

Good News

■ MAHOGANY SOUND

Using John Cockroft's work for optimizing transmission-line designs (*SB* 4/88), Mahogany owner Larry Sharp has synthesized all available information into a systematic plan for determining optimum stuffing density. The result is a 22-page booklet, "Quick & Easy Transmission Line Speaker Design." Accompanying diskette contains three copies of the LOTUS 123 worksheet under different names; booklet includes duplicates of TL Worksheet and woofer test data sheet. Mahogany Sound, 2610 Schillingers Rd., #488, Mobile, AL 36695, (205) 633-2054.

Reader Service #104

➲ THE PARTS CONNECTION

In conjunction with the publication of its new, expanded catalog ($5), Sonic Frontiers, Inc., announces the formation of The Parts Connection, a one-stop audiophile components shop. The 100-page 1994 catalog lists many new MIT cap listings, plus resistors, speaker tweakers, cable and connectors, transformers and expanded part listings. The Parts Connection, 2790 Brighton Rd., Oakville, ON, Canada L6H 5T4, (905) 829-5858, FAX (905) 829-5388.

Reader Service #113

⊂ ACOUSTIC INNOVATIONS

AI's new Sound Boards and SoundScreens were conceived as an elegant solution to problems of shrillness and lack of musical definition in home video and listening rooms with tile, glass, and high ceilings. The oak-framed three-panel SoundScreen doubles as an absorber and a room divider, and both are offered in two fabric choices. Acoustic Innovations, 6780 E. Rogers Circle, Boca Raton, FL 33487, (407) 995-0090, FAX (407) 995-0290.

Reader Service #114

■ LINEARX

FilterCAD, an MS Windows®-based engineering software program for the design and analysis of active filter circuitry, is ready for shipping from LinearX. The new computer-aided program contains the necessary filter synthesis equations for 125 separate circuit topologies, including many numerical solutions—plus target-generation system, built-in standard filter functions, and complete transformations for low-pass, high-pass, all-pass, bandpass, and band reject. LinearX Systems, Inc., 7556 SW Bridgeport Rd., Portland, OR 97224, (503) 620-3044, FAX (503) 598-9258.

Reader Service #108

■ FM ATLAS

Bruce Elving has released the 15th edition of *FM Atlas*, the largest directory for FM radio ever published. The 208-page book includes 102 pages of maps, plus directories with stations listed geographically and by station frequency. A separate listing shows more than 2,000 FM translators. Bruce F. Elving, Publisher, FM Atlas Publishing, PO Box 24, Adolph, MN 55701-0024, (218) 879-7676.

Reader Service #103

■ R.F. ENGINEERING

With two current-sensing outlets (450W each) and eight switched outlets (1kW each), R.F. Engineering's new AC-8 Power Controller automatically senses when a piece of equipment is turned on, then powers up the required devices. The AC-8 also provides power conditioning and surge protection. Richard Reisbick, R.F. Engineering, Inc., 8884 Wagner St., Westminster, CO 80030, (303) 430-8281, FAX (303) 430-4023.

Reader Service #118

■ R.F. SYSTEMS

The RC-8-PS, an intelligent power controller for the high-end home-theater system, features two current-sensing outlets (450W each) and eight switched outlets (1.8kW each), an RS-232 port, an RC-8-VX interface, and integrated power control for remote-controlled equipment. Two 20A circuits, with power conditioning and surge protection, permit utilization in high-current installations. Richard Reisbick, R.F. Systems, Inc., 8884 Wagner St., Westminster, CO 80030, (303) 430-8281, FAX (303) 430-4023.

Reader Service #122

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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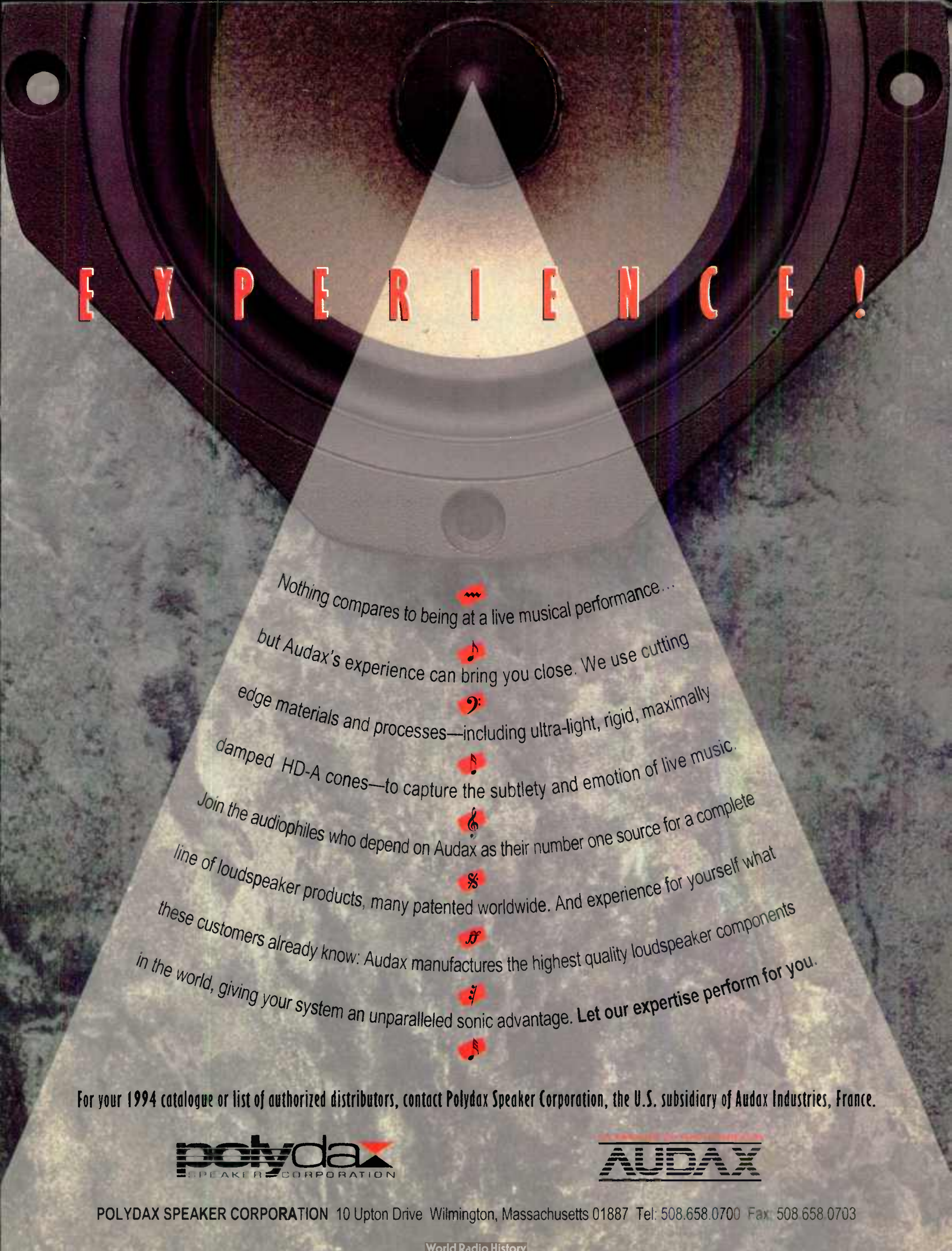
■ INTERACTIVE IMAGE TECHNOLOGIES

Electronics Workbench Version 3, now available on MS-DOS, Microsoft Windows, and Macintosh platforms, simulates analog and digital circuits as well as test equipment. Educators can introduce faults or malfunctions for students to troubleshoot. Up to 16MB of RAM, plus JFETs, MOSFETs, and controlled sources and switches to the parts bin. Interactive Image Technologies, Ltd., 700 King St. West, Suite 815, Toronto, ON, Canada M5V 2Y6, (416) 361-0333, FAX (416) 368-5799.

Reader Service #115

Continued on page 6

Reader Service #21 ➞


EXPERIENCE!
Nothing compares to being at a live musical performance...
but Audax's experience can bring you close. We use cutting
edge materials and processes—including ultra-light, rigid, maximally
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Join the audiophiles who depend on Audax as their number one source for a complete
line of loudspeaker products, many patented worldwide. And experience for yourself what
these customers already know: Audax manufactures the highest quality loudspeaker components
in the world, giving your system an unparalleled sonic advantage. Let our expertise perform for you.
For your 1994 catalogue or list of authorized distributors, contact Polydax Speaker Corporation, the U.S. subsidiary of Audax Industries, France.
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AUDAX
POLYDAX SPEAKER CORPORATION 10 Upton Drive Wilmington, Massachusetts 01887 Tel: 508.658.0700 Fax: 508.658.0703

Good News

Continued from page 4

■ LINEARX
LinearX introduces the pcRTA™, a pc-based realtime analyzer. Features include true RMS detection, precision four-pole MFL filters, a pink-and-white-noise generator, peak hold, curve inverting and storage, ASCII data export/import, ANSI A, B, C weighting (or flat), four-mike multiplexing, spacial averaging, and full printing facilities. The pcRTA comes with the M51 measurement microphone. LinearX Systems, Inc., 7556 SW Bridgeport Rd., Portland, OR 97224, (503) 620-3044, FAX (503) 598-9258.
Reader Service #105

■ POLYDAX
The 1994 Audax catalog, now available from Polydax, describes 30 new products, including such innovations as the HD-A (High Definition Aerogel) and HD-I (High Definition Inertial) cone materials. Other new products in the 267-page catalog include 10mm neodymium diaphragm tweeters, audio/video shielded woofers, dual-voice-coil drivers, and an expanded automotive section. Polydax Speaker Corporation, 10 Upton Dr., Wilmington, MA 01887, (508) 658-0700, FAX (508) 658-0700.
Reader Service #111

■ AVEL TRANSFORMERS
The resin-filled thermoplastic case on Avel's encapsulated standard toroidal transformer protects it from harsh environmental conditions and accidental damage, permitting toroids to be operated continuously at power levels up to 25% higher than nominal. Avel's ready-made toroidal transformer, with polyester tape finish, is the least expensive option for purchasing off the shelf to match J.I.T. schedules. The new "5000" range with single primaries is UL 506 recognized. Avel Transformers, Inc., 47 South End Plaza, New Milford, CT 06776, (203) 355-4711, FAX (203) 354-8597.
Reader Service #109

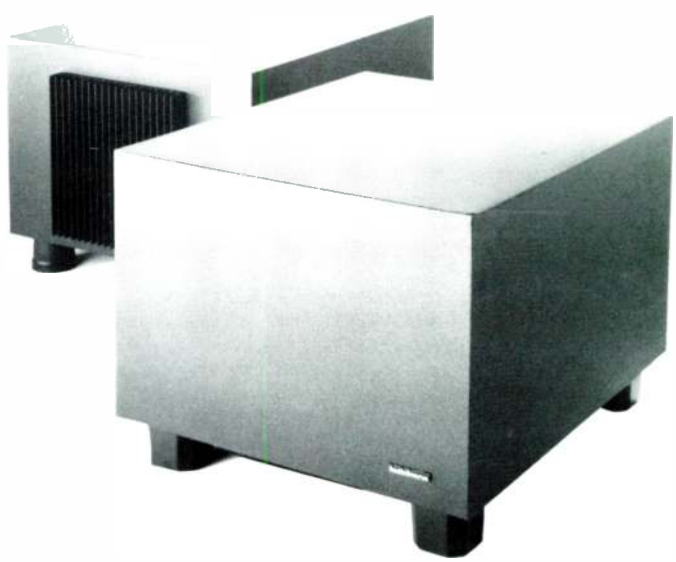
■ INTERNATIONAL JENSEN
IJI is introducing a new line of direct factory replacement speakers with a unique weather-resistant blue kapok cone, plus custom-fit magnet covers for speaker and vehicle protection during installation. The compact J226CX 6½″ and J225CX 5¼″ coax drivers and J165FR 6½″ and J124FR 4″ full-range units all feature a low-profile design, and their metric terminals are compatible with most manufacturers' wiring adapters. The speakers' metal grilles feature high-temperature frames. International Jensen, Inc., 25 Tri- State Intl. Office Ctr., Lincolnshire, IL 60069, (708) 317- 3700.
Reader Service #117

WATERWORKS ACOUSTICS
The SOUNDROCK™, Waterworks' new all-weather speaker shaped and textured to resemble a granite boulder, is designed to harmonize with any landscape plan. The fiberglass enclosure is sealed against the elements and comes with 50′ of direct-bury wire. Frequency response 40Hz–20kHz (±3dB); shipping weight 25 lb. Available at specialty audio retailers and installers in the US and Japan, or by calling (510) 522-0374.
Reader Service #123

■ AWS GROUP
ProSystems announces the release of a cost-effective alternative to piezoelectric devices—the 30HM2, a small professional "bullet" tweeter. Features include die-cast zinc housing, duraluminum diaphragm, 1¼″ copper-clad aluminum voice coil; typical efficiency of 110dB 1W/1m. The 30HM2 easily handles 30W (AES) of power. ProSystems, The AWS Group, Inc., 65 36th St., Wheeling, WV 26003-1946, (304) 233-2223, FAX (304) 233-2258.
Reader Service #120

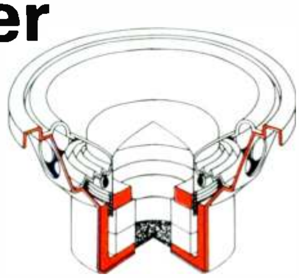
■ AUDIOPHILE AUDITION
Audiophile Audition, a program for sound buffs/music lovers aired by some 135 stations across the country, observes its ninth anniversay in April. The weekly program, which will henceforth be 100% classical, also features interviews and audio news, plus two all-binaural broadcasts yearly. For a station carriage list, 13-program playlist, and annual "Best of the Year" list send legal-size SASE plus four 29-cent stamps to Box 1621n, Ross, CA 94957.
Reader Service #119

AUDIOSOURCE
AudioSource's SW-series, a trio of active subwoofers, combine built-in amplifiers, 12″ drivers, and flexible two-way electronic and passive crossovers, and deliver low-frequency response specified at 20–200Hz. The SW Four and Five's on-board amplifiers produce 150W RMS throughout the subwoofers' operating range; the SW Six includes two additional built-in channels for integrated satellite power, plus infrared wireless remote control. AudioSource, 1327 N. Carolan Ave., Burlingame, CA 94010, (415) 348-8114, FAX (415) 348-8083.
Reader Service #116



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

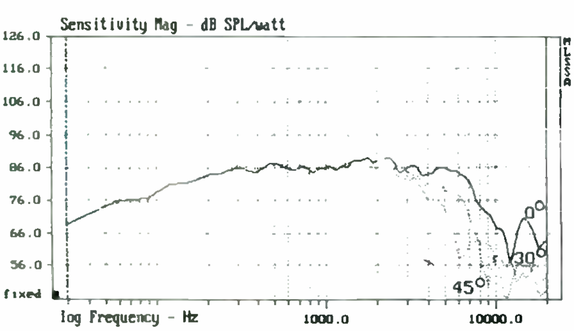
Specifications given are as after at least 45 minutes of high power, low frequency running, or 24 hours normal power operation.

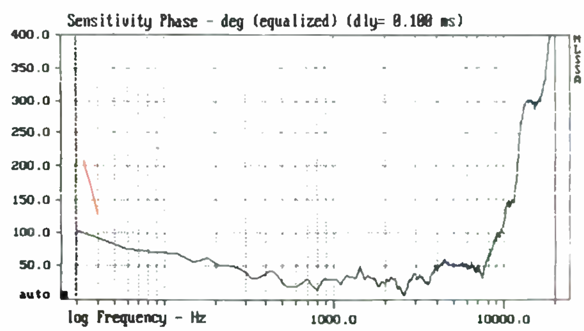
The 114-S is the first of Morel's new generation of woofers, featuring a powerful Neodymium magnet system which provides increased sensitivity, lower Qt and reduced distortion. For a 4" driver it is unique in having a large 54mm (2.125") diameter Hexatech aluminium voice coil.

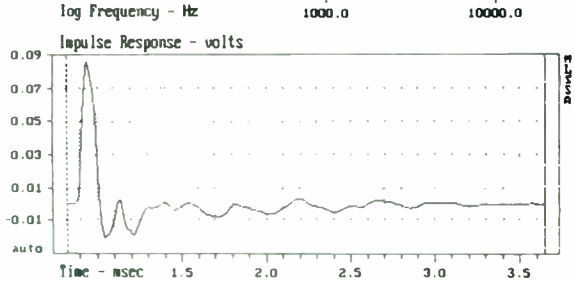
Benefits of this large voice coil diameter include a very high power handling capacity and lack of sound level compression. In addition, it allows the use of a very shallow cone profile. Coupled with the use of Damped Polymer Composite cone material and a rubber surround, this provides excellent dispersion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

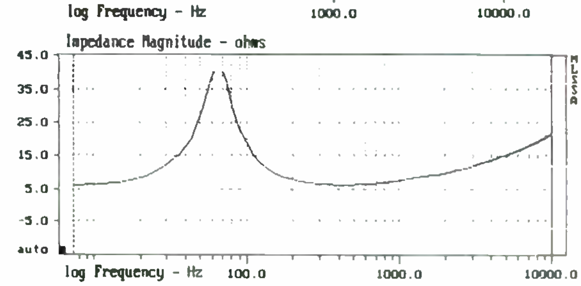
Frequency and phase response are very flat, while the roll-off is very smooth. The MW 114-S may be used either as a bass-mid range in 2-way systems, or as a mid-range in multi-way systems.

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.

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high fidelity range

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Industrial Estate,
Ipswich, IP3 9RT
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morel acoustics USA
414 Harvard Street
Brookline, MA 02146
Tel: (617) 277-6663
FAX: (617) 277-2415

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Speaker Builder

The Staff

Editor and Publisher
Edward T. Dell, Jr.

Contributing Editors
Joseph D'Appolito **Robert Bullock**
Richard Campbell **John Cockroft**
David Davenport **Vance Dickason**
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(603) 924-9464 **FAX (603) 924-9467**

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

—JOHN STUART MILL

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About This Issue

Speaker builders, it seems, are taking up the challenge of the excruciatingly poor sound in most public areas. **Hilary Paprocki** took his local school's cafeteria in hand and the account of his adventures offers instructive reading and a nice blueprint for other adventurous readers. The narrative begins on page 12. Finding just the right value of a crossover inductor is always a chore unless you build one of **Kim Girardin**'s multitap constructions, which he describes and defines beginning on page 16.

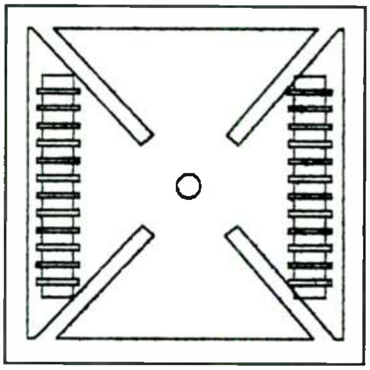
Those elusive bottom octaves have become a must for any serious music lover, but finding an affordable, effective performer is usually daunting. **John Sehring** was especially anxious to find an answer with accurate phase performance. His informative chronicle of search and discovery starts on page 20.

Alan Blumlein's 1932 patent for our stereo recording formula is now more than old—indeed, given the speed of audio's evolution, we could regard it as archaic. The search path for a new formula is strewn with tombstones marking some spectacular failures, and one rather esoteric and reticent survivor, Ambisonics. Starting on page 12, **Philip Witham**, an extraordinarily brave soul, sets out a radical, innovative, and wholly unusual proposal for a new way of recording and reproducing more of the signal. He also issues a call for volunteers to join him in a pilgrimage of discovery. Now it's your move.

Ralph Gonzalez suffers from the well-known speaker-building virus, "Neverletwellenoughalonus." Readers will be the beneficiaries of this new incarnation of his Delac S10 minimonitor (SB 3/91). Check out the new incarnation starting on p. 34.

Curiosity about a circuit's underpinnings plagues many. In welcoming our newest contributing editor, **Dick Campbell**, we note that he is one of the afflicted. His questions for many recording engineers not only grace our editorial page, but his curiosity about the inner workings of the Borbely/Gaertner crossover in 1/94 also prompted a PSpice® quest. His examination of BUF 124 with this popular analysis tool, beginning on page 39, is also a fine tutorial on using PSpice.

Other nourishing fare this time includes Contributing Editor **Vance Dickason**'s review of a Signet product, **Bob Wayland**'s tips on cutting plies, **Ed Schilling**'s aid on building spare power, and Contributing Editor **Bob Bullock**'s words on crossover order mysteries.



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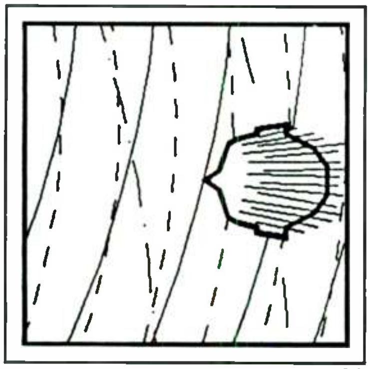

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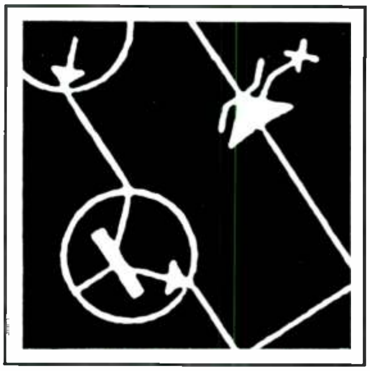

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Speaker Builder

THE LOUDSPEAKER JOURNAL

VOLUME 15 NUMBER 3 APRIL 1994

DEPARTMENTS

BLACK HOLE

High efficiency dampening products from ORCA

A better speaker damping material...

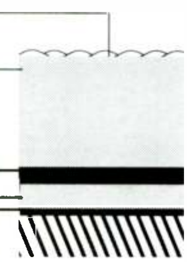
If you've been building speakers for some time, you know how much guesswork goes with speaker damping and stuffing. The choices seem endless: fiberglass, wool, Dacron, flat foam, convoluted foam, felt, tar, plus various "magic" compounds that you're invited to brush or pour into your new cabinets. Everyone has their own recipe, and who knows if it's a recipe for disaster? Or what effects the vapors emitted by these chemicals might have on the glues that bond your woofer surround to its cone and chassis? In this era of costly, space-age drivers and computer-assisted design, we think such risks are totally unacceptable. So we went to work to find the ideal solution.

The problems are fairly well-known: a driver transforms electrical energy into mechanical energy. This mechanical energy is transformed into acoustical energy which is radiated to the outside of the cabinet - the useful front wave - and to the inside - the sometimes-useful back wave. Unfortunately, it is also transmitted though the frame of the driver to the cabinet itself, which acts as a very large "cone" of very small excursion. This means that the spurious resonances and vibrations of the cabinet have to be controlled in a predictable and reproduceable way. That's how we came to BLACK HOLE 5 and the BLACK HOLE PAD.

First, THE PAD. It's a thin (1/16 inch) black flexible viscoelastic damping material (filled vinyl copolymer) with maximum performance between 50 and 100 degrees F (we hope that that covers the temperature range of your listening room) and excellent flame resistance - it meets UL94 V-O. Thanks to its outstanding damping characteristics, THE PAD will dramatically reduce the vibration energy stored in the walls to which it is applied.

Easy to cut and apply, THE PAD has a pressure-sensitive adhesive back: simply peel off the release paper and press hard onto a clean surface. You can use THE PAD on just about anything you suspect of vibrating: driver frames, thin panels like car doors, and, of course, the walls of your speaker cabinets. And it can be used to recess a driver without using a router: just laminate enough layers to match the thickness of the driver frame and apply to the front baffle. Finally, it is the ideal material for "constrained layer" wall construction, where two panels are laminated on each side of a damping material for optimum transmission loss. Because THE PAD has a fine grain leather finish, you can wrap an entire cabinet exterior and give it an attractive appearance at the same time!

For applications which require **maximum damping, isolation and absorption,** we've developed BLACK HOLE 5. One and 3/8" thick, BLACK HOLE 5 is a high-loss laminate that provides optimum acoustical damping performance. It consists of five layers:



Thin diamond-pattern embossing, densified with a polyurethane film surface. This unique surface layer dramatically improves the performance of the whole acoustical system, especially the lower mid-range and mid-bass frequencies where simple acoustical foam loses its effectiveness.

One-inch deep polyester urethane foam, structurally optimized for acoustical damping. Highly effective at "soaking" maximum sound energy with minimum thickness.

Barrier septum, 1/8 inch thick. Made of limp flexible vinyl copolymer loaded with non-lead inorganic fillers, it is a "dead wall" that isolates the vibrations in the walls of your cabinet from the vibrations created inside the enclosure.

Polyester urethane flexible open-cell foam, 1/4 inch thick. Thanks to special vibration-isolation characteristics, it decouples the vibrating structure (the wall) from the rest of the damping system, thus optimizing performance.

High-loss vibration damping material, same as The Pad. It is strongly bonded to the cabinet wall with pressure sensitive adhesive.

These layers are laminated using an adhesive-free mechanical and thermal process, thus optimizing performance and eliminating the risk of solvent fume damage. BLACK HOLE 5 can be used in any enclosure, as well as for acoustical panels to improve the characteristics of your listening room. *YOU PROVIDE THE MUSIC; BLACK HOLE FIVE WILL TAKE CARE OF THE NOISE!*

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New from ORCA!

AX-ON (Greek axon, axis): that part of a nerve cell through which impulses travel away from the cell body. AXON 8 speaker cable combines outstanding design features with component quality usually associated with the most expensive cable. With eight AXON 1 solid-core conductors and utilizing mylar/ polypropylene construction, AXON 8 offers outstanding performance for amp-speaker connections and perfectionist internal speaker wiring. Our superb AXON 1 AWG 20 solid core conductor is also available separately. Oxygen-free and 99.997% pure, it is ideal for most internal wiring applications.

Outer insulation: UL approved TPE
Cable geometry: non interleaved spiral
Individual conductor insulation: 105 degree Celsius, UL approved PVC
Cable equivalent gauge: total - AWG 11, 2 conductors - AWG 17, 4 conductors - AWG 14
Individual conductors: solid core AWG 20 copper, long-grain and ultra-soft, free of all contaminants and oxygen.
Cable core: crushed polypropylene
Inner envelope: mylar film

Reader Service #32

Guest Editorial

Are These Guys Making Music for Us?

by Richard Campbell
Contributing Editor

Once in a while I get involved in the design of a serious loudspeaker; one intended for mass production. Do I appear to insinuate that your latest loudspeaker home project was not serious? My apologies. All loudspeaker projects are serious, but the one-off do-it-yourself pair for your living room doesn't get designed or made like a production loudspeaker.

The production loudspeaker has to sound generally good to everyone, with all sorts of music playing through it in all sorts of room environments. Furthermore, it has to achieve all that with "cookie-cutter" assembly. Very tough requirements, believe me.

"All sorts of music" means that the loudspeaker designer has a monster library of CDs and tapes and a battery of willing listeners to provide criticism. I am a technocrat through and through, but I am also a one-time musician and I work with musicians in a professional recording relationship. My ears may be a tad presbycusic, but they still judge very well. I have unshakable confidence in them. I know measuring instruments tell only half the tale, so I depend on my willing listeners' ears, and mine, to tell me about the other half. We listen to a lot of music.

There is an astonishing amount of trash recording out there. I mean useless, horrible, yukky techno-crap! Does this industry try to relate to the home listener? Do the people who push the studio buttons have misplaced motives? Why can't a million-dollar studio with one of those obscene 64-track consoles just get a reverb tail to sound natural? Why are there four wall-mounted 15-inch woofers on each side of the mixing desk together with a proportionate battery of low-mids, high-mids, tweeters, and quint-amplified equalizers? I mean, are these guys making music for *us*? Who decides what it will sound like in *my* living room?

I have gone through many tracks on many CDs to find those four or five that can really tell me about my new loudspeaker. Of the 30-odd commercial CDs immediately to hand in my lab, there are only three or four mixdown masterpieces that I trust. The rest are "peculiar" in some way--weird is a better word, stamped with the personality of someone's ear from hell. Somehow, on the really good ones, the engineer sitting at that obscene console knew how to twiddle the knobs to make himself happy in his imagined living room, and me happy in my space.

You know what the people at Bose do? They too select certain tracks from CDs. They too have a battery of willing listeners. Bose has living rooms inside the factory that look just like yours and mine, averaged. The final judgment of marketability of a new product takes place there, with humans sitting on sofas and in easy chairs next to bookcases, table lamps, framed paintings on the walls, and a rug on the floor. No tube traps, slat absorbers, fractal diffusers, or designated sweet spot. Just plain folks in a just plain room. Of course, Bose has had only limited success doing it that way, right?

What's going on here? Can't these recording engineers get real? It's time to send the 90 percent who fail over to the 10 percent who succeed, to take a few lessons (they won't be let in).—R.C.

In Memoriam

Avery Robert Fisher, born in Brooklyn, NY in 1906 of Russian immigrant parents, died on February 26, 1994. His audio products introduced thousands of American families to "hi-fi" after World War II. Younger audiophiles and music lovers will remember him as a philanthropist whose gifts endowed awards and scholarships and music making at Lincoln Center. His most public gift was a $10.5 million endowment fund, a portion of which underwrote the refurbishing of the center's Philharmonic Hall, which was subsequently named in his honor.

The youngest of six children in a highly musical family, Fisher learned to play the violin, but after graduating from NYU decided against a musical career. He worked first in an advertising agency, eventually migrating to book publishing and to many years of book design for G.P. Putnam's Sons and Dodd, Mead & Company. At home in the evenings, Fisher designed and constructed music systems. He founded Philharmonic Radio as an outgrowth of his after-hours hobby.

After WWII, he sold his Philharmonic Radio and founded Fisher Radio. In staffing the new company he hired European engineers, offering much higher pay than was available in war-torn Europe. The young company early became known as a source of high-quality, well-engineered components that many called the Rolls-Royces of sound equipment. Although high fidelity was not, during its early years, a mass-market phenomenon, Fisher's products certainly were in the forefront of spreading the new idea of exceptionally high-quality sound to American homes.

In 1969, since marketing of hi-fi equipment was being taken over more and more by Japanese companies, Fisher sold his firm to Emerson Radio, who in turn sold it to Sanyo. Thereafter he devoted his life to philanthropy—"the repaying of old debts," as he called it.

Avery Fisher's interest in the arts, and especially in music, was evidenced by large gifts to a number of institutions. He served on numerous boards, including the New York Philharmonic, the Chamber Music Society of Lincoln Center, and the Marlboro Festival in Vermont. His generous annual awards for young musicians have been important to a number of careers, and will continue to be so for many years to come.—E.T.D.

A PUBLIC-AREA SOUND SYSTEM

By Hilary Paprocki

The doctor at the cocktail party, unwillingly listening to people's symptoms, is such a familiar image it's probably documented on one of the Hirsch "cultural literacy" lists. The lesson doubtless applies to all vocations and avocations: when people know you have special experience, they often enlist your expertise for their own projects or organizations.

I have assembled audio gear and systems for years, and for the past several have been an advisor at a local community college. (This is an enjoyable job, and I recommend it to anyone who is concerned about the future of kids stuck in the American pop-culture educational system.) Pretty soon I began providing equipment and mixes for entertainers and banquet speakers, and eventually, the school administrators asked me to design a new sound system for their cafeteria. It sounded like fun, so why not?

The original system, installed by a big, old, local audiovisual company, was rudimentary. All wiring was of the same shielded-pair material. Speakers consisted of six-driver columns, using 4″ table-radio speakers in small gypsum-board boxes with tiny transformers. The amplifiers were of the "if it works, it's good" school of design (*Photo 1*)—not too difficult to beat (*Photo 2*).

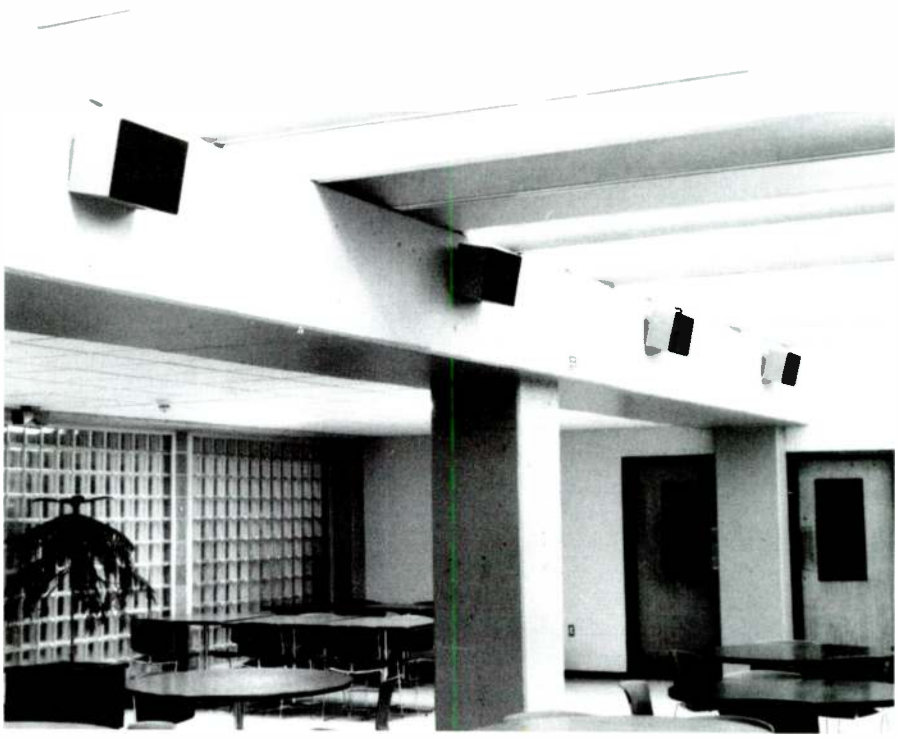


PHOTO 1: The installation. Boxes were painted with the same paint as the beams. They could probably use a little more downward tilt.

WHAT'S THE PLAN?

My main objectives were:

1. A very even distribution of sound throughout the area, which would mean using numerous individual small loudspeakers rather than a few big ones. Ideally, distribution would be so broad that one could stand in front of any loudspeaker with an open mike and not worry about feedback.
2. Enough volume to out-decibel a crowd of boisterous young people.
3. A peak-free tonal balance, preferably without electronic equalization. Equalizers are not really good for ultimate sound quality.
4. Good dynamics and clarity. I agree with the old German lensmakers: assuming all other factors are competently executed, a system that gets the contrasts right will probably produce the most lifelike results.

THE ROOM

The cafeteria measures approximately 50′ × 90′ and has three hard walls. A thin heavy-duty carpet covers about half the floor area. The ceiling is composed of concrete ribs, between which run fluorescent lamps and perforated-metal ventilators. Across the ribs are three heavy concrete beams, almost 30″ thick, one at each end of the room and one across the middle.

I decided to string loudspeakers in cabinets along the beams, a row of speakers at each end of the room facing in and rows on each face of the center beam facing out. Multiple arrival times might smear the sound quality, but given enough sources, we could avoid the sensation of two or three discrete arrivals. The final effect would be like listening to a Bose 901 system: the time focus might be a little hazy, but not as painfully distracting as a few hard echoes.

Four rows of four 8Ω speakers would provide two 4Ω channels, thereby allowing the use of a standard hi-fi amplifier without output transformers. Speaker transformers seriously degrade sound quality, and cost a lot. Don't use them.

IF EVER A WHICH THERE WAS

I spent a few weeks touring schools, restaurants, and sundry other public spaces looking for a loudspeaker smooth enough and tight enough for my taste, and cheap enough to install by the dozen. It didn't exist. My Mastercard and I temporarily bought a carful of small hi-fi speakers, took them to the school,

ABOUT THE AUTHOR

Hilary Paprocki is an industrial equipment salesman with 29 years in on this hobby. He has lived in Rochester, NY since the early 1960s.

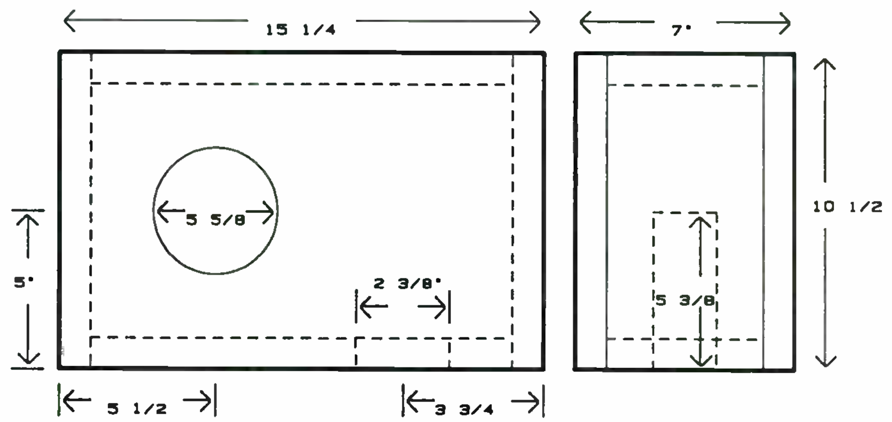


FIGURE 1: The box dimensions.

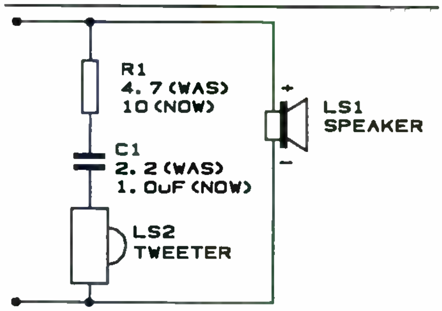


FIGURE 2: Diagram of the crossover networks. The crossover supplied should be replaced for a clear and natural sound.

and ran a bark-off. No winners emerged. Even the relatively expensive models from industry leaders like JBL sounded awful, with screaming tweeters and very approximate woofers.

The best system I encountered was the little Bose 101, a plastic-box single-driver system. Its tonal balance was imperfect, with the 300Hz-or-so howl that many 4" woofers seem to have. Unlike all the others, though, it didn't sound phase-confused up and down the audio range. Nor did it have any bad high-Q resonances or rings; it probably could have been equalized into an acceptable state. Still, it was a bit dynamically compressed, and I wanted to avoid running an equalizer.

About this time I received a flyer describing Soundolier's new Strategy Series of drivers. Even before reading the spec sheet on the model FA136, I was impressed. These drivers are built on a 6.5" frame, ideal for what I was doing. Moreover, they seem to be built of the kinds of stable, well-behaved materials audiophiles prefer. The cones are Kevlar® fiber-reinforced polypropylene. The woofer surrounds are of a convex Norsorex material, ideal for this application. The tweeter is small, coaxially mounted, fluid damped. The crossover incorporates a Mylar® capacitor rather than the usual low-fi electrolytic.

If the FA136 sounded anything like the way it should with all these features, this was my baby. The local distributor provided a single sample and I hurried back to the bench to box it.

BOXING MATCH

With systems in the cafeteria and a smaller adjoining area, this project was going to require about 500' of speaker wire. That's a lot, especially for picky audiophile nuts who aren't satisfied with the small stranded zipcord most people envision as "speaker wire." I had to find some kind of acceptable, affordable wire, and design the cabinets to suit.

We could all learn a lot from the audio pioneers' work on methods for bringing intelligible sound over thousands of miles of wire. Take your basic telephone wire—Belden 1242A, for instance. This cable is made of four individually insulated 22-gauge solid copper wires; typical stuff, sold by the mile at your local distributor's for about four and a half cents a foot. Communication with Belden revealed this particular model number to be composed of 99.9999% pure copper. Purity like this is in the league of multidollar audiophile cables. Moreover, the insulation is of polyethylene, which ain't Teflon® but is way above vinyl in its dielectric performance. Belden 1241A is excellent stuff.

I elected to use two separate runs of cable to each terminal; that is, to twist the four wires in each cable together and use them as an 18-gauge run. So the price goes up to nine cents. No problem there. And cable capacitance virtually disappeared, since the send and return lines weren't enclosed within the same jacket.

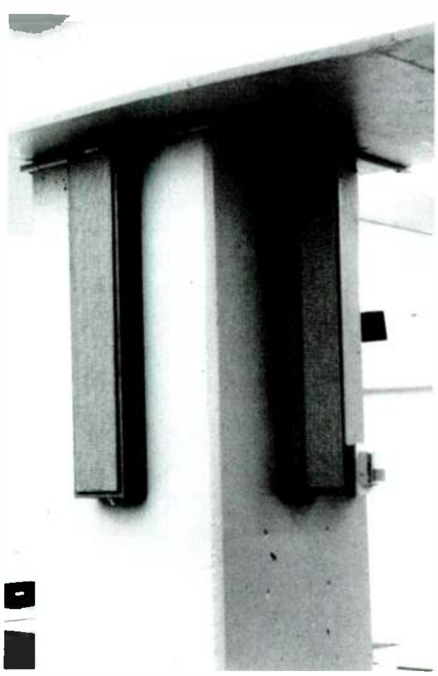


PHOTO 2: The old speakers in the cafeteria. I think I can beat this.

The largest cable resistance, at the longest run to and back from the farthest loudspeaker, would be a shave under an ohm, I figured. The Sitting Duck software produced box volumes for "Standard B4 Alignment" cabinets suitable for 0.5, 1.0, and 1.5Ω source resistances. The cabinet size tended to grow as source resistance increased (as the wire got longer), which was acceptable.

The final box was the 1.0Ω model, built of ⅝" chipboard (or fiberboard, or novaboard, or cheeseboard, or whatever this stuff is properly called). The port went on the bottom panel, to allow a little more clearance (I wanted the boxes to be on the shallow side) and to simplify the baffles. For economy, bracing was ignored (*Fig. 1, Photo 2*). Finally, the time came to listen.

The speaker didn't sound exactly perfect right off the bat, but it showed plenty of promise. A bit of a peak, around 60–80Hz, was actually rather pleasant. Worse was the overloud tweeter nonaudiophiles favor—easily fixed by replacing the factory 4.7Ω, 2.2μF series filter with a 10Ω, 1.0μF network. I agonized over the choice between a 10Ω and a 12Ω resistor, then measured the results and found the difference to be less than 1dB at 10kHz. This is the kind of thing we go nuts over. I left it at a slightly sparkly-sounding 10Ω. The audience would prefer it, and besides, a grille cloth installed on the finished product might dull the highs a bit (*Fig. 2*).

I tuned the bass response with a collection of rings cut from the same 2" plastic pipe that comprised the vent. Since the port respires from a side panel on the cabinet, you can listen to the system with the vent side turned up, and just place a ring over the opening to extend the port. A cabinet that is a little boomy is probably a little small. Fortunately, another alignment works with a smaller cabinet and a longer port. A ¼" ring glued to the inside end of the port tube removed most of the boom, leaving just a tiny, pleasant hint of bass heft.

We stuffed the boxes with cotton batting, effective, inexpensive, and nonobnoxious. The stuff won't attack your skin the way glass fiber does, but is a bit messy; have a vacuum cleaner handy. Figure about one-half square yard per box, and don't worry about fastening it inside. If it's there and not all bunched up, it'll work.

The 5⁄8" material used on the prototype caused a little panel vibration, so in the final batch of cabinets we used 3⁄4" stuff. The added mass helped physical stability and hence sound quality. People who mount their drivers in rubber are nuts. Imagine sending a specific electrical waveform to a loosely mounted speaker driver. The cone goes forward, the frame goes back in some weird nonlinear way, and who knows what the resultant transmitted acoustical waveform may be? [*Those rubber-encasing nuts argue that the rubber damps transfer of signals, lowering extraneous panel resonances.—Ed.*]

Don't bother mounting the drivers with threaded inserts. You'll be building these by the bunch, and detail work like that will kill you. Set up your drill with a small bit and your power screwdriver with a box of power-drive or drywall screws, and just drill the holes and run the screws in, one after the other.

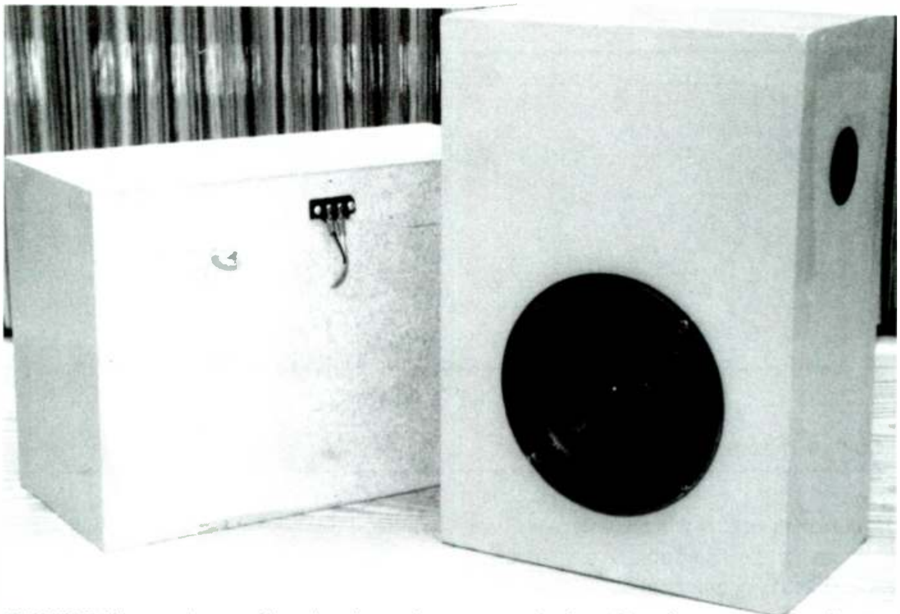


PHOTO 3: The new boxes. Note that the tuning ports are in the cabinet bottom panels, allowing a little more length inside the box and simplifying the grille panel.

My final choice for grilles was panels of 1⁄2" plywood with 7" holes to fit over the cabinet baffle panels. As these are flat and solid, they don't present a lot of trip points to generate diffractions. Cut extras; some will warp. They are wrapped in a nice, transparent knit black polyester from the discount store, 50" wide, at a big 99 cents a yard. *Do not use Velcro®* to mount the panels. When they fall off, and they will, you'll have to deal with whomever they fall on. Use two black drywall screws per box. Make holes in the cloth with your soldering iron and carefully drill holes, a little oversize, in the grille panels. Shoot the screws in and paint the screwheads with satin black paint.

I installed the speakers with plain old hardware L-hooks and screw eyes. (You should all ask your hardware stores to stock Rotanium Saf-T-Anchors, the only anchors that are any good. They're hard to find.) The L-hooks went into the concrete, the screw eyes went into the cabinet backs; and the resulting assembly provided a nice tilt to aim the drivers down toward the listeners. Avoid pointing speakers at walls. Your sound should hit the audience, get absorbed, and stop there.

SOUND

These speakers won't sound great in your stereo. That's not what they're for. But did they sound great in the cafeteria? Absolutely! They were sharp and precise, with smooth response and none of the nasty peaks and resonances of the cheap home hi-fi speakers I tried. Dynamics were clear and quick. There seemed to be a dip in the system's response around 250Hz, which is a blessing, since many mid-sized public rooms tend to boom in that range. (Was Soundolier really hip enough to do this on purpose?)

The installed cable resistance turned out to be a little lower than planned (I put the amplifiers in the ceiling), and the little bit of bass emphasis I hoped for never really materialized. But the bass was so clear you could easily distinguish one bass player's instrument from another's. Try that with any other cafeteria sound system—or most people's stereos, for that matter.

Treble was sweet and transparent. The cymbals in the chorus of James Brown's "I Feel Good" seemed to leap out of the boxes and into the room. Clean recordings sound clean. CDs with ridiculous loud treble sound as bad as intended. Live voice is crisp, natural, and intelligible. Sure, the speakers would benefit from a little equalization, but no one but the installer will notice.

These loudspeakers certainly represent an advance on what is available in commercial sound today; and they're cheap. So the next time your mom tells you about the fights at the bingo hall because nobody hears the numbers right, use this article to make her a heroine!

SOURCES

Atlas/Soundolier Div.
1859 Intertech Dr., Fenton, MO 63026
Strategy Series FA136 loudspeakers

Belden Wire and Cable
PO Box 1980, Richmond, IN 47375
1242A telephone cable

Premier Industrial Corp.
4500 Euclid Ave., Cleveland, OH 44103
Rotanium Brand concrete screw anchors

A MULTITAP AIR INDUCTOR

By Kim Girardin

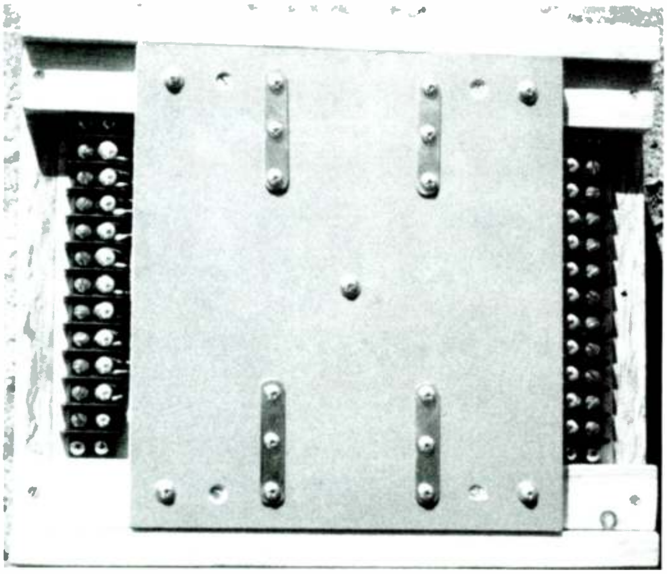


PHOTO 1: A 20tap coil with a built-in breadboard.

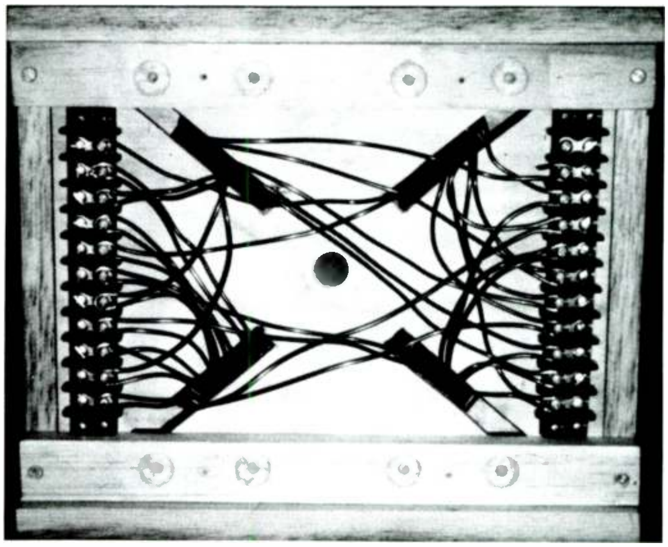


PHOTO 2: A 20tap coil with breadboard removed. The form measured 10" × 12" and was wrapped with ¾" pine.

Originally, I'd set out to make an inductance decade box. I bought two sets of coils with values of 0.1, 0.2, 0.3, 0.4, 1, 2, 3, 4 and 10mH. With these nine coils, when connected in series (inductance adds in series), I could get any value I needed (in 0.1mH increments) up to 21mH. The obvious problem with that scenario is that the DCRs of the coils also add and with the switches and connecting wire the total DCR was measuring around 2Ω. Generally, the acceptable limit for the resistance added to a speaker system by a passive crossover network is around 5% of the nominal impedance of the speaker, or 0.4Ω for an 8Ω speaker and 0.2Ω for a 4Ω speaker. Maybe the problem wasn't obvious, since I obviously built the thing before discovering the obvious.

I dismantled the box. If I needed a "custom" value I'd wind one. I made a spreadsheet to help in this. While playing around with core sizes, DCRs, and wire gauges I noticed that if you double the length of wire you triple the inductance. One thought led to another. After redesigning the spreadsheet to look at the change in inductance for each additional layer of wire, I decided to make a set of three 10tap coils. Big coils! 10GA air-core coils! If I measured the inductance as I wound, instead of counting turns, an accuracy equal to that of the meter could be maintained. I replaced my bargain 3% LCR meter with a BK Precision model 878. The 878 has an accuracy of 0.7% in the 100mH range, 1.2% in the 10mH, and 2% in the 1mH.

I'd make one coil to cover 0.01–0.1mH, another for 0.1–1 and another for 1–10mH. If a larger value was needed a 10mH wound on a ferrite bobbin could be used. I got an 80-lb spool of 10GA and one of 12GA. I made a couple 12GA 1mH 10taps. No problem. I made a 10GA 10mH 10tap. It was huge! I just looked at it for a long time. After a day or so, I decided to remeasure it to see if the values at all the taps had mysteriously drooped or something under the sheer weight of the thing. They all were the same as yesterday.

HENRIES ON TAP

Out of curiosity, instead of measuring from the "zero" tap as it had been wound, I began to touch the meter leads to random combinations of taps. I was getting readings all over the place. In-between values. I logged every possible combination and found I hadn't made a 10tap coil—I'd made a 55tap coil. Most of the values were less than 5mH. Some measured so close to others that the differences were insignificant.

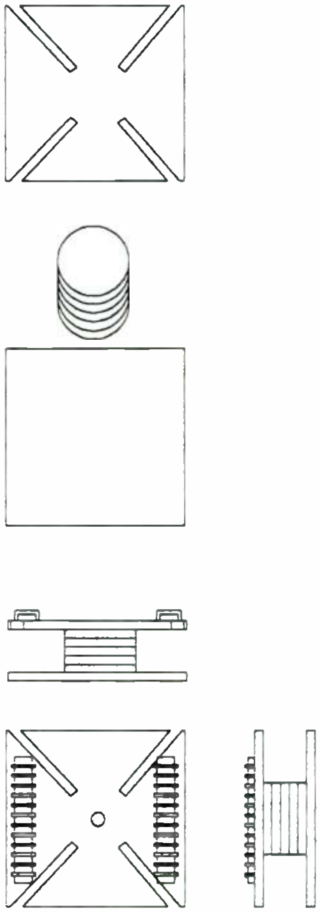


FIGURE 1: Winding spool.

ABOUT THE AUTHOR

Kim Girardin has been involved in sound reinforcement since 1974. He is currently employed as a technician by the Intelligo Corp.

TABLE 1

20mH AIR CORE PARTS LIST

	DESCRIPTION
1	BK 878 LCR meter
1	700′ 10GA magnet wire, about 23 lb
1	10-position barrier block, Cinch part 10-142
1	11-position barrier block, Cinch part 11-142
2	10″ × 10″ ½″ plywood
5	4″ diameter ½″ plywood
10	¾″ screws (brass) OR (1) 2″ screw (brass or stainless)
1	¾″ dowel
1	8″ 1 × 2
1	simple winding stand

In a couple of ranges there were gaps where it would have been good to have a couple values for tweaking.

I made a 10GA 20mH air core with 20 taps in 1mH increments. There are 210 values of inductance available, depending on which pair of taps you select (*Tables 2* and *3*). The values range from 0.0276–20mH. In other words, if used as a first-order low pass with an 8Ω load the range of frequencies covered is from about 65Hz to 40kHz—all this from one coil! The maximum DCR is 0.68Ω (*Table 4*). (I took DCR measurements with a Fluke 8060A.) This is about 0.05Ω more than a 10GA 20mH wound to optimized dimensions with no taps. 10GA is 1,001′/Ω. The discrepancy of 0.05Ω is equal to about 50′ of wire. About 15′ of that are in the leads coming out to the taps, so about 35′ (.035Ω) of wire are needed to make up for the chaos caused by bringing the wire out to the taps.

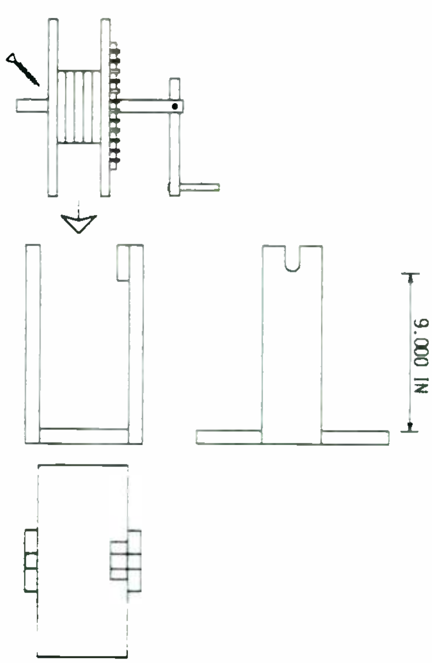


FIGURE 2: Stand and spool.

TABLE 2

A HINT FOR USING THE COIL

Tap	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19
1	1.000																			
2	1.998	.21																		
3	2.995	.65	.15																	
4	3.995	1.23	.46	.11																
5	4.979	1.85	.89	.36	.08															
6	6.004	2.55	1.42	.72	.30	.07														
7	6.985	3.27	1.98	1.10	.60	.26	.06													
8	7.976	4.01	2.58	1.64	1.02	.52	.21	.05												
9	8.999	4.80	3.25	2.20	1.44	.87	.40	.20	.05											
10	9.978	5.58	3.92	2.78	1.93	1.27	.79	.42	.19	.052										
11	10.97	6.38	4.60	3.38	2.40	1.73	1.14	.70	.39	.169	.039									
12	11.95	7.19	5.33	4.02	3.01	2.20	1.54	1.03	.646	.350	.149	.038								
13	12.96	8.03	6.09	4.68	3.60	2.72	1.99	1.4	.952	.584	.326	.145	.039							
14	13.99	8.88	6.87	5.39	4.24	3.30	2.49	1.85	1.328	.887	.555	.309	.137	.037						
15	15.00	9.79	7.67	6.12	4.90	3.89	3.01	2.31	1.725	1.226	.843	.539	.306	.133	.040					
16	15.98	10.61	8.44	6.84	5.53	4.49	3.54	2.79	2.148	1.588	1.150	.788	.501	.275	.122	.0388				
17	17.00	11.51	9.29	7.60	6.25	5.12	4.12	3.32	2.618	2.033	1.511	1.098	.757	.472	.286	.1125	.0306			
18	18.01	12.40	10.09	8.36	6.95	5.77	4.72	3.86	3.113	2.449	1.904	1.439	1.047	.711	.451	.2469	.1111	.0298		
19	19.02	13.28	10.88	9.12	7.67	6.44	5.35	4.42	3.625	2.908	2.310	1.798	1.357	.978	.664	.4140	.2345	.1040	.0276	
20	20.00	14.15	11.72	9.88	8.37	7.10	5.95	4.99	4.147	3.387	2.751	2.199	1.712	1.280	.9380	.6244	.4025	.2207	.0972	.0303

TAPS

A HINT FOR USING THE COIL

Trace down from tap 6 and over from tap 14.
Where the column and row intersect is the value 2.49mH.
Look at the values along the diagonal descending left to right through the value 2.49mH. Notice the values along these diagonals change in approximately 10% increments. By selecting taps with the same interval between them you can make rapid subjective tweaks.

For example:

TAPS	mH	%CHANGE
4-12	3.01	
5-13	2.72	10.7%
6-14	2.49	10.9%
7-15	2.31	10.8%
8-16	2.14	10.8%

This pattern will not hold true for any value using the 0 or 1 tap. Because there are so many more windings in the first two taps, they can affect changes of up to 500%!

BOXES CONTAIN VALUES IN mH

WINDING DOWN

The coil can be wound on a form that measures 10″ × 10″ with a 2½″ × 4″ diameter core (*Fig. 1*). (I chose to use a form 10″ × 12″ to allow enough room for a breadboard [*Photo 1*].) Four slots ½″ wide are cut into the top. The form is glued and clamped. Drill a ¾″ hole through the center of the form.

Mount the two barrier blocks with ¾″ brass or stainless steel screws *(Fig. 2)*. Insert a 12″ dowel through the hole so that 2″ protrude from the back. Drill a hole, at about a 45° angle (or so), through the dowel and countersink it. Through this hole attach the dowel to the core with a screw. A handle is attached to the dowel. Make sure the stand is tall enough to permit the form to turn. You may have to screw the base of the stand to something to secure it.

Remove the enamel from the end of the wire and solder. Terminate to the "zero tap." Start winding. After about three layers you'll be getting close to 1mH, so start measuring at this point. To take measurements you must connect one lead of the LCR meter to the zero tap and pierce the enamel with the other. (A sewing needle in the jaws of an alligator clip works well, but will dull quickly. Have several on hand or use a file to resharpen.) Position the wire at one of the four slots and measure. Add or subtract windings as needed. You should be able to get within 10 or 20µH of the desired value. Keeping the windings snug, pull the wire through a nearby slot, lay it across the terminal block to measure, clean off the insulation for a distance that will make good contact with the screw head, and then, after applying solder and wrapping it under the screw, take the wire back through the same slot to continue the wind (*Photo 2*).

The second mH will be achieved after two

SOURCES

REA Magnet Wire Company, Inc.
3600 East Pontiac St., Fort Wayne, IN 46896-0128
(or contact your local electric motor servicing shop)
10GA magnet wire

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

Contact East
335 Willow St., North Andover, MA 01845-5995
BK 878 LCR meter and Fluke 8060A

TABLE 4

DCRs FROM THE ZERO TAP

mH	Ω	mH	Ω	mH	Ω
1	0.11	8	0.38	15	0.56
2	0.17	9	0.41	16	0.58
3	0.21	10	0.44	17	0.60
4	0.25	11	0.46	18	0.62
5	0.28	12	0.49	19	0.65
6	0.33	13	0.51	20	0.67
7	0.36	14	0.53		

more layers. After that each additional mH will take "about" one layer. Do your best to keep the windings tight and compact. No

Continued on page 72

Product Availiblity

I have a variety of 10 and 20tap coils available. Take a look at the ad in the trade section of the want ads. For custom increments or values write: Kim Girardin, PO Box 1181, Winona, MN 55987.

TABLE 3

THE VALUES IN SEQUENCE

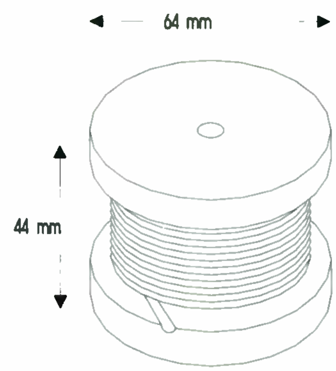
TAPS		mH	TAPS		mH	TAPS		mH	TAPS		mH
18	19	0.0276	5	8	0.5200	6	13	1.9900	6	19	5.3500
15	16	0.0288	11	15	0.5390	0	2	1.9980	3	14	5.3900
17	18	0.0298	10	14	0.5550	9	17	2.0330	4	16	5.5300
19	20	0.0303	9	13	0.5840	8	16	2.1480	1	10	5.5800
16	17	0.0306	4	7	0.6000	11	20	2.1990	5	18	5.7700
13	14	0.0370	15	20	0.6244	3	9	2.2000	6	20	5.9500
11	12	0.0380	8	12	0.6460	5	12	2.2000	0	6	6.0040
10	11	0.0390	1	3	0.6500	7	15	2.3100	2	13	6.0900
12	13	0.0390	14	19	0.6640	10	19	2.3100	3	15	6.1200
14	15	0.0400	7	11	0.7000	4	11	2.4000	4	17	6.2500
7	8	0.0500	13	18	0.7110	9	18	2.4490	1	11	6.3800
8	9	0.0500	3	6	0.7200	6	14	2.4900	5	19	6.4400
9	10	0.0520	12	17	0.7570	1	6	2.5500	3	16	6.8400
6	7	0.0600	11	16	0.7890	2	8	2.5800	2	14	6.8700
5	6	0.0700	6	10	0.7900	8	17	2.6180	4	18	6.9500
4	5	0.0800	10	15	0.8430	5	13	2.7200	0	7	6.9850
18	20	0.0972	5	9	0.8700	10	20	2.7510	5	20	7.1000
17	19	0.1040	9	14	0.8870	3	10	2.7800	1	12	7.1900
3	4	0.1100	2	5	0.8900	7	16	2.7900	3	17	7.6000
16	18	0.1111	14	20	0.9260	9	19	2.9080	2	15	7.6700
15	17	0.1125	8	13	0.9520	0	3	2.9950	4	19	7.6700
14	16	0.1220	13	19	0.9780	4	12	3.0100	0	8	7.9760
13	15	0.1330	0	1	1.0010	6	15	3.0100	1	13	8.0300
12	14	0.1370	4	8	1.0200	8	18	3.1130	3	18	8.3600
11	13	0.1450	7	12	1.0300	2	9	3.2500	4	20	8.3700
10	12	0.1490	12	18	1.0470	1	7	3.2700	2	16	8.4400
2	3	0.1500	11	17	1.0980	5	14	3.3000	1	14	8.8800
9	11	0.1690	3	7	1.1000	7	17	3.3200	0	9	8.9990
8	10	0.1900	6	11	1.1400	3	11	3.3800	3	19	9.1200
7	9	0.2000	10	16	1.1500	9	20	3.3870	2	17	9.2900
1	2	0.2100	9	15	1.2260	6	16	3.5400	1	15	9.7900
6	8	0.2100	1	4	1.2300	4	13	3.6000	3	20	9.8900
17	20	0.2207	5	10	1.2700	8	19	3.6210	0	10	9.9780
16	19	0.2345	13	20	1.2830	7	18	3.8600	2	18	10.0900
15	18	0.2469	8	14	1.3210	5	15	3.8900	1	16	10.6100
5	7	0.2600	12	19	1.3570	2	10	3.9200	2	19	10.8800
14	17	0.2660	7	13	1.4100	0	4	3.9950	0	11	10.9700
13	16	0.2750	2	6	1.4200	1	8	4.0100	1	17	11.5100
4	6	0.3000	11	18	1.4390	3	12	4.0200	2	20	11.7200
12	15	0.3060	4	9	1.4400	6	17	4.1200	0	12	11.9500
11	14	0.3090	10	17	1.5110	8	20	4.1470	1	18	12.4000
10	13	0.3260	6	12	1.5400	4	14	4.2400	0	13	12.9600
9	12	0.3500	9	16	1.5880	7	19	4.4200	1	19	13.2800
3	5	0.3600	3	8	1.6400	5	16	4.4900	0	14	13.9900
8	11	0.3900	12	20	1.7120	2	11	4.6000	1	20	14.1500
16	20	0.4025	8	15	1.7250	3	13	4.6900	0	15	15.0000
15	19	0.4140	5	11	1.7300	6	18	4.7200	0	16	15.9800
7	10	0.4200	11	19	1.7980	1	9	4.8000	0	17	17.0000
14	18	0.4510	1	5	1.8500	4	15	4.9000	0	18	18.0100
2	4	0.4600	7	14	1.8500	0	5	4.9790	0	19	19.0200
13	17	0.4720	10	18	1.9040	7	20	4.9900	0	20	20.0000
6	9	0.4800	4	10	1.9300	5	17	5.1200			
12	16	0.5010	2	7	1.9800	2	12	5.3300			

SLEDGEHAMMER AUDIO INDUCTORS

MADISOUND is pleased to offer a new line of LOW distortion inductors. We are utilizing the good magnetic characteristics of high permeable steel alloy to give you inductors with high power capability, low DCR and low distortion. The inductors show significantly lower distortion in the low frequencies and will give cleaner sound reproduction for bass drivers, especially subwoofers. Just as you would use polypropylene capacitors for a noticeably cleaner sound in your midranges and tweeters, Madisound Sledgehammer Steel inductors will give you a noticeably cleaner bass. Both the Sledgehammer Steel Bobbin and Sledgehammer Steel Laminate use hard drawn 15 gauge copper wire for maximum conductivity and low DC resistance.

SLEDGEHAMMER STEEL BOBBIN

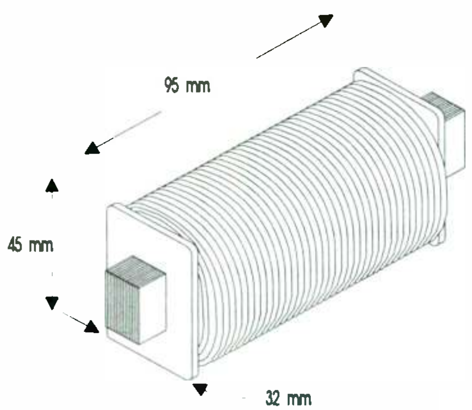
- 15 gauge copper wire
- 1" I.D. center core and 3/8" flange thickness for high current and less saturation.
- Saturation above 500 watts
- Durable construction, no breakage
- Low distortion and low DCR



mH	DCR Ω	Price	mH	DCR Ω	Price
1.0	.092	$11.25	3.0	.179	$13.50
1.25	.105	11.75	3.3	.189	13.60
1.5	.114	12.00	3.7	.204	13.75
1.7	.126	12.25	4.0	.213	14.00
2.0	.139	12.50	4.3	.226	14.75
2.25	.149	12.75	4.7	.237	15.00
2.5	.156	13.00	5.0	.248	15.25
2.7	.166	13.25	6.0	.288	16.25

SLEDGEHAMMER STEEL LAMINATE

- 15 gauge copper wire
- Audio grade 100% grain oriented steel alloy
- Lowest distortion
- Best for subwoofer applications
- Saturation above 400 watts
- Low DCR



mH	DCR Ω	Price	mH	DCR Ω	Price
4.0	.210	$11.25	8.0	.345	$13.50
4.5	.231	11.75	8.5	.357	13.75
5.0	.242	12.00	9.0	.364	14.50
5.5	.257	12.25	10.0	.376	14.75
6.0	.272	12.50	12.0	.426	15.50
6.5	.287	12.75	13.0	.450	16.75
7.0	.307	13.00	15.0	.505	17.50
7.5	.320	13.25			

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SEARCH FOR A BUDGET SUBWOOFER

By John F. Sehring

Some CDs deliver sounds as low as 10Hz, but the usual loudspeaker can't do them justice.[1] My budget doesn't allow the purchase of a full-blown subwoofer system, so I did some experimenting, and accomplished a surprising amount with equipment at hand.

WHY A SUBWOOFER??

First, because there is a trade-off with loudspeaker placement. For best stereo imaging, speakers should be raised up off the floor and centered on and well away from back and side walls. This location reduces the number and strength of early reflections, which can confuse the ear/brain and impair the credibility of an audio image. In most rectangular rooms, this central location tends to excite the fewest number of resonant modes, which occur as standing waves set up by sound energy bouncing between walls. Sounds around the modes' frequencies will be considerably louder.

Fewer than 10% of a room's modes are excited when a loudspeaker is at the room's center. This minimizes the strong effects of a room's frequency response on that of a loudspeaker. Unfortunately, it is also the *least* desirable location for bass reproduction, for the same reason. For best efficiency in reproducing the lowest frequencies we *want* to excite a room's resonant modes, as they most effectively distribute low-frequency energy.

Modes are best excited at points of maximum acoustic pressure—the walls, ceiling, and floor. A loudspeaker located at a boundary is most fully coupled to the room's modes, giving maximum loudness for a given amount of power; i.e., highest efficiency.

Successively better spots for maximizing bass are: (1) in the middle of the floor or wall; (2) at the junction of a wall and floor; (3) in a corner. The corner location is best from a bass point of view, as a loudspeaker located there is capable of fully exciting all of a room's modes, including the lowest frequency.[2,3]

Nondirectional low-frequency sound is also reflected off surfaces immediately adjacent to a loudspeaker. The effect of this "room gain" (not to be confused with room resonances) becomes stronger at lower frequencies. It is typically 6dB at 30Hz.

Another reason for using a subwoofer is to help the main speakers in the midrange. With a subwoofer, the audio spectrum is usually split using a crossover at some low frequency, generally below 100Hz. Energy below the crossover frequency is sent to the subwoofer(s) and energy above to the main speakers. So apart from augmenting bass output, a subwoofer can also relieve the main speakers and amplifiers of the burden of reproducing deep bass.

Deep-bass energy makes considerable power demands on amplifiers. It can also intermodulate higher-frequency sounds from the loudspeaker via doppler distortion, and by driving a loudspeaker into its nonlinear regions. Essentially, the higher frequencies are being radiated from a cone that may be moving excessively because of large low-frequency inputs. So a subwoofer can help a system sound cleaner in the midrange.

EQUIPMENT

Several years ago, I retired my 15-year-old Dynaco A-25II speakers, a two-way design with a 10″ woofer in a 1,200-in^3 resistive-port-vented enclosure. The −3dB response point is about 67Hz. It is capable of putting out energy as low as 30Hz without undue doubling or distortion. Low-frequency distortion is less objectionable than at higher frequencies. The impedance is well behaved at the low end, about 11Ω, only slightly reactive, at 20Hz.[4,5]

I knew that these Dynacos overloaded gracefully at low frequencies and displayed good cone control when fed infrasonic inputs. They appeared to be good subwoofer candidates.

My 30-year-old secondhand Fisher 35/35W RMS tube-type stereo amplifier was in fine condition, having a power bandwidth of 10–100kHz at full output, with reasonably low distortion. Tests with an audio oscillator, oscilloscope, and dummy loudspeaker load confirmed the low-end limit.

Electronic, computer, and ham-radio flea markets are good sources for such well-made units, from Fisher, H.H. Scott, Bogen, Harman-Kardon, Heath, and others. Always get a schematic (wiring) diagram if you can, and look for equipment that is physically heavy, which usually means lots of steel in the power supply and output transformers. A direct correlation exists between the weight of a transformer and its ability to deliver power.

I split the output from my Hafler DH-101 preamplifier to drive both the main power amplifier (a Dyna 400) and the subwoofer amp, using a Y-connector having one RCA phono plug and two RCA phone jacks. The Fisher's input impedance is 500kΩ, about 10 times higher than the main amplifier, so I expected no interaction with this direct connection.

Since a subwoofer needs to see only very-low-frequency energy, I turned the bass tone controls on the subwoofer amplifier to full boost and the treble tone controls to full cut. Loudness contour circuits provide bass boost, giving increasing bass at lower volume-control settings, so it was also engaged. The tweeter level control on the Dynaco speakers was turned all the way down—which on this loudspeaker turns the tweeter completely off.

By keeping much musical energy above the subwoofer range out of the subwoofers, I sought to prevent their output from interfering with that of the main speakers and spoiling imaging, transient response, and spatial and tonal balance. My goal was not to impair the performance of the main speakers in any way just to gain more low bass.

The main speakers are two-way systems in resistive-port-vented enclosures of about 2,000 in^3 internal volume. They are mounted on 24″ stands, about 6′ apart and centered 3′ from the long wall behind them (*Fig. 1*), which enhances imaging qualities but necessarily compromises their ability to generate low-bass loudness.

My listening room measures 19.5′ × 11′ ×

Continued on page 22

ABOUT THE AUTHOR

John Sehring received a BS in physics from Fairleigh Dickinson University. He has worked in mechanical and electrical engineering, and has been involved in earthquake testing and simulation, computer simulation of injection molding of plastics, and teaching physics. Audio, amateur radio, and personal computers are among his interests.

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DUPONT

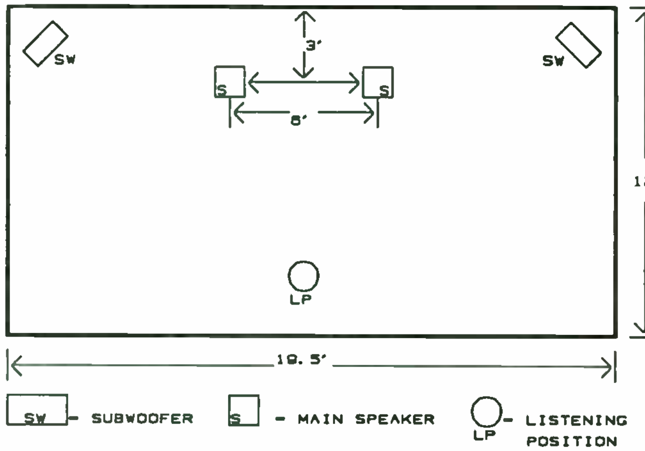


FIGURE 1: Loudspeaker placement in room.

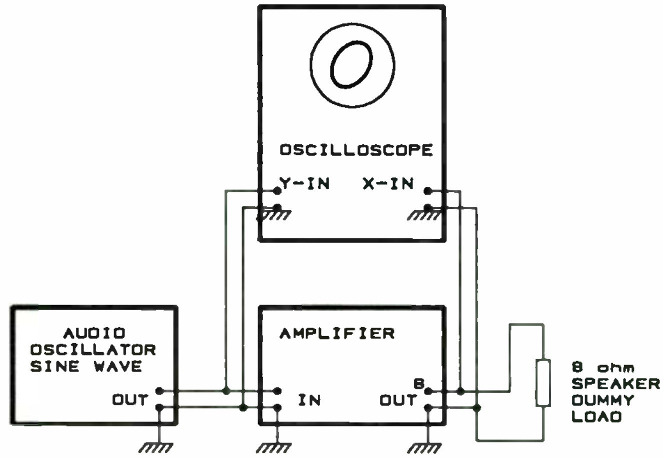


FIGURE 2: Setup for measuring amplifier phase shift.

Continued from page 20

8′. The lowest frequency room mode is associated with a room's longest dimension, its length. The formula

$$\text{Frequency(Hz)} = \frac{1{,}130}{\text{Wavelength(feet)}}$$

gives the relationship between the wavelength of sound in air and its frequency. Since a *half* wavelength of sound energy can be supported in a room as a standing wave, we divide the result by 2, which gives 29Hz for the lowest mode of the room.

This is the lowest frequency at which loudness can be efficiently generated in this size room. I wanted the subwoofer to support this mode. Numerous modes also exist in a room that are associated with combinations of (1) the room's length, width, and height; (2) bounce geometries; and (3) harmonics. This causes additional room/loudspeaker frequency response interactions.

PHASING

With a more sophisticated system, a crossover would simultaneously roll off the main loudspeaker's response and roll on the subwoofer's. With this simple subwoofer system, however, the low-frequency end of the main speakers' response is not rolled off, and both the subwoofer and the main speakers will therefore produce output at low (subwoofer) frequencies. To avoid cancellation and other interference effects, they must be in phase with each other in the subwoofer frequency range. The first step, therefore, was to find the overall phasing between the main amplifier and speakers and the subwoofer amplifier and subwoofers.

The main versus subwoofer phasing turned out to be close to 0° at 30Hz. I checked it by simultaneously feeding the same low-frequency sine wave signal to one main loudspeaker (driven by the main amplifier, tone controls flat) and one subwoofer (driven by the subwoofer amplifier, tone controls flat), with the two speakers set up closely facing each other an inch or so apart. I could achieve a high degree of cancellation of low-frequency sound with one set of loudspeaker leads reversed (180° out of phase). This indicated close to 0° phase difference between the two.

I then placed the subwoofers at the corners of the room, behind the main speakers—just the right spot to maximally excite all room modes, and achieve the most room gain (*Fig. 1*). Although it required lots of boost, I was able to excite the lowest room resonant mode at 29Hz. Those low notes were window rattling!

I was surprised how loud this setup could play with only 70W available, easily rattling walls at the room's resonant modes. The subwoofer loudspeaker's sensitivity at midband frequencies is 88dB SPL at 1m for 1W drive in an anechoic environment. The subwoofer's anechoic response is more than 22dB down at 30Hz, but room effects make up for some of that.

SUSPICIOUS PHASING

My ears are particularly sensitive to out-of-phase audio fields, and something didn't sound quite right. If a pair of loudspeakers is connected out of phase (180°), for example, you won't get any stereo effect at all. The sound image jumps about as you move your head and you can't localize the sound. Bass reproduction is also thin.

I repeated the phasing check as before, this time with the subwoofer amplifier tone controls fully engaged. Results now showed an out-of-phase condition between the main and subwoofer systems, where it had been in phase before.

To get actual phase-shift measurements of the amplifier, I hooked up equipment as shown in *Fig. 2* and fed the output of the oscillator to the Y-axis input of the oscilloscope and the output of the amplifier to the X-axis. (Any kind of scope will do.) You will see Lissajous patterns, which show us the amount of phase shift taking place in the amplifier (*Fig. 3*).

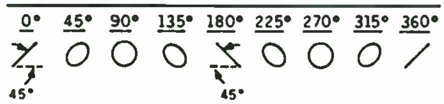


FIGURE 3: Phase shift (Lissajous) patterns.

With the tone controls set flat, phase shift through the amplifier was fairly close to 0° (in phase), except below 20Hz, where it was slightly leading. But with the tone controls set at full bass boost, treble cut, and loudness contour on, the phase shift between 20 and 60Hz roughly bracketed −180° (lagging). The lag increased with frequency. I had originally checked the phasing with the subwoofer amplifier tone controls set at flat, and missed this phase shift.

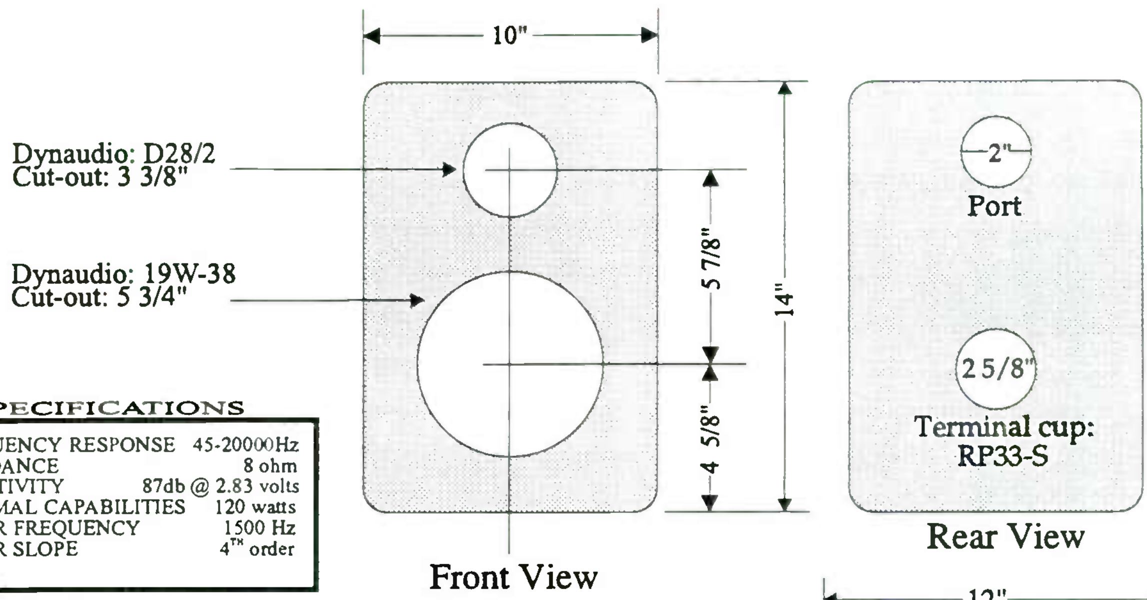
None of this surprised me—the frequency response of the amplifier is heavily skewed from flatness by all the tone-control and loudness-contour action, and requires electrical networks that have inherently large amounts of phase shift. So the subwoofers thus filtered were substantially out of phase with the main speakers. All that was needed was to reverse the polarity of wires to the subwoofers; that's the lucky thing about 180° shifts.

Too much higher-frequency (above 30Hz) sound was still coming from the subwoofers, however. The tone and loudness contour controls weren't able to roll off fast enough by themselves—and of course they weren't designed to do so.

FILTERING

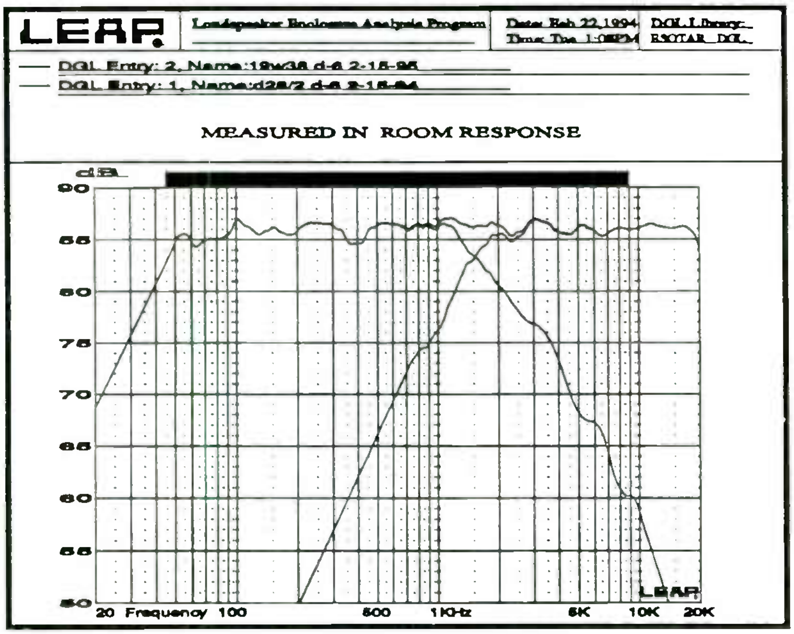
I needed something that would more rapidly attenuate frequencies above about 30Hz to the subwoofer. Subwoofer crossovers are commercially available, of course, but are complex and costly.

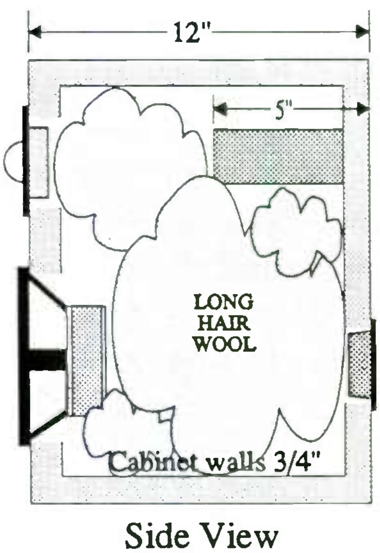
Continued on page 24



SPECIFICATIONS

FREQUENCY RESPONSE	45-20000Hz
IMPEDANCE	8 ohm
SENSITIVITY	87db @ 2.83 volts
THERMAL CAPABILITIES	120 watts
XOVER FREQUENCY	1500 Hz
XOVER SLOPE	4TH order





THE DYNOMONITOR/2

The most difficult problem a designer faces when creating a two-way system is how to obtain solid bass output while still reproducing detailed, non-resonant midrange with an absence of lobing. Two new Dynaudio drivers have provided us with an answer. The D28/2 tweeter with its hexagonal wire and precision construction enables us to use a computer optimized x-over point of 1500Hz without fear of tweeter distortion or failure. This very low x-over allows the critical upper midrange and lower treble to be reproduced by the tweeter, greatly reducing upper vocal colorations and eliminating lobing and beaming problems due to high x-over points. The low and middle frequencies are reproduced by the 19W38, a high excursion bass-mid with sufficient surface area for realistic bass output, and its single piece cone and dust-cover design offers the best mechanical transition from voice coil to surround.

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- NEW HIGH FREQUENCY UNIT WITH INCREASED DAMPING TO REDUCE SIBILANCE
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PRICE: ASSEMBLED: $647.00 pair KIT W/CAB.: $597.00 pair KIT W/O CAB: $437.00 pair

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Reader Service #31

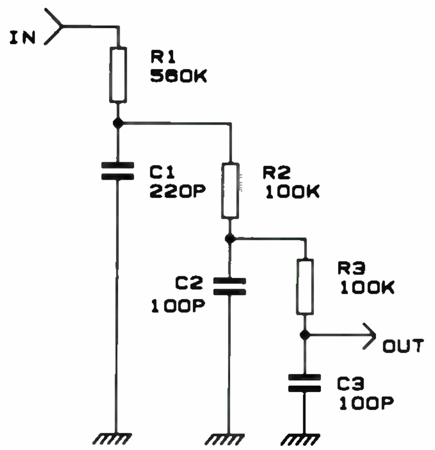

FIGURE 4: Original low-pass circuit.

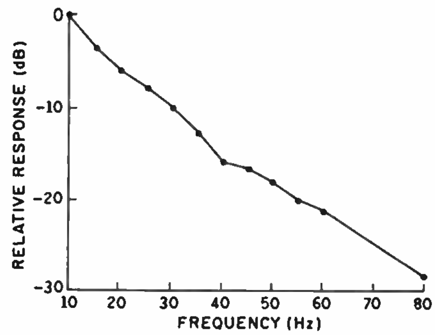

FIGURE 5: Subwoofer amplifier frequency response.

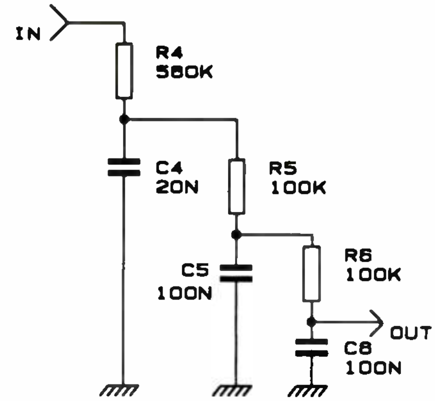

FIGURE 6: Modified low-pass circuit.

Continued from page 22

The subwoofer amplifier's schematic showed a switchable "high-cut" (low-pass) filter circuit (whose original purpose was to reduce high-frequency noise from LP scratches, tape hiss, and the like). The circuit consisted of three low-pass (first-order) RC sections (*Fig. 4*). The −3dB point of a single RC section is the frequency at which the impedance (reactance X_C) of the capacitor is equal the impedance (resistance R) of the resistor; that is, when $X_C = R$. I wondered what frequencies each of the sections was set to.

The first RC section used a 560kΩ resistor and a 220pF capacitor. Using the formula for capacitive reactance, the impedance of the capacitor equals the impedance of the resistor at about 16kHz.[6] At this frequency, therefore, the section's output is 3dB down. (The phase shift is then 45°.) Beyond twice this frequency, the response achieves the maximum rolloff rate of 6dB per octave for a first-order filter.

The second section used a 100kΩ resistor and a 100pF capacitor. Using the same type of calculation, I found its −3dB point to be at about 1.3kHz. The third section was identical to the second.

My idea was to modify this filter so it would pass only very low (subwoofer) frequencies. That way, the filter could be switched on and off. Circuit modifications could also be minor, and the chassis wouldn't need any.

I put a 20nF capacitor across the original capacitor of the first RC section and a 100nF capacitor across each capacitor of the second and third sections. (Use capacitors suitable for audio use; e.g., not disc ceramic). This gave a −3dB frequency of about 15Hz for each section.

Besides filtering out the higher frequencies, this combination provided some additional phase shift, bringing me closer to −180° at the low end. For good performance, you must have either 0° or some integral multiple of ±180° of phase shift, so to achieve an in-phase condition with the main speakers just reverse the subwoofer leads.

A plot of the amplifier's frequency response with the modified filter is shown in *Fig. 5*. Its ultimate slope—the combination of tone controls, loudness contour, and our modified low-pass circuit—is about 10dB per octave.

Higher-order filters give sharper frequency cutoffs (and greater phase shift), but also have more loss. Three ideal, cascaded first-order sections would give us a third-order filter. Its ultimate rolloff slope would consist of: three first-order sections multiplied by 6dB per first-order section per octave, yielding 18dB per octave.[7]

Our passive network doesn't behave like three separate ideal sections, because of excessive interaction between the sections. The second section's input impedance is much lower than the first's output impedance. As a result, it loads the first section excessively above 10Hz. The network then effectively acts like only two sections.

For this reason, we won't get maximum rolloff slopes (only 10 versus 18dB per octave), phase shift will be less (about 30° less lag), and the losses will be greater (10dB more) than an ideal third-order network. Use of IC amplifiers as active filters would be the optimum but more complex way of doing this.

You could also construct a filter circuit externally and connect it to the tape monitor loop of an amplifier. (Be aware of input and output impedances.) Or you could use the lowest-frequency band of an audio equalizer, boosting it fully and cutting all the rest.

Figure 6 shows the final low-pass filter circuit. With each section set for a −3dB point of 15Hz, it reaches its maximum rolloff slope by 35Hz.

Because of the inherent losses in passive networks, about the best we can do here is third-order (three first-order RC sections). The loss is 32dB at 30Hz, but amplifier gain more than makes up for it.

RELATIVE PHASE

With incorrect filtering it is possible for the subwoofer to be in phase with the main loudspeaker at, say, 30Hz and out of phase at some higher frequency. This would result in undesirable interference with, and cancellation of, the main speakers at the higher frequency.

At one point, I found reinforcement (in phase) at 30Hz between subwoofer and main loudspeaker, and cancellation (out of phase) at 48Hz. I modified the filter to give a lower cutoff frequency to make sure that the subwoofer was well attenuated by 48Hz.

To check this, face the speakers close together, as previously described, or use an audio oscillator and an inexpensive sound-pressure-level meter. Even a microphone connected to a tape recorder will do, using the VU meters.

My subwoofers are located in the far corners of the room, about 5′ farther away from the listening position than the main speakers. Using the relationship between frequency and wavelength from above shows that at 30Hz the subwoofers are one-eighth wavelength farther away from the listening position than

Continued on page 26

CAPACITOR IMPEDANCE

The impedance (reactance) of a capacitor is:

$$X_C(\Omega) = \frac{1}{2 \times \pi \times \text{frequency(Hz)} \times \text{capacitance(Farads)}}$$

where

$\pi = 3.14$

μF = microfarad = 10^{-6} farad

nF = nanofarad = 10^{-9} farad

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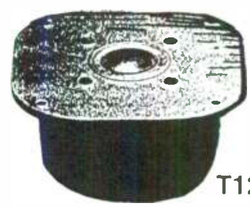
4C111

6C211S

7V513

4C212



T120ZLT

	4C111	4C212	6C211S	7V513	T120ZLT
DCR	6.16	6.78	3.10	6.83	6.0
Fs	81.72	72.80	41.37	46.81	–
Qes	0.482	0.58	0.389	0.250	–
Qms	2.851	2.33	6.114	2.91	–
Qts	0.412	0.47	0.362	0.28	–
Vas (L)	4.14	4.82	29.92	33.81	–
L (mH)	0.24	0.48	0.5	0.55	–
dB/W/m	88.56	86.8	89.2	93.25	93.5
VC length	7.2	12.4	11.5	13	–
Gap height	5	4	4	6	–
PRICE	$39	$39	$39	$79	$69

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Continued from page 24

the main speakers. This works out to 360° / 8 = 45° of additional phase lag.

For 90Hz energy, we would have a 90° phase difference between the two, which would cause extraneous localization of sounds. At 120Hz the subwoofer would be one-half wavelength away. This is equivalent to 180° of phase shift and would cause cancellation between the main and subwoofer systems at the listening position around that frequency. Therefore, we'd want to make absolutely sure the subwoofer was completely attenuated by then.

The overall phase shift is:

Phase Shift	Source of Phase Shift at 30Hz
−115°	Amplifier, tone controls, and subwoofer filter
− 45°	Subwoofer location versus main loudspeakers
+180°	Phase reversal at subwoofer connection
+20°	Total phase shift

and rises with frequency.

Figure 7 plots the *relative* phase of the main and subwoofer systems at the listening position. Up to the mid-40Hz region, they are quite close. At higher frequencies, the phase responses diverge somewhat. Since the output of the subwoofer drops rapidly with rising frequency due to the filter, this increasing phase difference is of little or no consequence. If the main loudspeaker's low-frequency response were rolled off with a subwoofer crossover, this particular problem wouldn't exist.

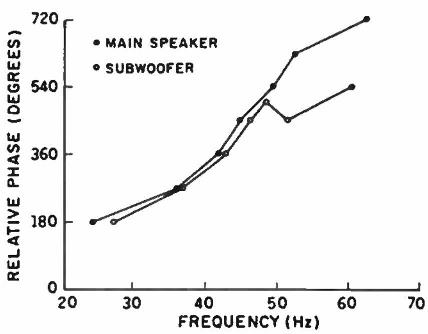

FIGURE 7: Relative phase versus frequency of main speaker and subwoofer.

LEVELS

Subwoofer filter frequency and levels are correct when the subwoofers are absolutely unnoticeable during programs without very-low-frequency content. If the subwoofer cutoff frequency or levels are too high, it will add boominess or tubbiness, and may degrade imaging. The subwoofer should be inaudible until some genuine low energy comes along, and then you'll hear it.

Clearly, getting a subwoofer system to play optimally with the main loudspeakers can be complex. You must consider not only relative amplitudes of the loudspeakers but their relative phase responses due to electrical (filtering, crossovers), electroacoustic (drivers, enclosure type), and acoustic-environment (enclosure location, room size and shape) effects. Keeping the subwoofer cutoff frequency very low lessens the problem.

WARPS

Warps are present in almost all LPs. They show up as out-of- phase, inaudible, very-low-frequency loudspeaker cone "thumps" every couple of seconds (60 / 33⅓ RPM = 1.8 seconds) in a subwoofer.

Port-vented loudspeakers (fourth-order) not specifically designed for use as subwoofers may "lose control" of cone motion at subsonic frequencies. This is because the loudspeaker becomes unloaded from this kind of enclosure at the box resonant frequency minus one-half octave. This can lead to large amounts of cone motion with highly compliant loudspeakers. So be careful, especially if you have a lot of subwoofer amplifier power available. Fuse the subwoofers.

Fusing should be no problem with a sealed acoustic suspension (second-order) or resistive-port-vented (third-order) enclosures. Some amplifiers use infrasonic rolloff to reduce this problem, which may restrict their very-low-frequency response.

SPEAKER MODS

One way to modify sealed loudspeakers used as subwoofers is to remove the fluffy fiberglass stuffing. One purpose is to damp mid-frequency resonances, but unfortunately, it also reduces the enclosure's ability to reduce the woofer's cone excursion (and therefore distortion) by decreasing its Q_{TC}. To some extent, however, this modification will degrade transient response. Make careful note of exactly how the batting is arranged, for future restoration. Leave undisturbed any material in or immediately around the vent.

You might also consider bypassing the crossover to feed the woofer directly. Be prudent about these modifications to avoid introducing any drastic alterations in impedance or frequency response, or other unwanted changes.

TRY IT!

The above caveats should not prevent you from trying this inexpensive type of subwoofer setup with whatever equipment you have, including a single-channel subwoofer. I used two subwoofers in order to preserve interchannel subwoofer stereo information.[8]

No, this simple low-pass subwoofer setup does not remove subbass energy from the main loudspeakers, so we're not getting the full benefits of such a setup. Perhaps it should be called "subwoofer augmentation." It doesn't correct for nonlinear phase versus frequency response at the low-frequency end of the main and subwoofer loudspeakers' response, so low-frequency transient reproduction may not be optimum.

But the price is right, the experiment is fun, and the results are gratifying. And what better way to evaluate whether you should take steps toward a more sophisticated subwoofer system?

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THE LINEAR-ARRAY SOUND SYSTEM

By Philip Witham

This article is a call for audio constructors to help try out a project that could be very interesting—a new system for reproducing sound that would, in theory, provide vastly improved imaging over stereo. To my knowledge, the only thing comparable in imaging accuracy and impact is binaural recording. The linear-array system doesn't require headphones, however, and would not (within generous limits) be affected by listening position. It should allow a small group of listeners to hear excellent sound simultaneously, and all else being equal, produce the best sound reproduction ever heard from speakers.

Extravagant claims along these lines have been made before, but bear with me. I'm not talking about a small or arguable subjective improvement, but a measurable, clear-as-day, box-you-in-the ears leap in sound imaging and the overall sense of realism and presence in reproduced sound.

This system should be able to accurately reproduce the sound field that occurred in the general area of the microphone, in two dimensions; that is, if the sound pressures are visualized from above, and the vertical dimension is ignored (by taking a flat-plane slice of the sound pressures at a certain height above the floor), a "field" of pressures exists at any given moment in the front-to-back and left-to-right dimensions. The system does not depend on any psychoacoustic effects or mimicking of the response of the head/ear shape to sounds coming from different directions. It would actually *reproduce* a 2-D sound field, which you would be free to walk around in and hear from different perspectives.

THE HITCH

There's one hitch: complexity. The imaging limitations of our current sound systems are (in my opinion) not really the fault of our equipment. We can routinely deliver a voltage to the terminals of a speaker that is an honest reproduction of the instantaneous sound pressure present at the microphone when the recording was made. And if we spend enough time and money on speakers and the listening room, they'll do a pretty good job of reproducing that instantaneous sound pressure—at least close to the speaker. However, there is simply not enough recorded information to represent a whole sound field. We need more channels—preferably lots more.

Considering that we have only two ears, why would we need more than two channels? A valid point—if you don't mind wearing headphones, binaural recordings can be wonderful. Or, through a digital signal-processing tour de force, we could manage binaural sound with speakers, canceling the L-R and R-L crosstalk, speaker errors, and room acoustics.

If you are willing to stand perfectly still, that is. If the said DSP system knew where your ears were in the room (with about 0.2″ accuracy), you could even move about. When

ABOUT THE AUTHOR

Philip Witham says: "During this reincarnation I am an electronic-hardware-and-software-design contractor living in Vista, CA. I've been into speaker design for 15 years and built an optically servoed subwoofer just out of high school, plus a number of other speakers and audio gizmos. For the past nine years, on and off, I've designed, built, and operated the instrumentation for Robert Truax's commercial rocket research company, Truax Engineering. Now I'm returning to my roots in audio. In previous incarnations I have been variously a cat, a tree, and an anaerobic stomach bacteria. But that's as far back as I can remember."

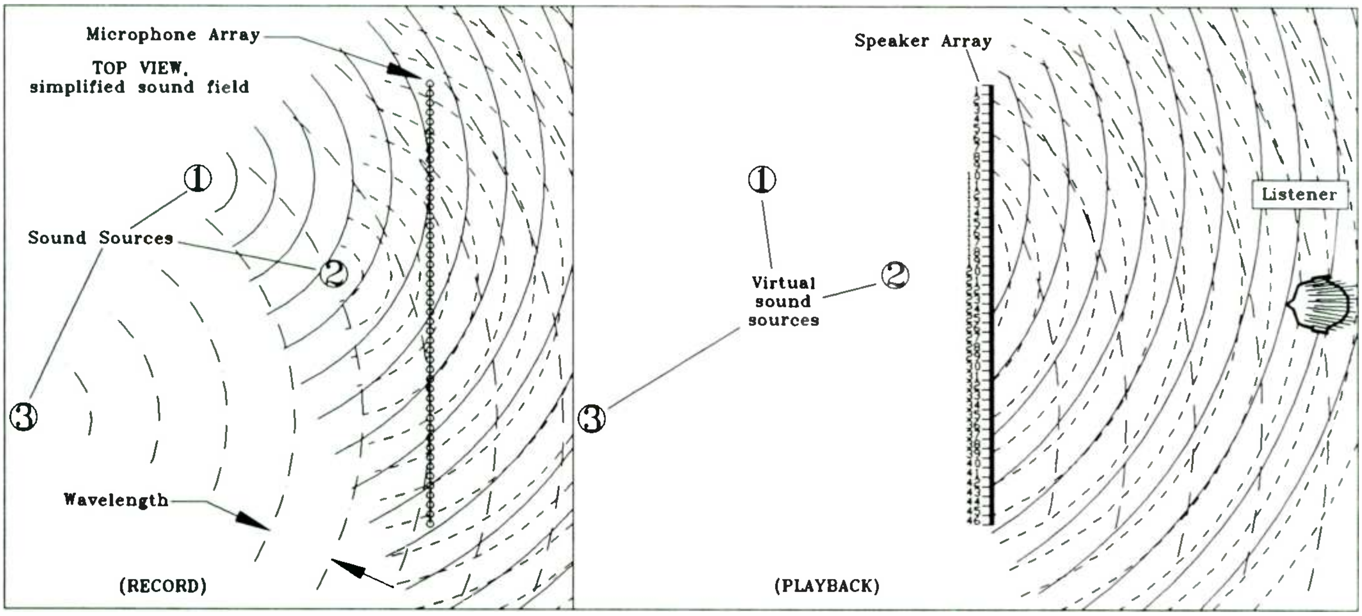

FIGURE 1: Imaging example.

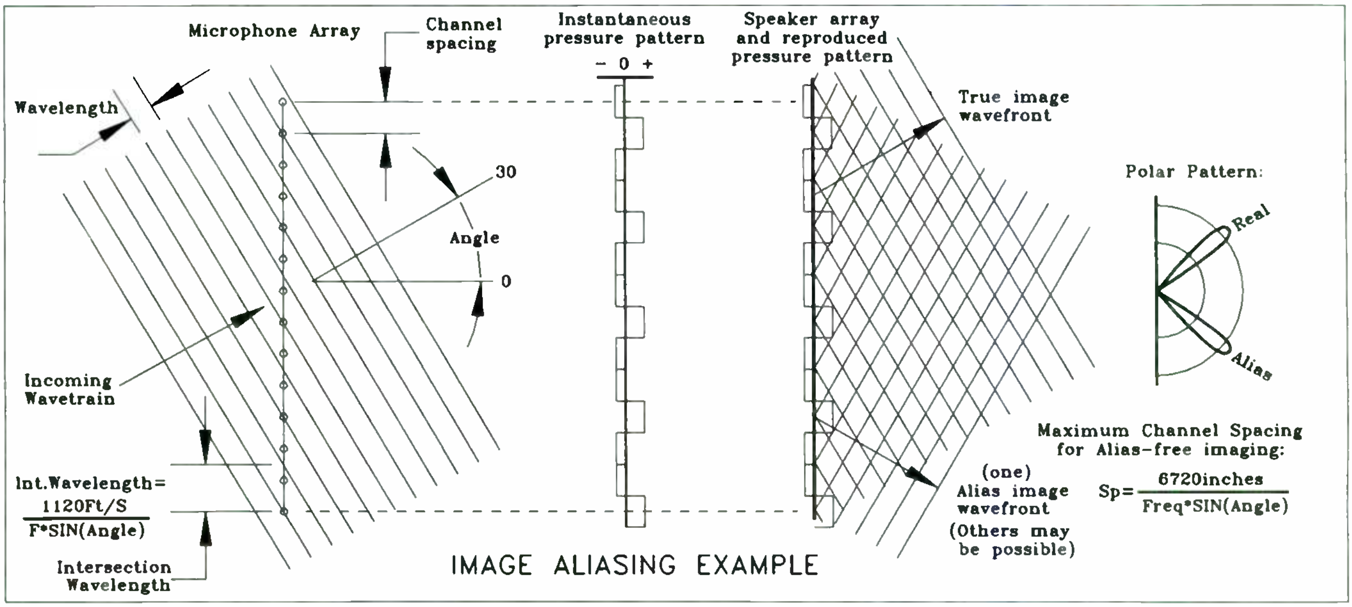

FIGURE 2: Image aliasing and formulas.

you turned your head, however, the reproduced sounds would turn with you. And the effect would only work for one listener at a time. Not very convincing. If you want listening-position-independent loudspeaker playback that sounds like live performance, there is no known practical alternative to recording more channels.

Of the several systems for improving imaging that have emerged over the years, quadraphonic and six-track movie sound are the only ones that ever "made it big." The ambisonics proposal by M. Gerzon, et al., looks better.[1] A concept similar to the linear array has been described for professional sound reinforcement.[2] Neither can do the job I want to perform here. I haven't run into the concept I describe below, and would like to know if anyone has considered it.

THE LINEAR-ARRAY SYSTEM

Using brute (channel) force, a system of perhaps 16 to 100 channels from a horizontal line of closely spaced microphones—spread about 5′–8′ across (*Fig. 1*) and feeding the same number of identically spaced speaker channels—could produce superb 2-D imaging. Channel spacing would be 1″–3″.

At this point, you're probably imagining 50 studio mikes, 100 speaker drivers, and 25 stereo power amps. Let's see—at about $25,000 for the mikes, $12,000 for the amps, and $20,000 for the speakers, plus assorted preamps, a huge multitrack recorder, three engineers, a laboratory, and six technicians, that's about half a million dollars per system, per year. Actually, the prototype implementation described below could be built by a group of amateurs (or one amateur with enough time and money). And it is not 25 times as complex as a stereo system.

For initial testing, the prototype speaker design is a single flat electrostatic diaphragm with an integrated amplifier, about 6′ wide × 3′ high, raised 3′ from the floor (*Fig. 5*). The microphone is a single 6-foot bar composed of small, inexpensive mike elements (*Fig. 4*), with an integrated preamp. A long multiconductor cable connects the mike and speaker.

No recorder is needed initially. The speaker could be set up in a listening room adjacent to a live band or orchestra (your buddy Jim and his kazoo), or the mike could be set up outdoors, or in another room.

HOW IT WORKS

How a linear array can form a complex, accurate 2-D image is very similar to how phased-array radar or phased-array sonar forms a search beam, or a hologram forms a 3-D image (*Fig. 1*). The sound waves from any individual source arrive at all of the microphones eventually, but separately in time, depending on the distance from that mike to the source.

When each channel is reproduced at the speaker, the sound pressure generated there joins all of the other channel's signals in the same relationship they had at the microphone. The resulting set of waves will strongly add up to a single wavefront traveling in the same direction as the original waves, and also with the same apparent source distance (curvature, as seen from above).

HOW IT SOUNDS

The audible effect is like a window in space, looking into the place where the microphone was located. Imagine that an orchestra was recorded from a mike position on the level of the musicians, at approximately front row. Imagine a wall built across the concert hall at front row, with your listening room built up against it. A rectangular hole of the same width as the speaker is cut into the wall, and only through this hole do you hear the music and see the orchestra.

As you walk around your listening room, the orchestra remains fixed in place, but your perspective on it changes. The sound level does not increase rapidly as you approach the hole (speaker), but the acoustic image gets much wider, just as the visual image does; you hear the widest panorama of sound up close to the window. If you stood off to the side, the direct sound from the orchestra would be largely reduced, with room reflections dominating. As you walked back to the center, individual instruments would come into (acoustic) "view" one by one. The relative directions between instruments would change with your perspective. Each individual sound source has its own dispersion pattern. Sounds would not seem to come from the speaker itself.

Like stereo, this system has no true vertical imaging. Also, if the spacing between channels is not close enough for a particular frequency being reproduced, an effect I call "image aliasing" will occur. This is similar to what happens if you digitally sample an audio signal at too low a sampling frequency (*Fig. 2*). The reproduced sound energy will be split between two directions or more (in digital sampling, two or more frequencies are produced). To a listener, several sound sources would appear, or the image (apparent direction of the sound source) would simply become less clear. Since the speaker produces the same total sound power regardless of how that power is dispersed, aliasing will also be accompanied by a frequency-response aberration.

The proximity (spacing) of channels needed to prevent image aliasing is related to the wavelength of sound at the particular frequency of interest. It also depends on the incoming angle of the wavefront. Using the second formula in *Fig. 2*, we can pick a maximum frequency and angle for "ideal" imaging, and calculate the required maximum channel spacing to achieve it.

The basis of this formula is the Nyquist sampling theory, which says basically that you will need at least two samples for every cycle of the signal. In this case, the "signal" is a slice of the sound-wave train that crosses the line of the microphone. Consider a single sine-wave sound coming from far off, at some angle. At any given instant, oddly enough, the wave train passing the microphone line produces a sine wave pattern of pressures along the line. The wavelength of this pattern is not the same as the incoming sound, but a multiple of the sound wavelength depending on the angle of incidence.

If the sound is coming in along the line of the mike ("90°"), the wavelength of the pressure pattern along the mike is the same as the sound, or 1,120 ft/s divided by the frequency. If the sound comes in at 45°, the slice pattern's wavelength is 1.414 times the sound wavelength, at 30°, 2 times, and so on. Head on to the mike, the pattern wavelength is infinite.

In other words, a single channel can reproduce a straight plane wave. Flat electrostatic loudspeakers, in fact, produce a wavefront that launches straight out of the speaker with little dispersion (if the ESL is sufficiently wide); which is a problem for many people listening to flat ESLs. The problem will disappear if a linear array is divided into small enough channels. (Note also the design of Quad's ESLs, using concentric-ring electrodes driven by a delay line to produce good dispersion.[3])

CHANNEL SPACING

I've chosen a channel spacing of about 1⅛″ for a prototype design in order to produce superb imaging of all frequencies up to 12kHz, at up to a 30° angle to either side. This means that the width of the soundstage can cover 60° at your listening position, for sounds recorded a fair distance from the mike, and the imaging will be excellent at up to this frequency. Frequencies of 6kHz or below could come in at any angle and be reproduced properly; 20kHz sounds must come from 16° to either side or less. Still not bad. From observations of typical stereo speaker spacings, listening distances, and a sense of the minimum width desired for a "real sound stage" presence, I chose a 6′ speaker width, which divided by 1⅛″ gives us 64 channels.

For typical musical sounds, the channels could likely be spaced much wider than 1⅛″, since music is not composed of pure sine waves. Most music contains little high frequency, and our ears are imperfect—their ability to locate a sound source begins to level off somewhere in the upper midrange. (A perfect set of ears would become more precise in locating ability with decreasing wavelength.)

Several techniques could considerably reduce the number of channels required with no serious reduction in quality. And the alternative is merely stereo. All of this makes the job much easier, requiring fewer channels, perhaps 24 to 48. But for the sake of experiment, it is much easier to mix adjacent channels together (effectively reducing the number of channels) than to throw the system out and start again if we need more channels. Better, also, to set a high standard as a reference in order to properly judge just how cheap a system we can get away with.

MIRROR IMAGE

Another way the linear-array system "fails" is the way sound coming from behind the microphone is reproduced. It would be imaged just as faithfully as the front-facing sounds, but will be "mirrored," sounding as though it was coming from in front (behind the speaker). In most situations this means the recording hall (studio, stadium) reflections and audience sounds would be recorded along with the music. It's the same story with stereo, but these sounds would be imaged much "better" with the linear array.

The solution would be to use cardioid mike capsules or other directional types. Moreover, since the proposed test speaker is a dipolar type, the sound produced from the rear of the speaker is just as strong as from the front, and would be a faithful mirror image of what is heard from the front. (From the back of the speaker, it is the sound from in front of the microphone that is flipped around, and the sound coming from behind the mike is imaged correctly.)

Why use a linear array rather than some other configuration? Because we are gravity-bound creatures, and most of the sounds we listen to occur over a more-or-less flat area. Three-D imaging is possible with a similar system, using a rectangular array of channels, but that would require a multiple of the already intimidating number of channels. A linear array would get us more bang for the buck. Also, this system does not preclude the addition of several rear or surround channels.

Figure 3 shows how the ESL will disperse most of the sound in the vertical plane. If you kneel below the bottom of the speaker, or stand above the top, the sound level suddenly drops. This well-known characteristic of planar speakers is actually a very desirable trait, and reduces the effect of room acoustics. The vertical position on the speaker from which the sound seems to originate is directly in

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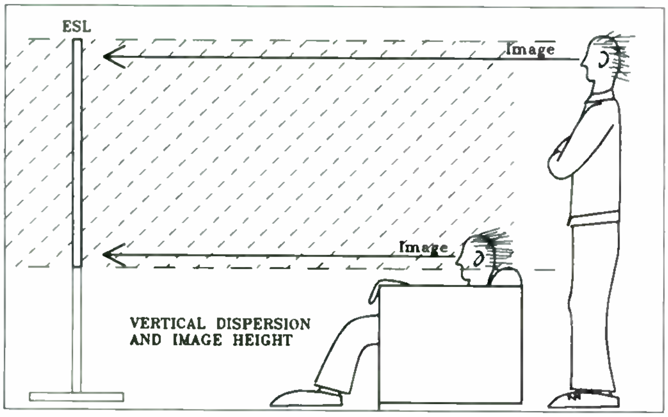

FIGURE 3: Vertical dispersion and listeners, apparent sound-source plane.

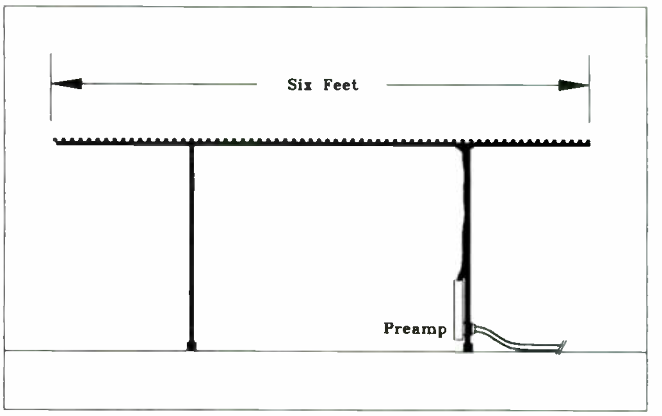

FIGURE 4: Microphone.

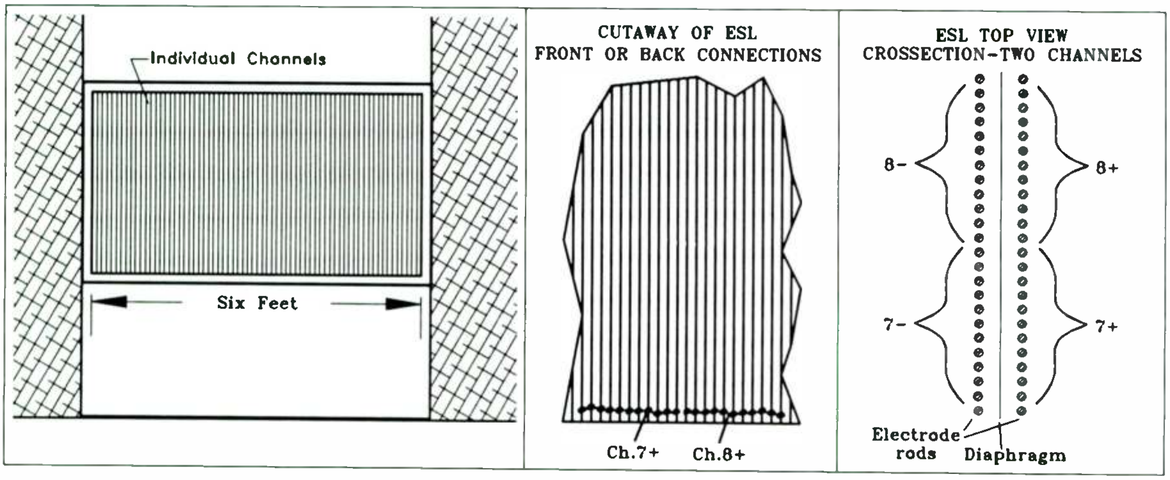

FIGURE 5: Electrostat detail.

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front of your ears. Or, if you visualize your face's reflection in the speaker diaphragm, the sound originates there.

In the case of a linear-array system, the sound will appear to originate from a flat plane on the level of your ears, as deep as the recorded hall. The speaker diaphragm should be tall enough to cover the range from below the lowest seated listener's ears to above the tallest standing listener's ears. By my measurements, this is from 30″ to 73″ above the floor, or at least 43″ of diaphragm height raised 30″ from the floor.

PROTOTYPE DESIGN

To allow us to get it up and running at a lower cost, the electronics would initially be built as 16- or 32-channel modules. The speaker would have all 64 channels of connections. The speaker is a single flat electrostatic diaphragm about 3′ × 6′ (*Fig. 5*), following the designs of Roger Sanders and Barry Waldron.[4,5] (This magazine's classified section gives the address of Barry Waldron's ESL users' group, under "Clubs.")

In a complete system, the bass must be actively crossed over at about 400Hz to a horizontal line of four to eight dynamic woofers. But for initial testing, we may omit the woofer system. 64 stator electrode grid pairs are spaced every 1 1/8″. Since the stators in this type of electrostat

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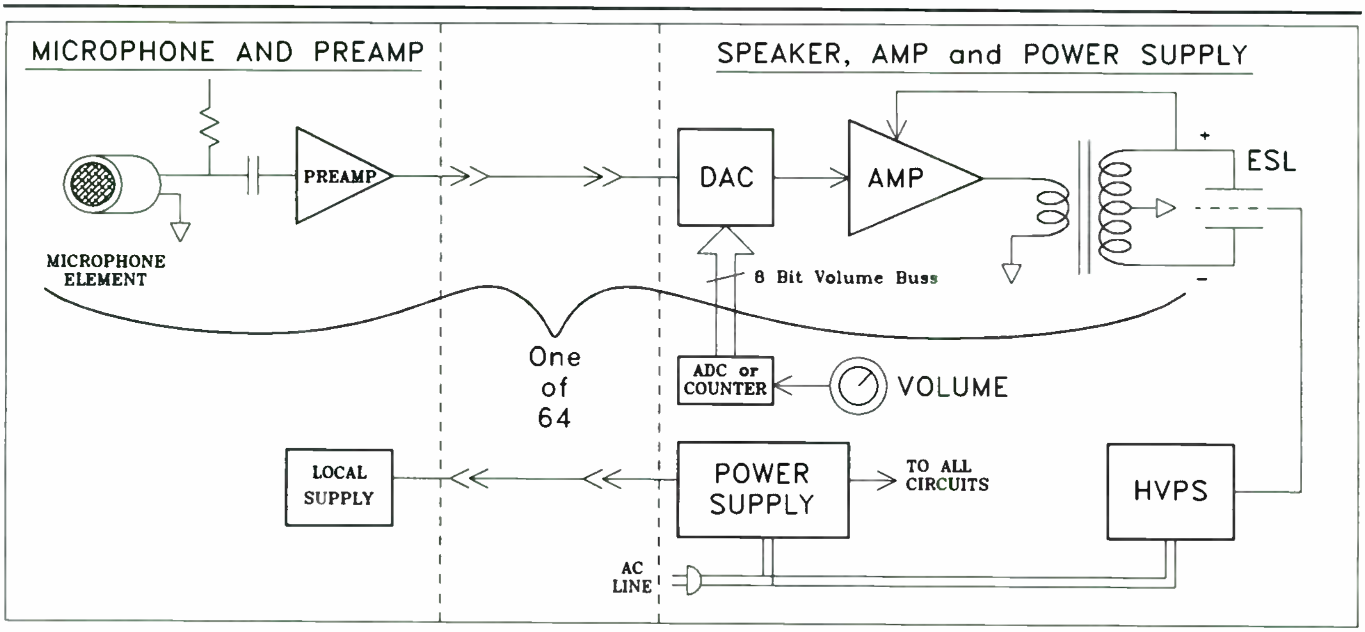

FIGURE 6: Electronics block diagram.

Continued from page 32

are composed of vertical 1⁄16″ welding rod (or music wire) sets with air gaps separating them, it should be simple to connect each small 1⅛″ group of rods as a separate channel. Only one high-voltage power supply would be needed to charge the single diaphragm. No individual channels connect to the diaphragm.

As long as the spacing between channel centers is precise, the panel can be physically divided into some number of units. The areas below and above the ESL should be covered floor to ceiling with solid panels. The panels aid in extending the low-end response of the speaker by effectively giving it an infinite vertical height, forcing the back wave to go around the sides. This ESL size should have flat response and high output down to about 200Hz. Other types of planar drivers (Magnepan, ribbon, and so forth), and even dynamic drivers could be used, but these would not perform as well as an ESL, or be as simple.

The microphone (*Fig. 4*) would use 64 inexpensive electret mike capacitor elements mounted on a 6-foot-long bar. Preferably, pressure-zone microphone capsules should be mounted on a single long plate, but omnidirectional capsules would also work. A custom 64-channel mike preamp and power supply would be incorporated with the mike locally to eliminate the need for separate microphone cables. Each channel would use one differential amp IC and a few small parts. The preamp output would be 64 high-level signals on a single multipin connector, and the mike cable need be only a standard multiple-twisted-pair, shielded cable under 1″ in diameter.

Since we are driving 64 very small electrostatic speakers, so to speak, the capacitive load on each channel is also small, perhaps 100pF. For several reasons, the amplifier would be located directly at the base of the ESL, and short high-voltage wires would connect the amp outputs to the speaker. Each channel of amp would require one amp IC (or two output transistors and one quarter of a quad op-amp chip), plus miscellaneous small parts (*Fig. 6*). Each amp would drive a small output transformer, and the amp feedback loop would sense the transformer's output for best performance. One power supply would feed the whole amp bank.

Connection of the input signal to the amp would be by the same multipin style connector used for the mike preamp output. Gain control should be provided at the input of each amp channel, with one common control signal. A cheap 8-bit multiplying D-to-A converter in each channel will accomplish this, with the volume-control signal being a digital number input supplied in common to all channels. Each channel's analog signal goes in the reference input pin of a D/A IC, and comes out the analog out pin(s). A single digital encoder pot would supply the volume-control number.

The result is linear, accurate, cheap, and simple. And no, this does not involve digital sampling, since the audio signals go through an entirely analog path. The whole amp would probably be divided into four identical 16-channel PCBs and a separate power-supply box.

RECORDING ALL THOSE CHANNELS

If the linear array's performance in actual testing lives up to its promise, it will need a reasonably priced recording system. Phase differences among the separate tracks make multitrack analog recorders (such as those used in studios) probably useless. As a temporary (expensive!) demo method, multitrack digital studio recorders could do the trick. Sixty-four channels of audio can be recorded on a modified consumer video recorder as 65 separate FM channels (with the extra channel being an overall compression control track). This technique is probably too expensive for anything but rocket instrumentation and other government-funded projects, however. The rest of us will just have to wait.

Digital consumer video recorders (to be introduced next year) have internally a data rate of about 20 million bits/sec. Thanks to several handy relations among all the separate channels, the linear-array signals can be sampled and compressed to 20 Mb/s without too much complexity, yielding CD dynamic range or better, and similar quality overall.

THE MARKETPLACE?

Will a market exist, and when? If the linear-array system sounds ridiculously expensive, consider the development cost of radio, TV, and hi-fi; not to mention the personal computer, the VCR, the Camcorder, which are just as complex. If the linear-array system is more desirable than present systems, the cost will come down with mass interest. How well the system works can be determined only by trying it.

The system would be backward and forward compatible with stereo; that is, the linear-array speaker could reproduce stereo, and linear-array recordings would sound fine on a stereo system. Playing stereo on the linear array is accomplished by using DSP or an analog (LC) delay line to delay successive speaker channels up to a few milliseconds. This allows you to generate a virtual sound-source location and shape by using the correct set of delays—a left and right pair of columns located a few feet behind the speaker panel, for instance, for near perfect dispersion. Particular dispersion patterns could also be generated to suit the room, type of music, number of people listening, and so on.

One interesting "dispersion pattern" is to focus the output on a L/R pair of vertical lines located at the listener's ears. An SPL much higher than present at the speaker panel would be generated, and the room response would be proportionally reduced, a sort of headphoneless headphone (but boy, would it sound weird if you moved!). A flat speaker of this type could have video projected onto it, and the combination might be knockout.

LET'S DO IT

Since in all things the great god Murphy reigns supreme, I would prefer to be reporting the results of actual testing. But after sitting on the idea for years, I've decided it would be better to see it implemented sooner with the help of other audiophiles.

Therefore I am proposing a collaborative effort among *SB* and *TAA* readers and myself. I can supply the design, especially of the circuitry, PCBs, and overall system, and can do some of the construction. We especially need an experienced ESL builder to handle the detailed speaker design and construction. The major modules of the system are: microphone, mike preamp, power amplifier, and ESL. Each might be subdivided further or be joint projects of local groups. The whole mess should come together in the end somewhere within driving range of southern California.

Anyone game for some High Audio Adventure? I'd be happy to field questions and welcome your comments, especially to discover any related previous work and any well-grounded technical criticisms and potential bugs.

REFERENCES

1. M. Gerzon, et al., "Ambisonics," JAES (Nov. 1985): 859.
2. A. Berkhout, "A Holographic Approach to Acoustic Control," JAES (Dec. 1988): 977.
3. R. Williamson, "The Quad 63," *SB* (1/82): 16.
4. R. Sanders, "An Electrostatic Speaker System," Part I, *SB* (2/80): 20.
5. R. Sanders, "The Sanders Electrostatic Speaker," *TAA* (4/75): 18.

A REVISED TWO-WAY MINIMONITOR

By Ralph Gonzalez

Several years ago, using the Dynaudio 17W75 woofer and D28af tweeter, I built what I considered a no-holds-barred minimonitor. The difficulty of finding a crossover to handle the transition between these drivers led me to develop the Loudspeaker Modelling Program (LMP) for computers, as I described in "An Introduction to Frequency Response and LMP" (*SB* 1/87, p. 18; *SB* 2/87, p. 42). In the third installment, "LMP: A Construction Example" (*SB* 3/87, p. 38), I presented the final speaker design.

Popular at the time, particularly in this pairing, the Dynaudio drivers still have merit. The 7" woofer has a rigid "sandwich" hemispherical cone that acts like a piston up to its (rather sudden) cutoff at 3kHz; the cone of most comparable drivers begins breaking up by 1kHz. The woofer also has high thermal-power handling, though it is a bit underdamped and has limited low-frequency linear excursion capability (X_{MAX}). The 1" soft-dome tweeter is still unsurpassed for sheer power handling, and has well-controlled on- and off-axis responses.

I had intended to use a first-order crossover for the theoretical advantage of minimum-phase response, but ultimately concluded that real-world driver limitations made this impossible. Assuming its frequency response is nearly flat, a minimum-phase loudspeaker will have near-zero phase shift in its frequency band. Because of the difficulties and compromises required to achieve this, plus the lack of conclusive evidence of any audible benefit, few such speakers exist. Instead, I used a second-order crossover at a fairly low 1.5kHz, to avoid the woofer's 3kHz jaggedness.

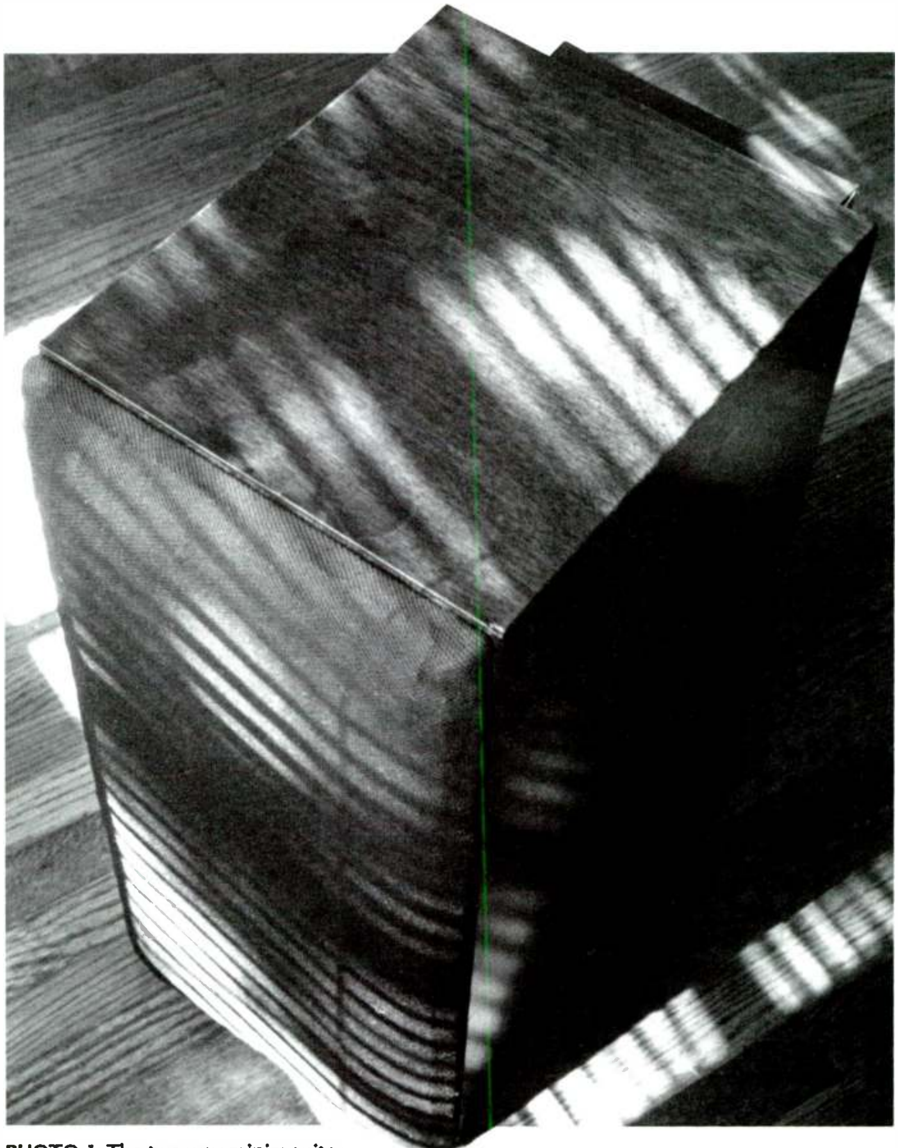

PHOTO 1: The two-way minimonitor.

The enclosure used ¾" veneered particleboard, to which I added a *lead lining*. The absence of enclosure and driver resonances helped the speakers "disappear," producing sharp imaging and good soundstaging. The special driver characteristics also contributed to a dynamic and uncolored sound.

SPEAKER THEMES

I had planned to add a midrange to allow the driver latitude for a minimum-phase design, but never got around to it. Instead, I built several new speakers along "thematic" lines, trying to follow a theme to its logical limits rather than optimizing all areas of performance. The goal of the Delac S10 (*SB* 3/91, p. 32), for example, was to obtain a musical, high-resolution performance in relatively small listening rooms. I decided to bring the Dynaudio speakers up to date and to focus on a theme.

I once had the pleasure of listening at length to a pair of Celestion SL600 speakers. This is a fairly conventional box speaker:

ABOUT THE AUTHOR

Ralph Gonzalez lives in Wilmington, Delaware with his wife and son. He is a computer science professor at Rutgers University and has a PhD in systems engineering from the University of Pennsylvania, and a BA in Math from the University of Delaware. Mr. Gonzalez enjoys listening to music, building speakers, playing fretless bass guitar, and writing audio-related computer software.

6½" woofer, 1" tweeter, non-minimum-phase crossover. What sets it apart is (1) drivers designed using high-tech analysis to control resonances, and (2) an extremely rigid (and expensive) enclosure built with aerospace technology. Recalling the remarkable clarity and "openness" of these speakers, I chose as my theme a mini-monitor- like response (not necessarily reaching the frequency extremes) with pistonic, nonresonant behavior over the entire frequency range.

A FEW CHANGES

Because of limited baffle space and the frustrations common to most three-way designs, I stayed with the two-way approach. I've begun to avoid soft-dome tweeters, because their domes are breaking up over most of their frequency range. (The dome breakup is exploited to achieve extended high-frequency response.) In any case, the Dynaudio tweeter has always sounded a little "sibilant" to me, and other speaker builders have noted this quality as well.

I substituted a metal-dome tweeter. Few can be crossed over at 1.5kHz, however, and few maintain a smooth response over their entire range. The aluminum-magnesium Elac 25DT31 can do both, and has garnered praise for its role in designs by TDL, Acoustic Energy, Monitor Audio (who co-developed it), and others. A relatively heavy, insensitive dome (not unlike that in the Celestion SL600), the Elac 25DT31 is a few years old and is hard to find in the US. My friend Paul Eppes finally located a distributor in Saginaw, MI, The Court Street Listening Room.

I've tested this tweeter using a second-order 1.5kHz crossover, powered with a 70W amp. I drove the amp heavily into clipping (which produces a great deal of high-frequency energy) without any problems. Since the enclosure walls were already inert and the Dynaudio woofer is free of resonances in its passband, all that remained to fulfill my theme was to update the crossover.

PLANS

I used nonstandard second-order high- and low-pass sections in the crossover (*Fig. 1*), centered at about 1.5kHz. The LMP model suggests that in combination with the drivers the result approximates a fourth-order acoustic crossover. (Therefore, the tweeter's polarity is not reversed.) The attenuating resistors in the tweeter crossover reduce its sensitivity by 2dB. The tweeter is a fairly constant 8Ω load so you can substitute an 8Ω L-pad to allow adjustment.

The enclosure remains as in the *SB* 3/87 article. After accounting for foam lining to reduce internal standing waves, the enclosure volume is about 12 liters. The Dynaudio woofer is underdamped in this enclosure, so I originally experimented with the aperiodic loading principle. While this reduces the magnitude of the impedance peak at the woofer's resonance, it doesn't seem to have much effect on its acoustical output. Therefore, I decided to stay with the easier-to-predict sealed box, packing long-fiber wool behind the woofer to bring Q_{CB} down to about 1.0. The –3dB point is around 50Hz.

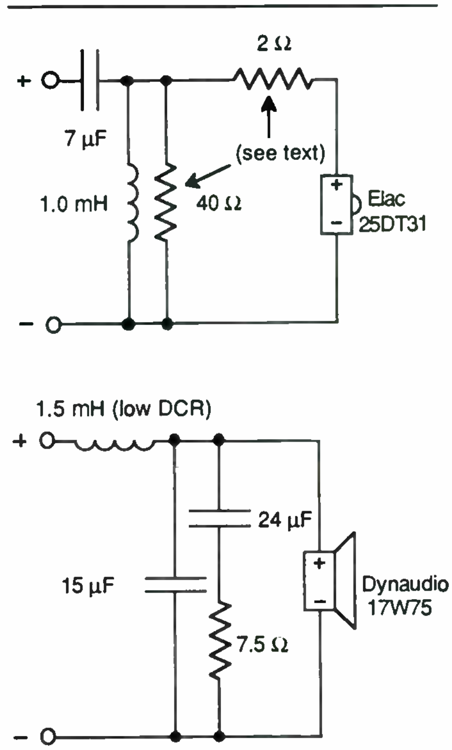

FIGURE 1: Crossover plans.

Figure 2 shows the dimensions I used (they aren't critical). Of note is the ¾" step on the baffle where the woofer is mounted to eliminate interdriver time delay. Place ¾" foam on the baffle to avoid reflections from the step, leaving a 3-inch-high × 4-inch-wide cutout for the tweeter. An alternative is to angle the front panel back or use a conventional box tilted back a little.

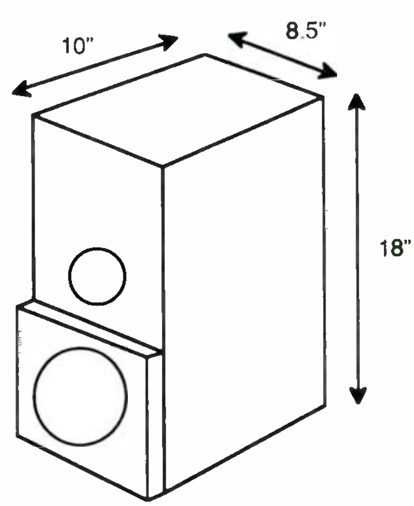

FIGURE 2: Outside dimensions of enclosure.

In truth, eliminating interdriver time delay (aligning the zero-delay planes) is generally unnecessary unless you're trying to achieve minimum-phase response. Many conventional crossovers can be made to produce a flat frequency response in spite of moderate amounts of interdriver time delay. However, I designed this crossover with the assumption that the baffle step is used, and I can't guarantee the results if you omit it.

The *SB* 3/87 article makes a few additional suggestions for treating the enclosure, particularly the use of a felt/lead sandwich on the inside walls. Use your imagination to eliminate enclosure colorations as best as possible. I believe this is the key to good imaging, since your ear no longer detects the enclosure as a sound source.

MEASUREMENTS

My friend Alan Nettleton took some time out from watching his new son to help measure the results with his MLSSA system. *Figure 3* shows the results, using the 2dB tweeter attenuation given in *Fig. 1*. The hump below 1kHz is partly due to a 2dB rise in the microphone's response. The rolloff below this is produced by the limited "window" size allowable when making FFT measurements in a small room, and should be disregarded. The peak at 25kHz, inaudible to humans, is also due partly to the microphone.

A mild exaggeration remains in the 3–6kHz range, but otherwise the on-axis curve is well balanced and extended. The 20° off-axis curve agrees closely with the on-axis curve, indicating excellent dispersion. The best listening axis probably lies midway between the 0° and 20° curves.

Above- and below-axis measurements (not shown) indicated that lobing is symmetrical in the vertical plane, as expected given (approximately) fourth-order acoustical slopes. Listening 20° below axis produces a broad dip centered at the crossover frequency. The dip produced when listening above axis (which is more likely to occur) is just as deep but fortunately affects a much narrower band of frequencies.

LISTENING

I always enjoy reading an author's comments, even though I know they are hopelessly subjective. So here goes.

A fellow audiophile, Fred Masterson, joined me recently to listen to the speakers. (Fred, a psychology professor at the University of Delaware, brings a healthy skep-

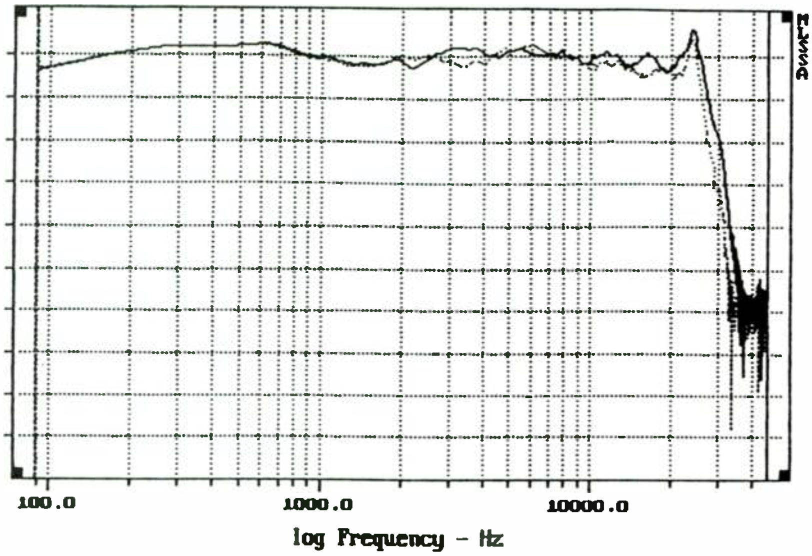

FIGURE 3: MLSSA on-axis (solid) and 20° (dotted) anechoic response, 10dB/div. (see text).

ticism to the subjective review process). Our listening was informal, rather than "under-the-microscope." We placed the speakers on stands to bring them to ear level, and powered them by my customized Dynaco MK IV tube amps.

I had a pair of similar-sized speakers using less exotic drivers. In comparison, the Dynaudio/Elac system had a superior sense of ease and refinement, avoiding hardness or congestion during crescendos. The bass response made the other speakers sound slightly "muddy," suggesting that good thermal-power handling and a dynamic nature (which Dynaudio drivers have in abundance) are more important to bass clarity and "tightness" than good bass damping (which the Dynaudio 17W75 lacks).

The treble was impeccable: extended, detailed, and smooth, never sounding overemphasized or harsh. In the midrange, voices were reproduced naturally. Overall the tonal balance was even and natural—solo piano was very realistic.

Image specificity and soundstage reproduction have always been strong points of this design, as a result of the nonresonant enclosure. The listener has no sense of sound coming from the boxes; rather, a window stretches between the speakers, into which the listener peers. In this sense the speakers are monitorlike: they put the orchestra in front of you instead of surrounding you with it. (My small, cluttered listening room may have been partly responsible for this characteristic of the sound.)

Fred and I wrote our respective assessments independently, and I offer his comments below. (I should note that Fred normally prefers panel speakers for their lack of box colorations.)

SOURCE

The Court Street Listening Room
1305 Court St., Saginaw, MI 48602-4170
(517) 792-3816

FRED SAYS

"The speakers are extremely neutral. The woofer blends seamlessly into the tweeter. The midrange and highs are clear and undistorted—for example, brass instruments produce just the right amount of 'bite' without sounding strident. Bass response was impressive for a speaker this small. I did not have the subjective impression that bass was lacking (though side by side comparison with larger, full-range speakers or live orchestras would reveal that it was). The bass seemed ample and was well controlled.

"Imaging was hampered by the small size of the listening room. Even so, the speakers threw a realistic soundstage. Some systems make music sound as if it occurs in the listening room, while others create the impression that one has been transported to the acoustical space of the original performance. These speakers did the latter."

ALTERNATIVES

If you have a Dyna 17W75/D28af system, should you upgrade it? That depends. If you're happy with the sound there's obviously no reason to upgrade—but did that ever stop you before? Perhaps, like me, you like the "theme" of speakers that act like pistons at all frequencies.

Continued on page 72

EXPLORING THE BUF 124 WITH PSPICE®

By Dick Campbell
Contributing Editor

The article "Modular Active Crossovers" (*SB* 1/94, p. 20), by Erno Borbely and Jean-Claude Gaertner, describes a general-purpose, highly adaptable equalizer that takes good advantage of the Sallen-Key circuit topology. The buffer amplifier design, called BUF 124 in kit form by Welborne Labs, is particularly interesting because it is optimized for this application and has excellent performance characteristics.

The BUF 124 is a good example of the care and feeding of very high quality audio circuits using discrete semiconductors where it would otherwise be extremely difficult to accomplish something similar using combinations of op amps. I put this circuit into PSpice®, and would like to share some of my observations.

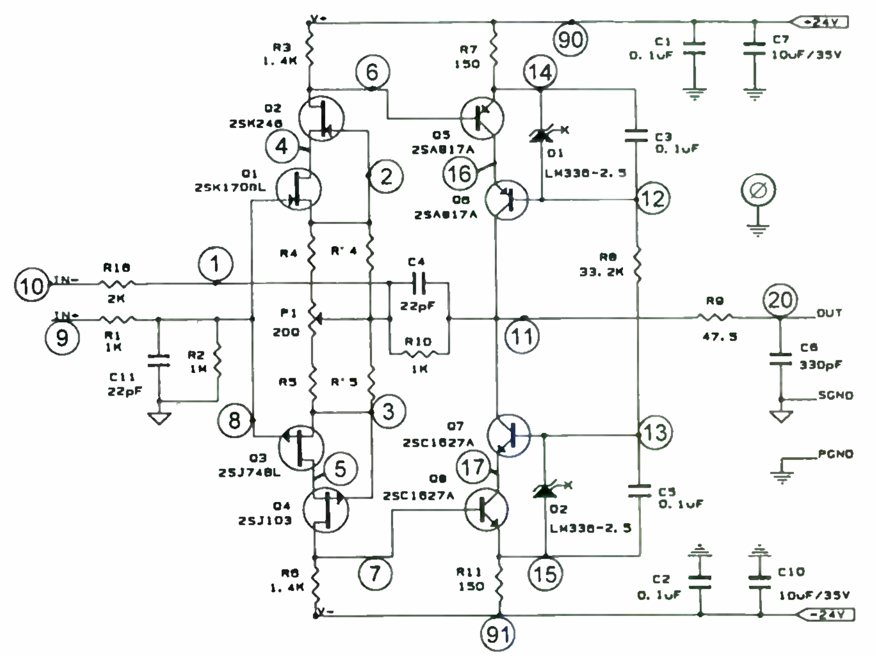

FIGURE 1: BUF 124 circuit diagram with numbered PSpice nodes in circles. In SPICE node "0" is always ground.

WHAT IS PSpice?

Anyone in the electrical engineering profession is familiar with the circuit simulation program SPICE (Simulation Program with Integrated Circuit Emphasis). This program was originally written at Berkeley in FORTRAN and credited to L.W. Nagel (May 1975) at the University of California Berkeley; an improved version called 2G has been widely distributed in the UNIX public domain. I still have my DECUS nine-track tape of SPICE 2.G for the VAX, which I carried from consulting job to job when I needed it and couldn't find it on a client's computer. It is an immensely powerful circuit analysis program.

In January 1984 the MicroSim Corporation placed a PC version on the market called "PSpice," and shortly thereafter released a "student's version." This restricted version,

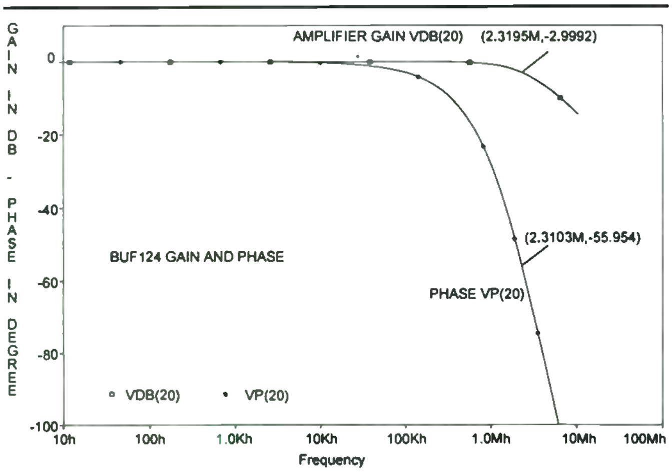

FIGURE 2: Gain (dB) and phase (degrees) of the BUF 124, input to output. The pointer marks the –3dB point at 2.32MHz. The phase shift is about 56° at this frequency.

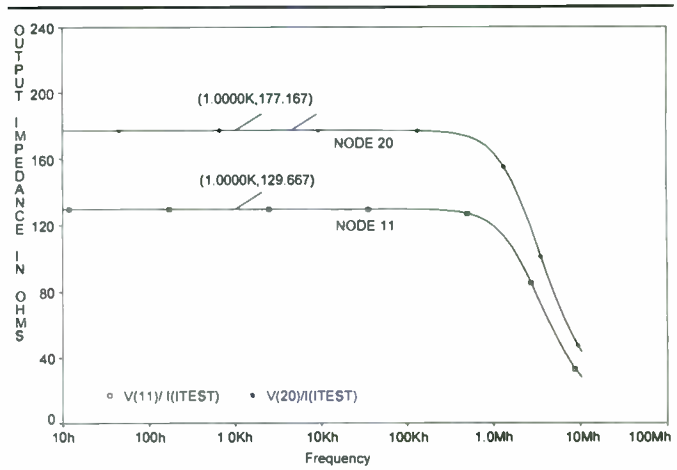

FIGURE 3: Output impedance magnitude of the BUF 124, in ohms. Node 20 includes the 47.5Ω isolation resistor, node 11 is at the Q6-Q7 collectors.

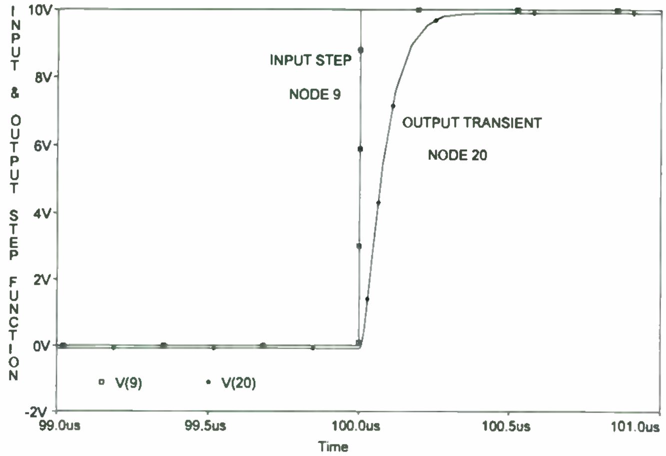

FIGURE 4: Output of the BUF 124 with a 10V step input. The initial slew rate at the start of the output rise is about 70V/μs and the rise time to 90% output amplitude is about 200ns.

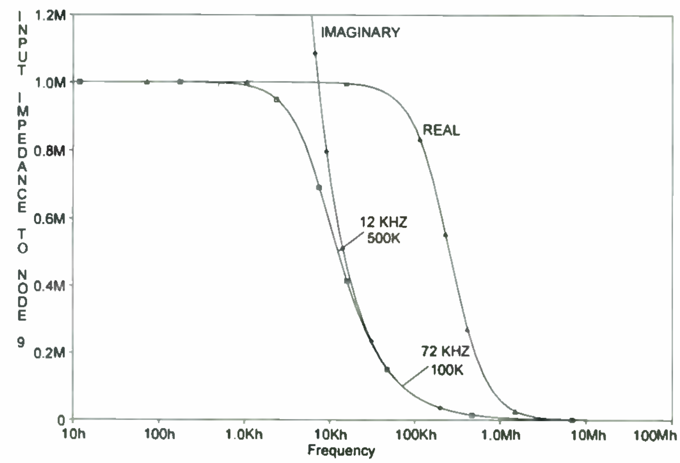

FIGURE 5: Real, imaginary, and magnitude parts of the input impedance to node 9 of the BUF 124 showing effects of the 22pF rolloff capacitor.

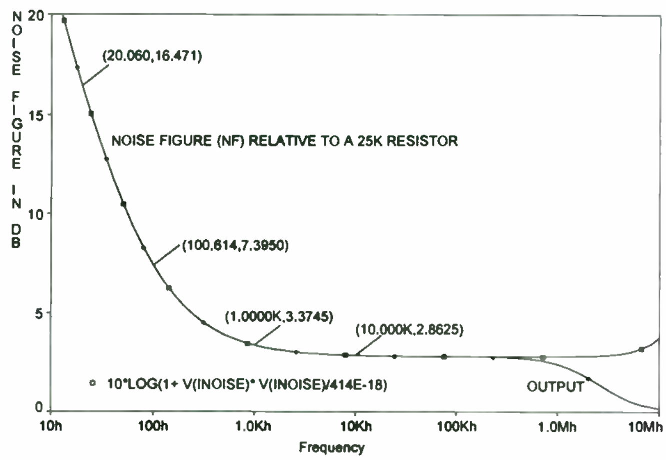

FIGURE 6: Noise figure (NF) of the BUF 124 relative to a 25k resistor at 27C, which has a uniform noise density of 20.34^{-9}V/$\sqrt{Hz}$ (see PSpice listings). PSpice always computes noise in V/$\sqrt{Hz}$. The NF is computed as 10*LOG(1+[amplifier noise power/reference noise power]), hence an NF of 3dB indicates that the amplifier is contributing the same noise power as the reference resistor.

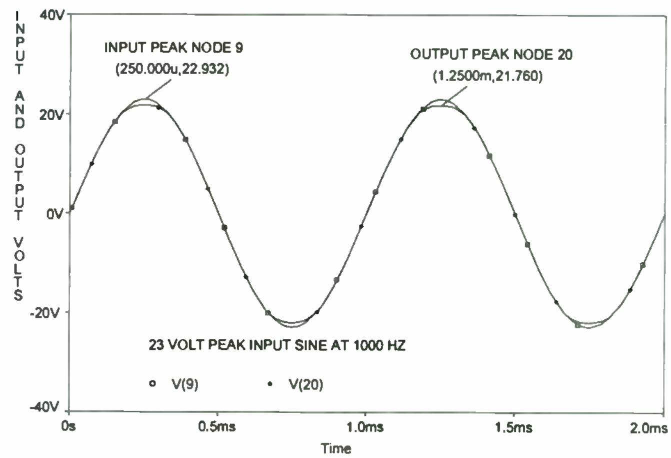

FIGURE 7: Severe odd-order distortion at 23V peak sine input. The slight difference between positive and negative peaks is caused by the 88mV offset.

which is distributed free of charge, is useful only for small circuits with a maximum of 30 semiconductor nodes, but it has most of the power of its unlimited cousin.

To me, PSpice represents the ultimate application of algorithmics to the numerical solution of some very lengthy and complicated models. The Borbely-Gaertner BUF 124 circuit is a real challenge to PSpice because of the JFET cascode stage direct-coupled to a bipolar cascode stage, and the lot in push-pull topology. In the real world, when I switch on the power supply, the branch currents cause the circuit to stabilize to its quiescent point in microseconds. But in PSpice, with ten semiconductor devices direct-coupled together, who does what to whom when the DC analysis begins? And when?

THE SOURCE FILE

Figure 1 is a reproduction of the BUF 124 circuit from the *SB* article with a set of numbers added (in circles). These are the node numbers I shall use to create a netlist that will indicate to PSpice how all of the parts are connected together.

The source file the PSpice program processes (Listing 1) is straightforward, with most lines showing <part name> <node connections> <value>. Note, for example, that R8 is connected between nodes 12 and 13 and has a value of 33.2kΩ. This article is not about how to write PSpice source files, but I will point out a few things. The Js are JFETS and the Qs are bipolar transistors, and each has a model name following the node connections. If you look farther down the listing you will see each of the models defined—in excruciating detail.

The line beginning VIN sends 1.0V of AC into nodes 9 and 0 (node 0 is always ground). The command line near the bottom, beginning .AC, orders the AC generator to sweep in third-octave steps from 10Hz–10MHz. Back near the top, the line beginning with .NODESET V(11)=0 is what keeps the software from going crazy trying to figure out how to start up the analysis of these eight transistors. It says, "hold node 11 at ground during power supply ramping then start the analysis from there." Without it, on my fast 486, the computation struggles for six or seven seconds while ramping the supplies to only 1% of final value, all the time attempting to converge node 11 anywhere less than infinity.

THE OUTPUT FILE

Having done all that, and received no errors during processing, what can we find out about the BUF 124? Listing 2 is a fragment of the output text file showing the DC condition at each node and the currents flowing at each DC voltage source. Notice that node 11 is 88mV below ground—very close to balance. The power-supply current is 13mA and the 2.5V reference currents are about 1.13mA, right on the money. If you

LISTING 1. PSpice BUF 124 Source File

See *SB* 1/94, p. 28 (*Fig. 6*) for circuit.

```
*** asterisks begin comments or deactivated commands ***
.OPTIONS NOMOD NOPAGE ITL1=200 RELTOL=.01
NUMDGT=3
.NODESET V(11)=0
.OP ;forces dc node calculations

VCC 90 0 24 ; power supplies
VEE 91 0 –24
* – pulse params:(vlow vhigh delay rise fall dwell [period])
* VZAP 9 0 PULSE(0 10 .1M 1N 1N .1M)
* – sine params:(voffset vpeak freq delay theta)
* VSIN 9 0 SIN(0 .1 1000 0 0
VIN 9 0 AC 1 ;audio generator input connections
* ITEST 20 0 AC .1 ;for poking current into the output

* Input Section: 10 is inverting * 9 is noninverting
* RSOURCE 9 0 .01 ;for shorting the input during ITEST
RTERM 10 0 10MEG ;to keep PSpice happy at node 10
R18 10 1 2K
R1 9 8 1K
C11 8 0 22pF
R2 8 0 1MEG

R3 90 6 1.4K
R6 91 7 1.4K
R4A 2 1 130 ;upper leg of the pot P1 +/– R4
R5A 3 1 190 ;lower leg of the pot P1 +/– R5

* d g s (drain gate source)
* | | |
J1 4 8 2 J2N5457 ;must use J instead of Q as in SB
J2 6 2 4 J2N5459
J3 5 8 3 J2N5460
J4 7 3 5 J2N5462

R10 1 11 1K ;RC feedback network
C4 1 11 22pF
R9 11 20 47.5 ;RC output network
C6 20 0 330P

RL 20 0 1MEG ;for loading the buffer if needed

*  c  b  e (collector base emitter)
*  |  |  | transistor push-pull array
q5 16 6 14 Q2N2907
Q6 11 12 16 Q2N2907
Q7 11 13 17 Q2N2222
Q8 17 7 15 Q2N2222

* little batteries to simulate 2.5V references
* because I can't find an LM336 model!
VRU 14 12 2.5 ;upper PNP pair
VRL 13 15 2.5 ;lower NPN pair 37 3

R7 90 14 150  ;Q5,Q8 emitter loads
R11 91 15 150
R8 12 13 33.2K
C3 14 12 0.1U ;pointless when using batteries for
C5 13 15 0.1U ;the LM 336's but here for completeness

* J1 and J3 should have matched IDSS. The IDSS can be approximated
* by computing BETA*(VTO^2).
.MODEL J2N5457 NJF(BETA=1.125M BETATCE=-.5 RD=1
RS=1 LAMBDA=2.3M
+     VTO=–1.372 VTOTC=–2.5M IS=181.3F ISR=1.747P
N=1 NR=2
+     XTI=3 ALPHA=2.543U VK=152.2 CGD=4P M=.3114
PB=.5 FC=.5
+     CGS=4.627P KF=10.45E–18 AF=1)

.MODEL J2N5459 NJF(BETA=377.6U BETATCE=–.5 RD=1
RS=1 LAMBDA=4.167M
+     VTO=–4.473 VTOTC=–2.5M IS=181.3F ISR=1.747P
N=1 NR=2
+     XTI=3 ALPHA=2.543U VK=152.2 CGD=4P M=.3114
PB=.5 FC=.5
+     CGS=4.627P KF=3.227E–18 AF=1)

.MODEL J2N5460 PJF(BETA=1.107M BETATCE=–.5 RD=1
RS=1 LAMBDA=20M
+     VTO=–1.75 VTOTC=–2.5M IS=222.4F ISR=2.177P
N=1 NR=2
+     XTI=3 ALPHA=29.8U VK=400.1 CGD=2.34P M=.4822
PB=1 FC=.5
+     CGS=2.92P KF=673.9E–18 AF=1)

.MODEL J2N5462 PJF(BETA=2.006M BETATCE=–.5 RD=1
RS=1 LAMBDA=27.5M
+     VTO=–2.271 VTOTC=–2.5M IS=222.4F ISR=2.177P
N=1 NR=2
+     XTI=3 ALPHA=29.8U VK=400.1 CGD=2.34P M=.4822
PB=1 FC=.5
+     CGS=2.92P KF=1.255F AF=1)

.MODEL Q2N2222 NPN(IS=3.108E–15 XTI=3 EG=1.11
VAF=131.5 BF=217.5
+     NE=1.541 ISE=190.7E–15 IKF=1.296 XTB=1.5
BR=6.18 NC=2
+     ISC=0 IKR=0 RC=1 CJC=14.57E–12 VJC=.75
MJC=.3333 FC=.5
+     CJE=26.08E–12 VJE=.75 MJE=.3333 TR=51.35E–9
TF=451E–12
+     ITF=.1 VTF=10 XTF=2)
*
.MODEL Q2N2907 PNP(IS=9.913E–15 XTI=3 EG=1.11
VAF=90.7 BF=197.8
+     NE=2.264 ISE=6.191E–12 IKF=.7322 XTB=1.5
BR=3.369 NC=2
+     ISC=0 IKR=0 RC=1 CJC=14.57E–12 VJC=.75
MJC=.3333 FC=.5
+     CJE=20.16E–12 VJE=.75 MJE=.3333 TR=29.17E–9
TF=405.7E–12
+     ITF=.4 VTF=10 XTF=2)

.AC OCT 3 10 10000K ; audio generator sweep command
* .TRAN 2E–5 2E–3 ; tran params:(interval total)
* .FOUR 1K V(20)
* .NOISE V(20) VIN
.PROBE ;switches on the graphics post-processor
.END
```

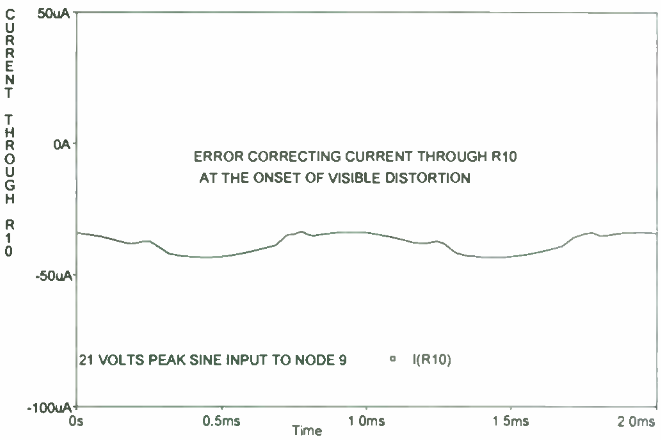

FIGURE 8: Error correcting feedback current through R10 of the BUF 124 at the distortion threshold of 21V peak sine input. Same time axis as *Fig. 7*. The bumpiness is caused by having only 50 samples per cycle in the .TRAN analysis.

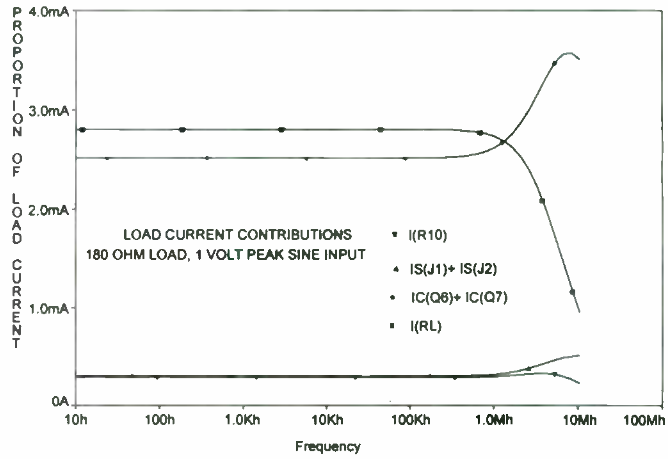

FIGURE 9: Output current contribution into a 180Ω load. The load current drops at high frequencies due to the 330pF capacitor C6. The Q6+Q7 collector current rises for the same reason.

look at Q5 to Q8 you will see the collector currents, IC, to be about 10mA. The drain current on the cascode JFET pairs, ID, is c. 1.86mA. This was achieved with 130Ω in the upper source leg, and 190Ω in the lower of J1 and J3. That's 320Ω total, simulating the 200Ω pot (P1 on the schematic) with a resistor on each side.

The bad news is 651mW of dissipation. This little baby is going to run warm, but that's the price we have to pay for the superb Class A performance I am about to show you.

PROBE GRAPHICS

The command .PROBE, seen in the source file, activates the PSpice graphics post-processor, which allows plotting of all of the variables specified in the source file. *Figure 2*, for example, shows the gain and phase from V_{IN}, which is connected to the noninverting input, to node 20, the output. The –3dB point is at 2.32MHz and the phase shift at this point is about 60°. The phase shift at 20kHz is 0.6°!

I already know more about this circuit than I could have learned in a hour on the bench, and I never switched on a voltmeter or lifted a soldering iron. Now I'm going to have some fun and make some measurements that are difficult to do on the bench. I measured the output impedance by connecting a 0.1A current generator to node 20, which flows through R9, and measured the resulting voltage at that node. I plotted V(20) divided by I(R9) and *Fig. 3* shows the result. At first I found it hard to believe that the output impedance of the BUF 124 was absolutely constant at 177Ω over the entire usable frequency range. I double-checked this by applying a voltage source and measuring the current, and got the same thing exactly.

Next, I measured the rise time. I connected a 10V pulse to the input and plotted the output (*Fig. 4*). The *SB* article says 200ns and, sure enough, there it is. The slew rate appears to be around 70V/μs based upon the initial slope.

The input impedance to node 9, computed as the node voltage divided by the current through R1, is plotted in *Fig. 5*. The magnitude, real, and imaginary parts are shown. At

LISTING 2. Fragment of PSpice Output File

**** SMALL SIGNAL BIAS SOLUTION TEMPERATURE = 27.000 DEG C

NODE	VOLTAGE	NODE	VOLTAGE	NODE	VOLTAGE	NODE	VOLTAGE
(1)	–.127	(2)	.107	(3)	–.477	(4)	2.475
(5)	–1.973	(6)	21.550	(7)	–21.520	(8)	–1.72E–09
(9)	0.000	(10)	–.127	(11)	–.088	(12)	19.770
(13)	–19.770	(14)	22.270	(15)	–22.270	(16)	20.480
(17)	–20.510	(20)	–.088	(90)	24.000	(91)	–24.000

VOLTAGE SOURCE CURRENTS
NAMECURRENT

VCC–1.331E–02
VEE 1.331E–02
VIN–1.717E–12
VRU 1.146E–03
VRL 1.126E–03
TOTAL POWER DISSIPATION 6.33E–01 WATTS

low frequencies, the 1 Meg resistor is in control, but above 2kHz or so the effects of the 22pF capacitor are felt. The magnitude drops to 500kΩ at 12kHz and to 100kΩ at 72kHz. This is not an important issue for the Sallen-Key filter, which rarely has component impedances above 30kΩ.

I then asked PSpice to measure the noise. *Figure 6* shows the plot resulting from computing the noise figure (NF) based upon that of a 10k resistor, which I chose since the Sallen-Key resistive circuit components are in that ballpark. A 10k resistor has a Johnson noise of 12.9^{-9}V/Hz and the noise figure is computed as 10 * LOG(amplifier noise/resistor noise). As you can see, the NF is just below 0dB at 1kHz rising rapidly at lower frequencies in the 1/f noise region. This region from 50Hz–1kHz is still some 70dB below the buffer's peak capability, however. If this noise level is translated to an equivalent SPL emanating from an amplifier/loudspeaker chain, I can legitimately apply A- weighting to this noise, which will actually cause it to droop—not rise—below 1kHz, rendering it effectively inaudible.

Next, I thought I should explore distortion. Since the BUF 124 is all Class A, there will not be a problem with distortion increasing as the signal gets smaller, which we see in many op amps. But how big a signal can I put through this thing? The trick is to observe the very sensitive feedback error-correction signal, which in the presence of no distortion will be exceedingly small. In the BUF 124, distortion correction from feedback will appear as a current through R10.

I applied a 23 volt-peak 1kHz sine wave to the input and asked PSpice to show me two cycles of the wave by use of the .TRAN command. *Figure 7* shows both the input and output waves, with obvious odd-order distortion. I then increased the signal, starting at 15V peak, and noticed the R10 current getting a bit wiggly at 21V peak (*Fig. 8*). This value of peak input signal is probably a reasonable limit to specify. PSpice can do a Fourier analysis and compute the total harmonic distortion (THD) based upon the sum-squared amplitudes of the first nine harmonics. The results are:

INPUT PEAK VOLTS	THD %
23	1.64
22	0.82
21	0.13
20	0.13
10	0.0067

No human would ever detect 1% distortion on a short transient music peak, so I consider

HOW TO COMPUTE JOHNSON NOISE

Johnson noise power for a resistor, v^2/R is given as 4*k*T*BW where

k = Boltzman's Constant = 1.38^{-23}
T = Kelvin temperature = C+273
R = resistance in ohms = 2.5E4 (for 25k)
BW = bandwidth = 1Hz in our analysis
v = resistor Johnson noise voltage

For a 25k resistor at 27C (300k) the noise power is $414^{-18}V^2$/Hz, or a noise voltage of 20.35^{-9}V/√Hz (the square root of 1Hz = 1√Hz, so it plots on the same frequency axis).

The tilt upward of the amplifier noise at low frequencies is caused by "flicker" noise in the semiconductor junctions and by current-induced noise in the circuit's resistors.

21–22V peak to be the rated maximum for the BUF 124. I might judge distortion based on the long-term RMS value of speech and music, which is about 12dB below peak. This 4:1 ratio of volts puts the RMS value passing through the buffer at around 5.5V, where the distortion products are down in the noise.

I was still wondering about the constant output impedance of 177Ω, and also about the output current contribution of the collectors of Q6-Q7 push-pull pair as compared to the output current contribution by the J1-J3 sources. I ran the buffer with a resistive load of 180Ω and, as expected, found one-half the input voltage at the output. I asked PROBE to plot the load current, the sum of the two Q6-Q7 collector currents, and the sum of the two J1-J3 source currents. As you can see from *Fig. 9*, about 80% of the load current is furnished by the bipolar transistors and about 20% by the JFETS.

SUMMARY

The Borbely-Gaertner BUF 124 unity-gain buffer is a wonderful design featuring high input impedance, constant low output impedance, low noise and distortion, and all Class A operation. As the article says, "the heart of the crossover is the buffer," and I am happy to confirm a perfect match to the requirements of Sallen-Key filters.

My friend Freddy used to say of his old wooden boat, "The only thing keeping her afloat is that all the worms are holding hands." An op amp with hundreds of transistors and tens of internal feedback loops reminds me of Freddy and his worms (sorry, Walt). It's a pleasure to see a minimalist (eight transistors—count 'em!) and sensible approach that can only result in a cleaner signal.

Keep in mind that the semiconductors I used in the PSpice simulation are those available in the US, not those from European manufacturers mentioned in the article. Also, they are used "out of the box" with no attempt to perform the device screening mentioned in the article. The authors suggested a 10% match between Q1 and Q3 (or J1 and J3 in

Continued on page 72

SOURCES

Microsim Corporation
20 Fairbanks, Irvine, CA 92718
(714) 770-3022
PSpice software

Old Colony Sound Lab
PO Box 243, Peterborough, NH 03458
(603) 924-6371; FAX (603) 924-9467
Evaluation PSpice with Prentice Hall book

Welborne Labs
971 E. Garden Dr., PO Box 260198, Littleton, CO 80126-0198
(303) 470-6585; FAX (303) 791-7856
BUF 124 Rev. C

Wayland's Wood World

CLEAN-CUT PLYWOOD

By Bob Wayland

You just cut a piece of very expensive hardwood plywood or veneered medium-density fiberboard (MDF) and the cut edges look as if you used a dull chainsaw. We've all faced this problem, oftener than we wish. There are, thank heaven, ways around the torn-edge disaster.

First, it may help to understand why this happens. If you are using a hand circular saw, as you cut, the teeth of the blade come up from the middle of the plywood and start to cut the thin outer-surface veneer, which has only a tenuous bond to the cross-banding underneath. If your blade is on the dull side and not perfectly flat, the veneer will splinter. The teeth catch on the veneer and pull it up away from the cross-banding.

With a table saw, the tom cut line is on the top, where the teeth of the blade come up through the plywood after the initial cut. This is the back of the cut, not the front (*Photo 1*).

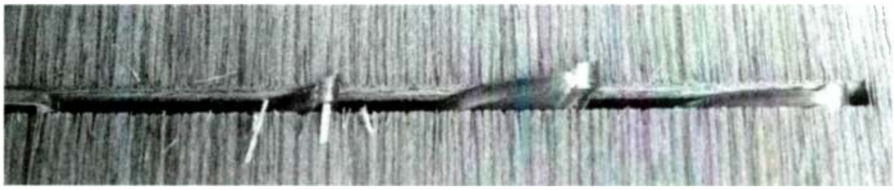

PHOTO 1: Splintering caused by improper sawing technique.

(Actually this is a worse case, the Neanderthal woodworker approach. The saw blade is a rip type and the plywood is ⅛", 3-ply mahogany—just to prove a point.) The slightest misalignment of the saw, blade, and/or the piece being cut, allowing the back part of the cut to rub against the sides, will make matters worse.

Even if the blade is perfect, you'll be lucky if the veneer doesn't splinter. At the front of the cut, where the teeth are cutting down into the body of the piece, the veneer is not splintered because the cutting action is pushing the veneer downward into the cross-banding.

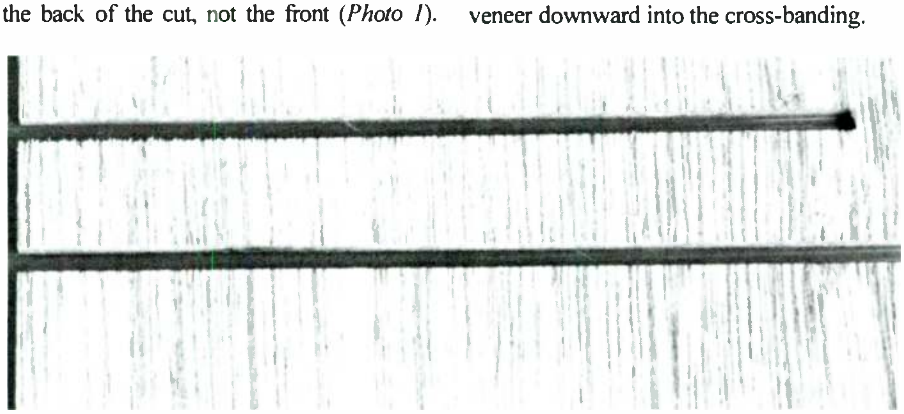

PHOTO 2: Effects of blade height on the smoothness of the cut. In the top cut the blade just cleared the surface; in the bottom cut it was set high enough to clear the gullets of the saw blade.

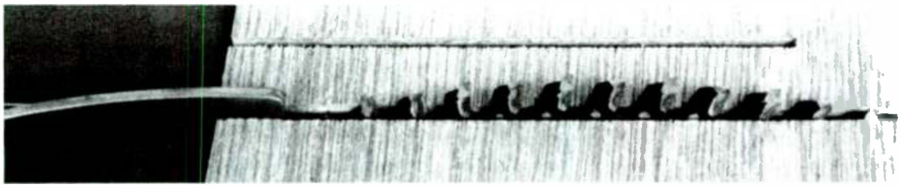

PHOTO 3: Fine-cutting plywood saw blade set with the gullet of the teeth at the surface of the plywood.

With this knowledge, how do we avoid the problem? The answer is another of woodworking's little secrets: *Always cut veneered boards so that the veneer will be held in place.* I will show you some simple techniques that can help you come up with your own solutions.

Before you start be sure your saw blade is perfectly flat and very sharp. You'll probably want to consider buying a saw blade especially designed for cutting plywood; e.g., a Freud TK-301 (6½"), TK-303 (7¼") or LU85 for table saws. If your saw blade has been used quite a lot, consider taking it to a professional sharpening shop.

BLADE HEIGHT

Saw-blade height also has an effect on the cut's smoothness. In *Photo 2*, the blade was set so the teeth just cleared the surface for the top cut. The bottom cut was made with the teeth extending high enough for the gullet between the teeth to be at the surface (*Photo 3*). The bottom cut is smoother.

Opinions on blade height vary. Kelly Mealer, for example, plays it safe, suggesting that the blade extend not more than ⅛" above the top of the material.[1]

Recall that for the under side (upper side on a table saw), the cut is much smoother, for two reasons: (1) the teeth are cutting down through the veneer into the cross-banding; and (2) the veneer is being supported (albeit poorly) by the saw table. If you place the piece you are cutting with a hand circular power saw so that the good side (the side exposed in

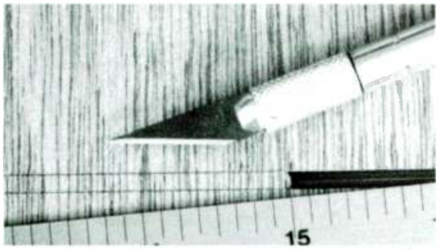

PHOTO 4: Scoring with a sharp knife before you cut can greatly reduce splintering.

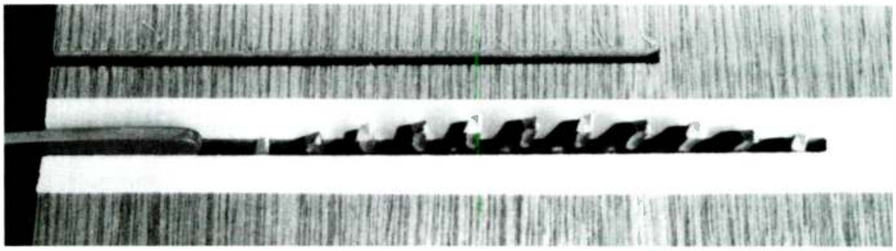

PHOTO 5: Placing masking tape along the cut will offer some restraint on splintering (a quick and dirty approach—and not the most effective).

the finished enclosure) is down, resting on a sheet of scrap plywood or particleboard, you'll produce a cleanly cut edge.

I have an old 4′ × 8′ piece of flooring particleboard that I use for my cutout subsurface. To make it last as long as possible, I usually set my hand circular saw so the teeth cut a ¼″ or less into the particleboard and cut where I haven't cut previously. I know this seems to contradict what I suggested above; but remember that the subboard keeps the surface veneer in place. (On a table saw this is not as simple, but with a little ingenuity you can accomplish the same effect.)

Another help is to attach a sheet of ¼″ plastic to the bottom of your circular hand power saw. Then lower the blade very carefully and cut a zero-clearance opening. The plastic will support the veneer, holding it down and keeping it from splintering. (If you attach the plastic with screws, be sure they're recessed to prevent gouging the veneer.) You can do the same on a table saw by placing a throat plate (table insert) of a sawable material and carefully raising the saw blade up through the insert, creating a zero-clearance guard.

You can also score with a sharp marking knife (e.g., an X-acto) just where you want the cut to be, then carefully align your cutting so that you cut down the center of the score. With care you can score both sides of the cut to produce a clean cut on both sides, as shown in *Photo 4*. If you want more insurance, you can also run masking tape along the cut and remove what is left after you're finished (*Photo 5*).

When you make the cut, slow down, but not enough to cause burning. A good way to judge your cutting speed is to listen to the sound of the saw. If the saw starts to reduce its turning speed, you're going too fast.

SOURCE

Woodworker's Supply
1108 North Glenn Rd., Casper, WY 82601
(800) 645-9292
Saw blades for cutting plywood

REFERENCES

1. *The Table Saw Book* (Taunton Press, PO Box 5506, Newtown, CT 06470-5506: $25.95 + $3 p&h).

Craftsman's Corner

HORNER IN THE CORNER

My corner-speaker project was built from Bruce Edgar's articles "A 70Hz Mini Horn " (*SB* 2/83, p. 7) and "The Edgar Midrange Horn" (*SB* 1/86, p. 7). The bass section uses a Pyle 6½" driver, the midrange is a 1½" dome (Audax HD13D37), and the tweeter is an E-V T-35. The crossover is a 6dB 500/4.5kHz unit from A&S Speakers. The cabinet is oak veneer plywood, finished with Danish Oil.

This speaker and its brother took about 30 hours to build, and cost $375. People admire the distinctive appearance of the pair, and usually wish to know more about them. I use them as the rear channels for my home-theater system.

Wes Baruth
Tarkio, MO 64491

PHOTO 1: IMP and Mitey Mike.

IMP & MITEY MIKE

The photo shows my IMP and Mitey Mike. To all the people who have worked on this project, my thanks—a great job! All worked fine from the first moment. I am very impressed with the results, which I have compared to a Neutrik—words fail me, fantastic. A lot of thanks to Bill Waslo for this great project.

Wladi Turkewitsch
6596 Gordola, Switzerland

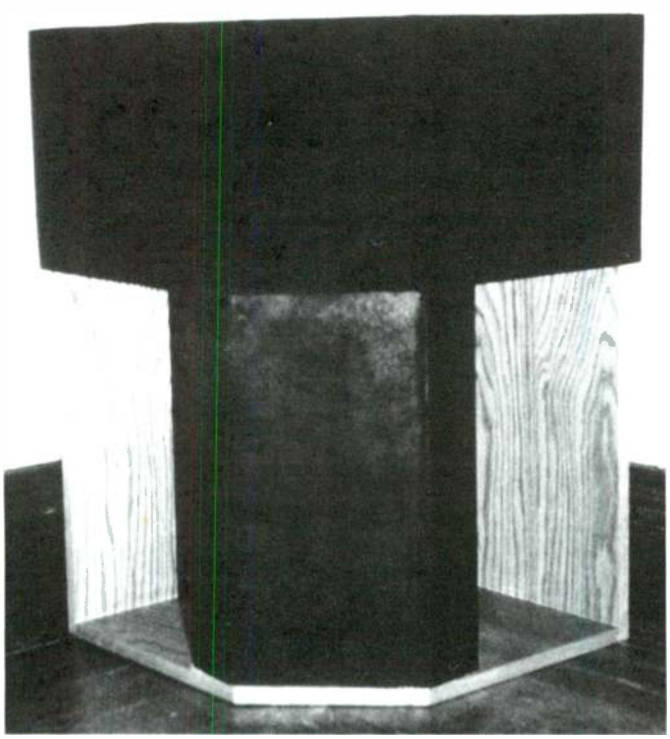

PHOTO 1: Corner speaker (front).

PHOTO 2: Corner Speaker (rear).

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BUILDING SPARE CHIP AMPS

I read Mark Gadzikowski's "More Power for Less" (*SB* 1/91, p. 36). The idea of a low-cost amplifier on a chip sounded good, but 20W at that distortion only made me curious. I have a NAD 3020, Marantz 240 (American made), and a few other amplifiers, so I had no need to fool with that "amp on a chip." Then I read Chris Clarke's letter in *SB 1/92* (p. 72), which mentioned the TDA 1514 AU. It was quite an ordeal, but I found six TDA 1514 AU chips. I killed two, but built four successful amps with 1% resistors and polypropylene caps.

I had a couple of big transformers on hand, so I built two ±16V power supplies. I realize this is low, but they *sound great*. I plan to build a proper power supply later. I wish I knew the placement and value of the capacitor Chris Clarke spoke of to get rid of the DC offset voltage, although according to specs it is only 2MV.

My speakers are similar to the ones in "Octaline Meets D'Appolito," by William Wagaman (*SB* 2/91, p. 38). Same drivers, almost same cabinet, similar crossovers, very close stuffing density. I built them before I ever saw that article. I'm sure his are much nicer/better than mine, but the article told me I'd been on the right track. I'm using one TDA per channel for the tweeters and one for each pair of woofers of the NAD to give four outputs. I split the output of each chain. The speakers were biwired so the hookup was very easy.

The possibilities do exist for active crossover multiamp systems at very reasonable cost. I worked out a foil pattern and used Radio Shack materials to etch it. The pattern is easy and cheap to reproduce. I'd never tried anything like this before, and the two boards I made worked perfectly. The chip is a 9-pin in-line device, with Pin #1 nearest the stripe printed on the body.

Ed Schilling
Eastover, SC

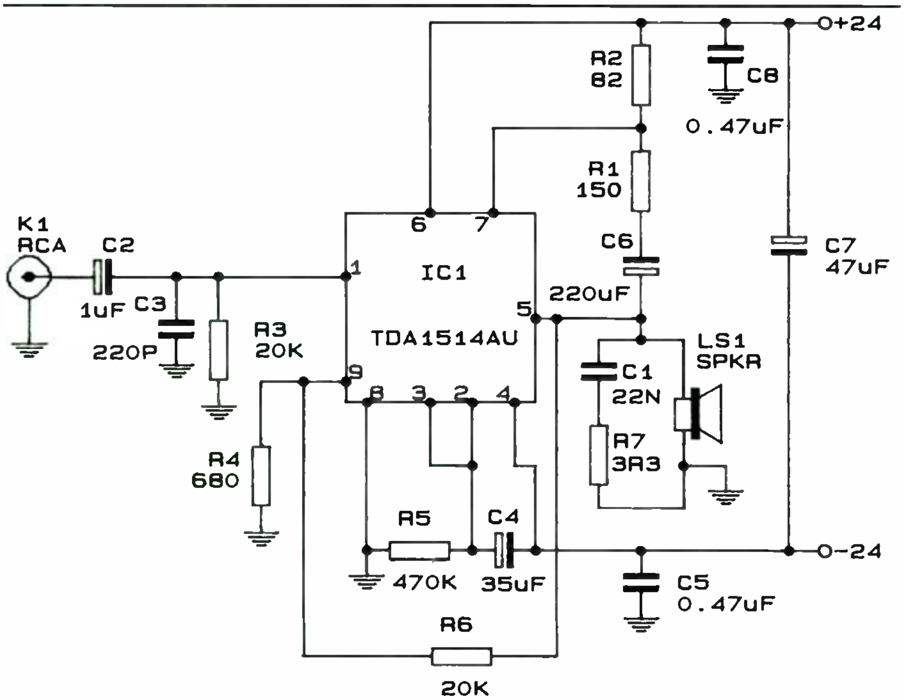

FIGURE 1: Schilling TDA1514AU Power Amp.

TABLE 1

SCHILLING POWER AMP PARTS LIST

Qty	Reference	Part
Capacitors		
1	C1	.022μF
1	C2	1μF
1	C3	220P
1	C4	33μF
2	C5, 8	0.47μF
1	C6	220μF
1	C7	47μF
Resistors		
1	R1	150
1	R2	82.5
2	R3, 6	20k
1	R4	680
Miscellaneous		
1	IC1	TDA1514AU
1	K1	RCA
1	LS1	SPEAKER

TDA1514AU SPECIFICATIONS

S/N 82dB

Slew rate 10V/VS

Power output with ±27.5V, R_L = 8Ω, 32W at 0.003%d_{TOT}

Power output with ±24V, R_L = 4Ω, 40W at 0.003%

Can be bridged for outputs up to 100W at 0.1% with ±24V, R_L = 8Ω

Externally adjustable closed-loop voltage gain of 20–46dB (2 resistors)

THE PC MUSIC HANDBOOK — BKPC2

Brian Heywood and Roger Evan — **$14.95**

The IBM PC has become one of the most popular business and personal computers ever designed. This proliferation of personal computing power has brought the possibilities of computer music within the budget of most amateur and professional musicians. This book takes you through the creative capabilities of the PC and helps you choose software and hardware for making music. Of great interest to the professional musician, the gifted amateur, or the just plain curious, it explains both the benefits to enjoy and the pitfalls to avoid, while suggesting a number of possible music systems that could bring out the best of your music skill or creativity. United Kingdom, 1991, 172pp., 5½" × 8½", softbound.

POWER VACUUM TUBES HANDBOOK — BKVN4

Jerry Whitaker — **$89.95**

Vacuum tubes are gaining favor as a technological choice, in a variety of electronic applications. This reference will help engineers refresh their understanding of the underlying principles and work with new developments as they arrive. The author covers the entire world of vacuum tube technology, with specific attention given to high-powered and high-frequency applications. Throughout the book, the style is clear and accessible as it describes and analyzes new applications in communications, medicine, and the military; provides guidelines for the design, specification, and maintenance of systems using vacuum tubes; and gives to-the-point advice on critical issues such as cooling and protecting tubes. 1993, 700pp., 6" × 9".

WOODWORKING VIDEO — VDNM1

National Academy of Mobile Electronics — **$49.95**

This video, aimed at auto sound installers but great for everyone, will help you to change forever the way you build enclosures. By learning techniques such as wood bending and contouring, viewers are able to design and construct enclosures which are much more aesthetically appealing than the norm. Through a process known as production-line woodworking, you will learn to reproduce work in a fraction of the time a job usually requires. Unique uses of several specialty tools such as standard and plunge routers are demonstrated, as well as the use of jigs. Also included is a section entitled "Build a Five-Sided Box in Twenty Minutes." Packed full of innovative, timesaving, and moneymaking ideas, *Woodworking* is a must for the professional and do-it-yourselfer alike. Formerly Old Colony's #VDWWG. Produced by Michael Breault. 1992, VHS, 60:00.

PASSIVE CROSSOVERS VIDEO — VDNM3

National Academy of Mobile Electronics — **$49.95**

This *Passive Crossovers* video, also geared toward auto sound, explains how a passive crossover directs the proper frequencies to the appropriate transducer, as well as discusses the different components in a passive crossover network. Details such as the different types of capacitors, inductors, and resistors are covered, along with instruction on how to interpret specifications for each component and the formulas needed to derive the proper component values. The design phase of this video covers two- and three-way first through fourth order Butterworth filters. Throughout this video, the circuit designs and topologies are laid out in easy-to-follow schematics, providing you with simplified yet accurate information for designing passive crossover systems. By Michael Breault. 1993, VHS, 30:00.

SUBWOOFER ENCLOSURE VIDEO — VDNM2

National Academy of Mobile Electronics — **$49.95**

Subwoofer Enclosure is designed to leave the viewer with a good understanding of the theory and technique needed to build a quality speaker enclosure. The acoustic theory portion covers sound, decibels, quarter waves, SPL, and more. Transducers are explained in full detail, as well as the formulas necessary to determine the proper or optimum size for acoustic suspension and tuned port enclosures. Calculating speaker displacement and port area and length are also taught, along with the process of deriving Thiele/Small parameters from any loudspeaker. Oriented toward auto sound, this video's combination of basic theory and fundamentals with helpful hints and techniques makes it of benefit to any speaker design enthusiast. Produced by Michael Breault. 1993, VHS, 55:00.

REVOLUTIONARY SYMDRIVE BALANCED LINE DRIVER

NEW THIS ISSUE

Recently developed in Austria, new Symdrive® technology substitutes JFETs for tubes and yet still provides rich, tube-quality sound through a balanced line driver unit which is used within your existing preamp. Its supply voltage can thus vary from +30VDC to +100VDC, and since it draws only about 25mA in the stereo version, it can be built into almost any stereo design without using a separate DC source. Should your existing preamp have a lower supply voltage, you would indeed need to use a separate supply. Symdrive is best suited to drive two power amps in a bridged configuration, but can be applied in other ways as well. From Hudelist Stage & Studio Engineering. Further information available on request. Complete instructions included. Cases not supplied. Kits are configured for 110V/60Hz, but are supplied without power cord. PLEASE SPECIFY WHETHER FOR USE WITH SOLID STATE OR TUBE PREAMP. Please allow 4-6 weeks for delivery. Purchasing options available:

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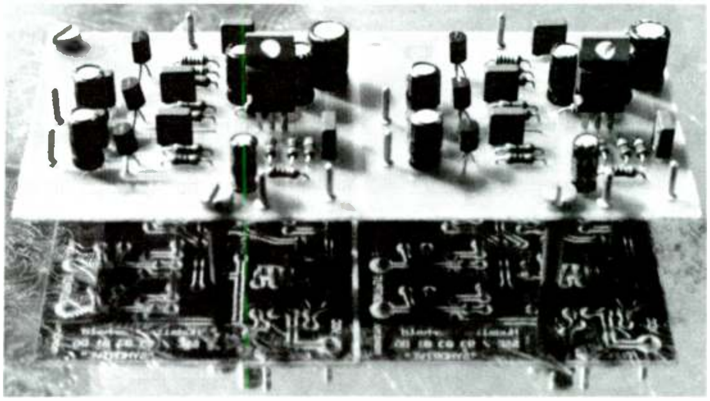
PCBG-6B	Symdrive Supply Unit PC Board, without parts	**15.95**
KG-6A	Symdrive Balanced Line Driver, complete unassembled kit, mono	**49.95**
KG-6AA	Symdrive Balanced Line Driver, complete assembled unit, mono	**69.95**
KG-6B	Symdrive Supply Unit, complete unassembled kit	**79.95**
KG-6BA	Symdrive Supply Unit, complete assembled unit	**99.95**
KG-6SA	Symdrive Stereo Unit plus Supply Unit, fully assembled in plastic case	**194.95**



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> $200.00	6.00	30.00	25.00	30.00	40.00	50.00

NEW THIS ISSUE

RCA RECEIVING TUBE MANUAL — BKAE4 — $12.95

Radio Corporation of America

This is a new reprint of a classic 1959 manual prepared to assist those who worked or experimented with vacuum tube circuits. Of great interest to engineers, service technicians, experimenters, students, radio amateurs, audiophiles, and all others technically interested in tubes, the material in this book is wide in scope as well as deep in detail. Chapters include: Electrons, Electrodes, and Electron Tubes (Electrons, Cathodes, Generic Tube Types, Diodes, Triodes, Pentodes, Beam Power Tubes, Multi-Electrode and Multi-Unit Types, Television Picture Tubes); Electron Tube Characteristics; Electron Tube Applications (Amplification, Rectification, Detection, Automatic Volume or Gain Control, Tuning Indication with Electron-Ray Tubes, Oscillation, Deflection Circuits, Frequency Conversion, Automatic Frequency Control); Electron Tube Installation (Filament and Heater Power Supply, Heater-to-Cathode Connection, Plate Voltage Supply, Grid Voltage Supply, Screen-Grid Voltage Supply, Shielding Dress of Circuit Leads, Filters, Output-Coupling Devices, High-Voltage Considerations for Television Picture Tubes, Picture-Tube Safety Considerations); Interpretation of Tube Data; Receiving Tube Classification Chart; Tube Types—Technical Data; Picture Tubes Characteristics Chart; Electron Tube Testing; Resistance-Coupled Amplifiers; Circuits; Outlines; Index; and Reading List. Filled with charts and graphs. RCA Technical Series #RC-19. 384pp., 5⅜" × 8⅜", softbound.

75 YEARS OF WESTERN ELECTRIC TUBE MANUFACTURING — BKAE5 — $19.95

Bernard D. Magers

Subtitled "A Log Book History of Over 750 WE Tubes Including Dates of Manufacture," this enlightening volume was written by the WE Senior Engineer charged with the sad duty of conducting the phaseout of the Kansas City Works in 1988. With a listing and description for each, the book chronicles the history of the 785 tubes Western Electric produced during 1913-1988. And in addition to the tube specs and backgrounds, also included are sections on Nomenclature, References, A Brief History, "D"-Tube Specifications, Other WE Tubes, and A Summary of Manufacturing Notes. Many photos and drawings. 1992, 144pp., 8¼" × 11", softbound.

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Edward R. McCue and Richard H. Talaske, editors

Plans, photographs, and descriptions of fifty music education facilities from around the world, with supplementary explanatory text and eleven essays on the design process. 1990, 236pp., softbound.

ORIGINS IN ACOUSTICS — BKAC13 — $33.95

Frederick V. Hunt

A history of acoustics from antiquity to the time of Isaac Newton, this volume surveys sources beginning with the Ancient Greeks and Romans and documents experiments and observations by scholars from the arab world during the Dark Ages and by pre-Newtonian scientists in Europe. 224pp., hardbound.

HALLS FOR MUSIC PERFORMANCE: TWO DECADES OF EXPERIENCE, 1962-1982 — BKAC11 — $33.95

Richard H. Talaske, et al., editors

Along with co-editors Ewart A. Wetherill and William J. Cavanaugh, Talaske examines the standards of quality and technical capabilities of eighty performing arts facilities through the use of drawings, photos, and technical and physical data. 1982, 192pp., softbound.

ACOUSTICS OF WORSHIP SPACES — BKAC10 — $33.95

David Lubman and Ewart A Wetherill, editors

Drawings, photographs, and accompanying data of existing worship houses provide vital information on problems and answers concerning the acoustical design of chapels, churches, mosques, temples, and synagogues. 1985, 91pp., softbound.

THEATRES FOR DRAMA PERFORMANCE: RECENT EXPERIENCE IN ACOUSTICAL DESIGN — BKAC12 — $33.95

Richard H. Talaske and Richard E. Boner, editors

Recent theatre designs by acoustical consultants in North America and abroad are explored in this volume, by means of descriptions, plans, and photos. Also included are essays on theatre design and an extensive bibliography. 1987, 167pp., softbound.

SILVER SONIC BALANCED INTERCONNECT — MWRSSBL1 — $2.99/ft.

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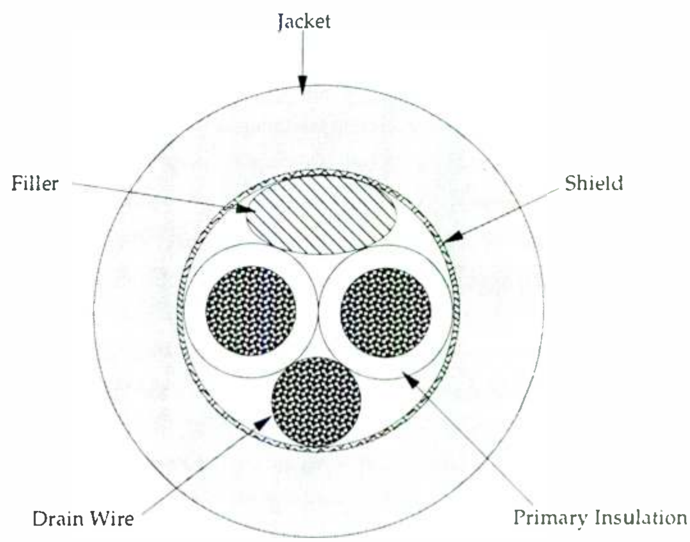

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Product Review

Signet's SL280B/U

By Vance Dickason
Contributing Editor

Signet's SL280B/U, Audio Potentials Corp., 1920 Enterprise Pkwy., Twinsburg, OH 44087, (216) 425-8222, FAX (216) 425-9339. Price: $700/pair.

Signet loudspeakers were introduced in early 1990 as the brainchild of Audio-Technica's president, Jon Kelly. Since that time, the loudspeaker divisions of Audio-Technica, Design Acoustics, and Signet have split from the original company and are now part of Audio Potentials, a new company captained by former A-T president Kelly.

The Signet line, as originally conceptualized, is composed of four moderately priced high-end two-way speakers using European drivers. The subject of this review is the relatively new vinyl cabinet version of Signet's top model, the SL280B/U. This speaker was designed by Andy Lewis, a former AR engineer and subsequently a Signet engineer, who nowadays does design work for Apogee Acoustics.

The SL280B/U is a compact (25" height) two-way design using an 8" poly cone woofer and a 1" aluminum diaphragm tweeter. Both drivers are mounted on a flat baffle in the traditional woofer below the tweeter format, with the tweeter recessed and the woofer mounted on the cabinet's surface.

For damping any tweeter diffraction off the protruding woofer frame, the SL280 has a ¼-inch-thick die-cut foam pad covering the tweeter mounting flange and the adjoining area above the woofer frame. The vent is mounted on the rear of the enclosure, which is finished in an attractive black oak vinyl on all six sides. A custom panel with two sets of gold-plated five-way binding posts provides the amplifier connections. Appropriate removable shorting bars allow optional biwire operation.

WOOFER LOW END

The woofer is a variant of the venerable SEAS P21REX series. This model, unavailable off the shelf, uses the Dynamic Damping option, which consists of a few shorted turns of wire at the top and bottom of the voice coil. This patented system's effect provides a modicum of magnetic "braking" as the voice coil reaches its travel extremes, which has obvious benefits in reflex and transmission-line enclosures.

Combining a cast frame, poly cone, rubber surround, and hard poly dust cap, the P21REX is a very capable performer. After measuring its parameters via the LMS/LEAP system, I performed a computer simulation using the measured enclosure net volume of about 1ft^3 with an 8" length 2.5" ID port tube. With an f_S of 35Hz and a Q_{TS} of about 0.4, the P21REX produces a Chebychev/Butterworth type of alignment where the initial rolloff is fairly shallow, followed by a steeper slope beginning at a lower frequency.

Figure 1 gives the 2.83V and 10V (1W and 12.5W) vented box simulation frequency response. The f_3 at the 1W level for the simulation is 46Hz. Looking at the group delay and excursion curves in *Fig. 3*, you can see that

PHOTO 1: The Signet SL280B/U.

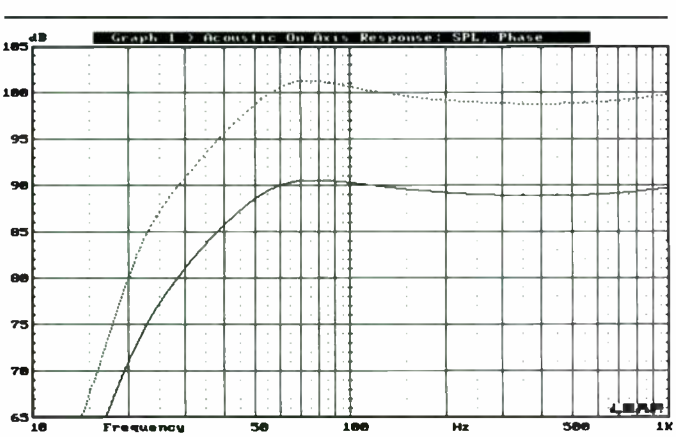


FIGURE 1: LEAP low-frequency simulation of Signet SL280 woofer (solid = 2.83V, dot = 10V).

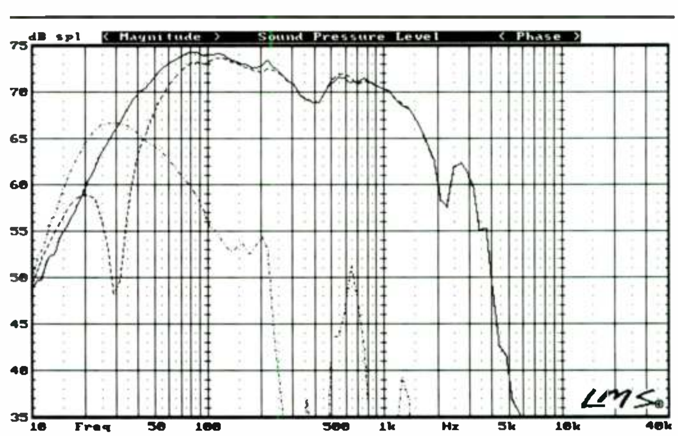


FIGURE 2: Near-field woofer frequency response (solid = woofer-port, dash = woofer, dot/dash = port).

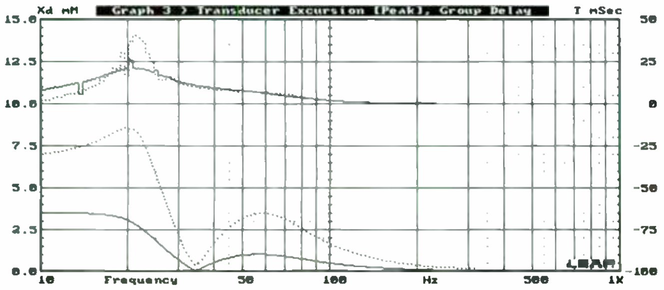


FIGURE 3: Group delay and cone excursion curves for *Fig. 1* (solid = 2.83V, dot = 10V).

FIGURE 4: Woofer full-range on-axis anechoic frequency response (does not include rear port radiation).

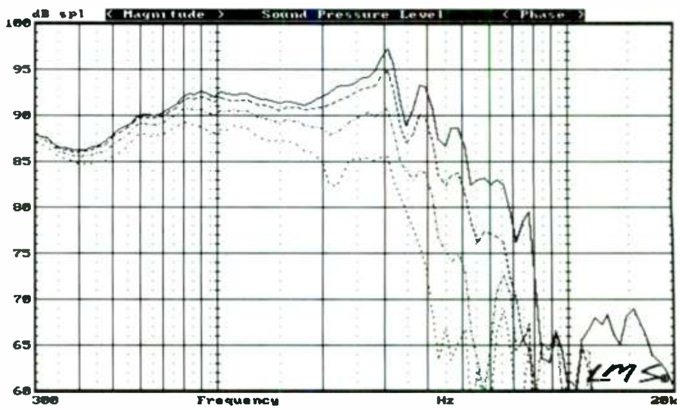


FIGURE 5: Woofer on- and off-axis frequency response (solid = 0°, dash = 15°, dash/dot = 30°, short dash = 45°).

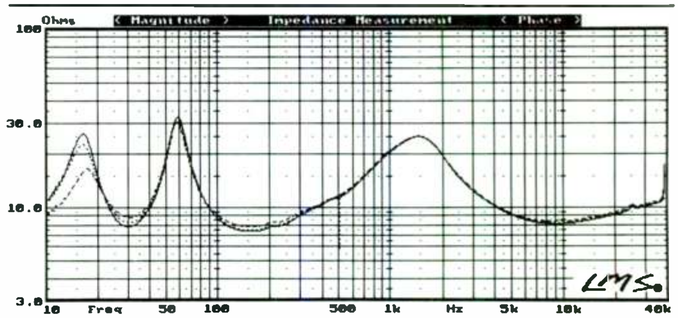


FIGURE 6: Dynamic system impedance (solid = 0.9V, dot = 2.83V, dash = 6.33V).

the group-delay profile is similar to the PSB Stratus Mini (a Chebychev/Bessel), reviewed in *SB* 3/93 (p. 48). Although this yields a transient performance inferior to a QB3 type of alignment, the result is subjectively more than satisfactory and provides reasonably low bass extension.

At 10V the SPL reaches approximately 100dB and the excursion is at a maximum 3.5mm above the tuning frequency at 58Hz, with the same level of excursion below the tuning frequency at 28Hz. Given a driver X_{MAX} of 3mm, and using the X_{MAX} plus 15% criterion as the limit of acceptable distortion, the P21REX remains linear up to a moderate and typical loud listening level of most home speakers. This does not factor in the Dynamic Damping feature incorporated into this woofer (LEAP cannot model this particular woofer variation), which suggests that this product should be working quite well at the upper limits of its dynamic range.

LEAP's prediction of the low-end response was confirmed by using LMS. Normally, this could be done by the ground-plane method. Since the port is located on the rear, however, the ground-plane technique will not allow for the proper integration of the port and woofer diaphragm acoustic outputs. To accomplish this, I made near-field measurements of both the port and the woofer and then combined them mathematically.

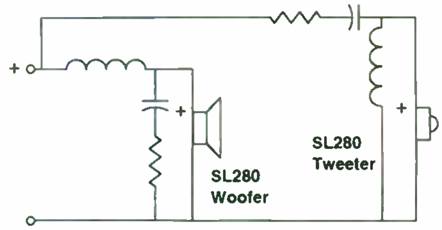


FIGURE 7: SL280 crossover topography.

This is accomplished by placing the mike within ¼" of the center of the woofer dust cap and then at the center of the port, level with the port flange (rear baffle surface), for the second measurement. The results are then scaled according to their radiating area and

FIGURE 8: Woofer response with and without low-pass network.

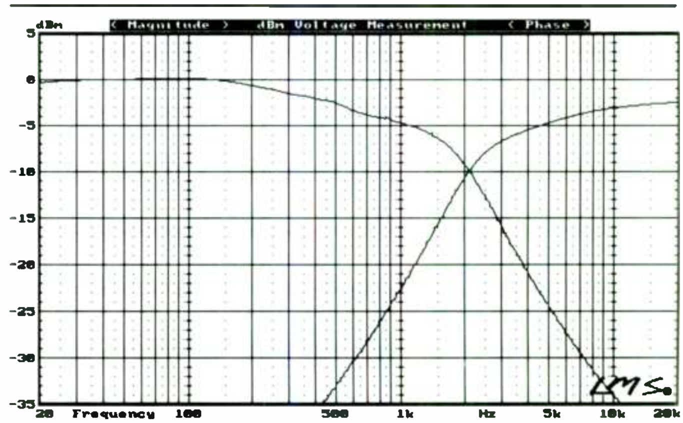


FIGURE 9: SL280 crossover transfer functions (measured).

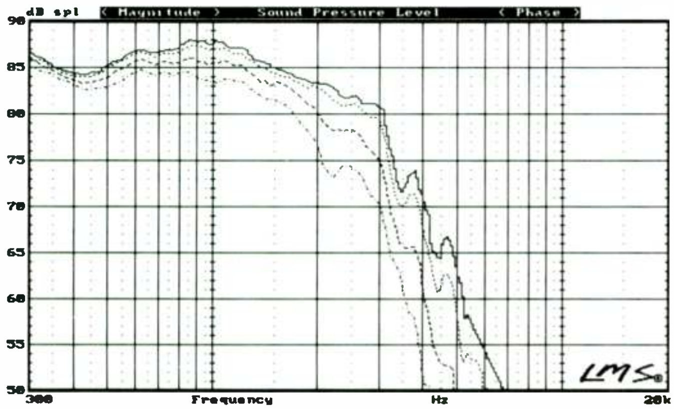


FIGURE 10: Woofer frequency response with low-pass network (solid = 0°, dot = 15°, dash = 30°, dash/dot = 45°).

FIGURE 11: Tweeter on-axis frequency response.

FIGURE 12: Tweeter on- and off-axis frequency response (solid = 0°, dot = 15°, dash = 30°, dash/dot = 45°).

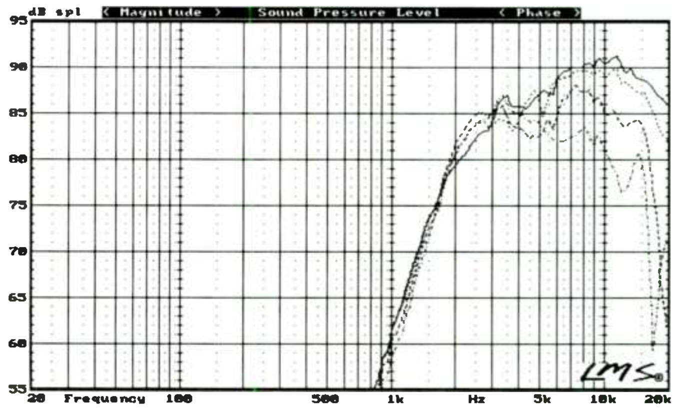


FIGURE 13: Tweeter frequency response with high-pass network (solid = 0°, dot = 15°, dash = 30°, dash/dot = 45°).

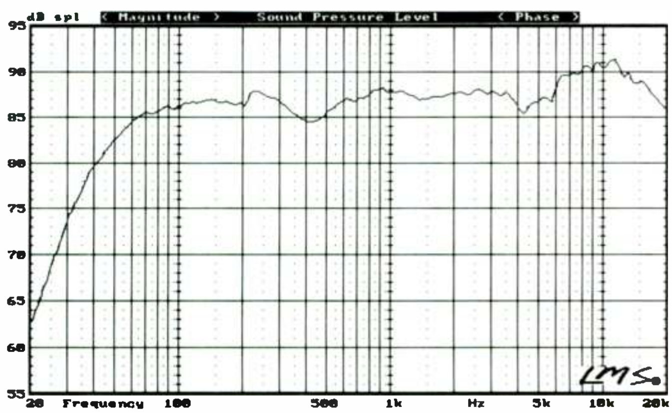


FIGURE 14: Full-range on-axis system frequency response (does not include rear port radiation).

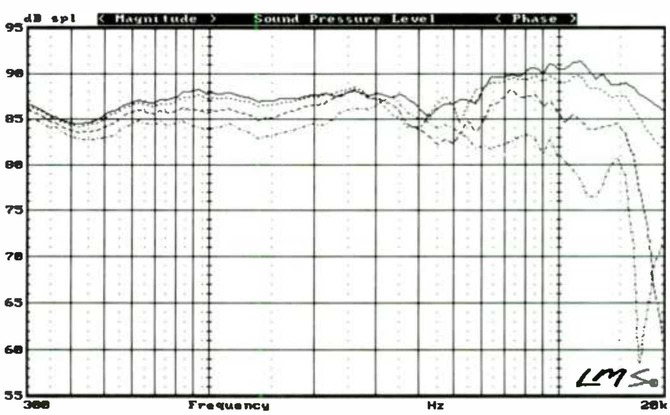


FIGURE 15: On- and off-axis system horizontal frequency response (solid = 0°, dot = 15°, dash = 30°, dash/dot = 45°).

translated from a near-field sound-pressure level to a far-field sound-pressure level, using the formula

$$P_{FAR}/P_{NEAR} = 0.2831(S_D{}^{.5})$$

where S_D is given in square meters.[1] I then converted these results to decibels by multiplying them by 20Log. This can also be accomplished by using the decibel conversion tables in the appendix of Beranek's *Acoustics*.[2]

After the curves have been scaled according to their respective areas, the port curve is subtracted from the woofer curve (since it is very nearly out of phase with the woofer). The result, illustrated in *Fig. 2*, has an f_3 of 47Hz, which is quite close to the simulation. This method works well and produces a good correlation with low-frequency-response data produced by ground-plane measurements of speakers with the port mounted on the front.

WOOFER RESPONSE

Figure 4 shows the woofer's full-range response without any crossover at a 2.83V input level. I spliced a ground-plane measurement with a gated sine-wave measurement made on a short 5½′ tower, both taken outdoors. I didn't bother making a third splice of the near-field measurement below 100Hz to include the port, so this measurement does not show the complete low-end response. The response is fairly smooth and has a continuously rising response profile (the anechoic response caused by the front baffle diffraction) out to 3kHz, where a sharp breakup mode occurs. *Figure 5* is the woofer's off-axis response. The peak at 3kHz is mostly an on-axis phenomenon, which is normal. The smooth off-axis transition should easily allow for a 2–2.5kHz crossover frequency (the response is down only 4dB at 30° off axis at 2.5kHz). The

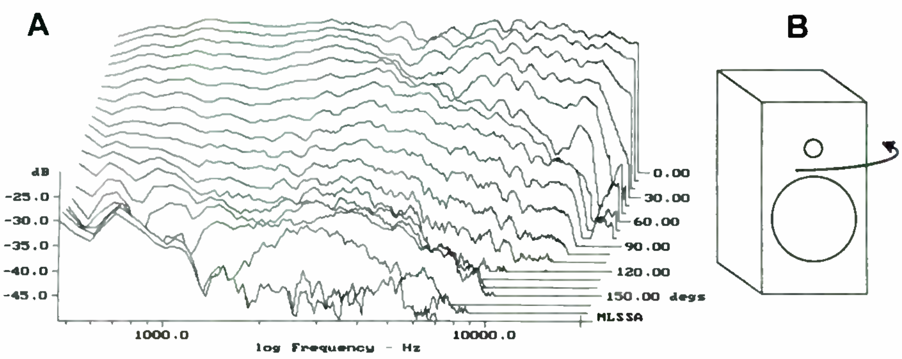


FIGURE 16: a. MLSSA 0–180° on- and off-axis horizontal response display; b. direction of the polar plots.

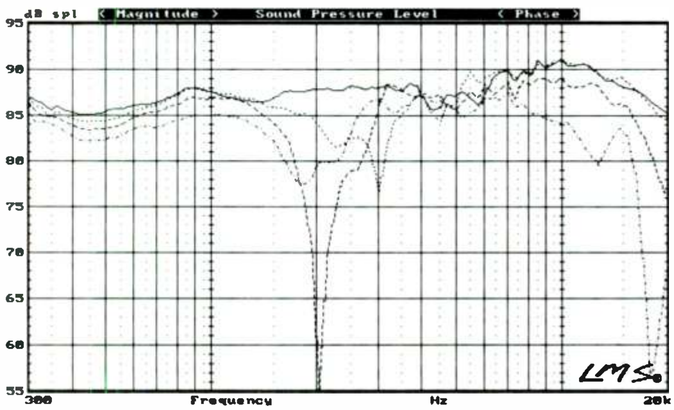


FIGURE 17: On- and off-axis system vertical frequency response (solid = 0°, dot = 15°, dash = 30°, dash/dot = 45°).

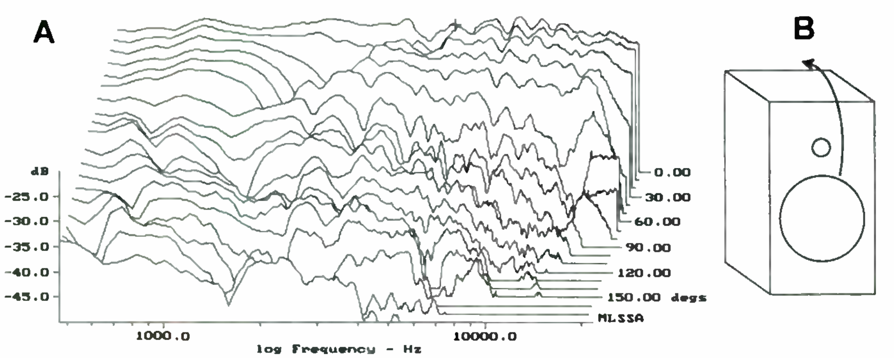


FIGURE 18: a. MLSSA 0–180° on- and off-axis vertical response; b. direction of the polar plots.

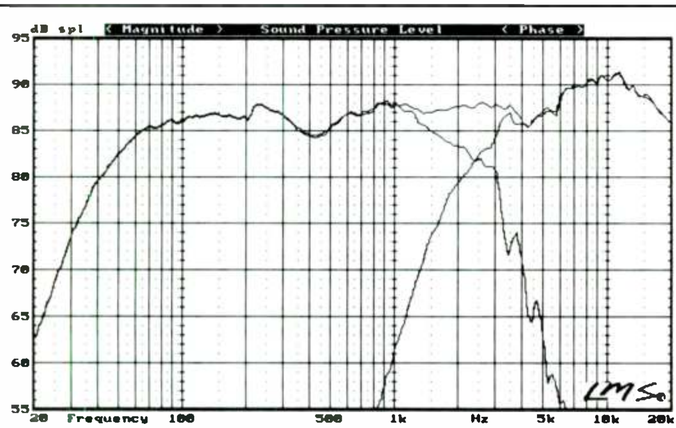


FIGURE 19: Woofer, tweeter, and system frequency-response graph.

FIGURE 20: Frequency response comparison, in phase and tweeter electrically out of phase.

FIGURE 21: Full-range (does not include rear port radiation) anechoic frequency magnitude and phase response.

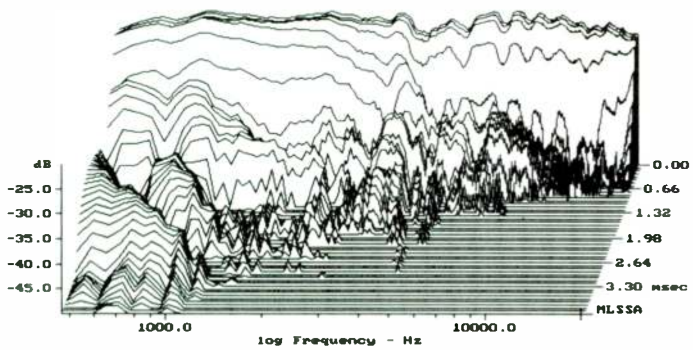


FIGURE 22: MLSSA cumulative spectral decay curve.

2½" port diameter was a good choice for this box and is about as large a vent as could be tolerated and still fit the depth of the enclosure. *Figure 6* shows the woofer impedance (actually the system impedance including the tweeter and network) done at different voltage levels of 0.9V, 2.83V, and 6.33V (0.1, 1, and 5W). This test is made by taking both a current shunt (admittance) and a voltage curve at each power level (actually the voltage curves were only taken at the 0.9V level and then scaled for the other voltages), and then using Ohm's law.

Since R = E/I, the voltage curves are divided by the current curves to produce the impedance reading in volts. Voltage is converted to ohms using the LMS decibels-to-linear conversion utility—once to convert to VdBm, and again to convert it to ohms. The result shows the nonlinear effects of the port tube, which is typical of virtually all vented loudspeakers. As voltage increases and the velocity of airflow through the tube increases, the volume of air through the tube steadily decreases, until the port is effectively closed at very high SPL levels. However, for any 8" woofer, a 2½"–3" vent is satisfactory for subjective performance, as it is with the Signet SL280.

WOOFER CROSSOVER

Figure 7 is the Signet SL280 network topography. This speaker's 2.5kHz crossover frequency should prove an adequate compromise for producing a good system power response, as well as attenuating the tweeter quickly enough to prevent excessive excursion. The woofer response, with and without the network, is illustrated in *Fig. 8*.

The low-pass section required to do this appears to be a first-order filter with an impedance conjugate. The value of the resistor is lower than you would expect for an impedance conjugate (sometimes called a "zobel"), however, which means that a better description would be that of a second-order section with some resistive response shaping; something that is easy to do using a computer optimization program.

The woofer response is contoured starting at about 450Hz and begins its rolloff at 1kHz, with an additional breakpoint at 3kHz. Over the two-octave spread between 2kHz and 8kHz, the woofer response falls 48dB, which

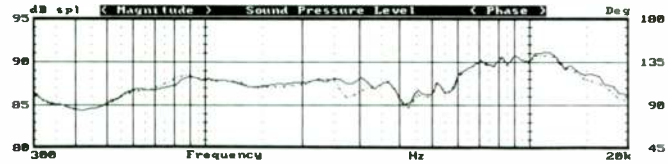


FIGURE 23: Frequency-response comparison for both SL280 speakers supplied by manufacturer.

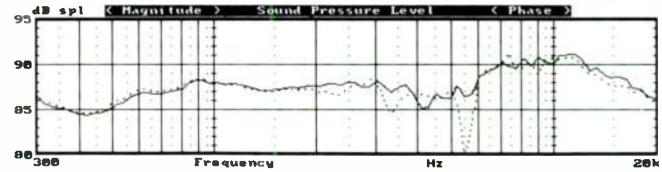


FIGURE 25: Frequency response with and without grille frame in place (solid = without frame, dash = with frame).

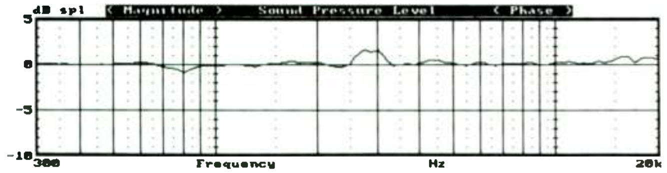


FIGURE 24: Difference curve for *Fig. 23*.

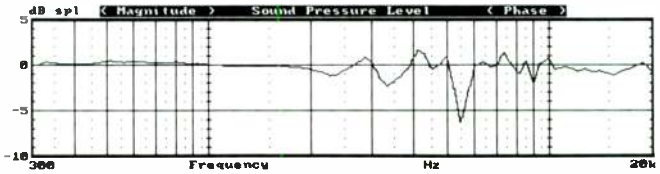


FIGURE 26: Difference curve for *Fig. 25*.

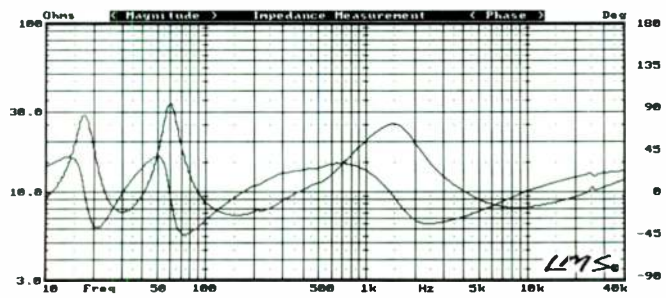


FIGURE 27: SL280 impedance magnitude and phase.

is approximately fourth-order in that region. This levels off the response to produce a flat anechoic profile, leaving an overall efficiency level of 87–88dB.

Figure 9 shows the measured (not simulated) transfer functions of both woofer and tweeter networks. The voltage changes in the response begin at about 200Hz with a second breakpoint at 1.8kHz, typical of almost all low-pass woofer crossovers.

Figure 10 shows the woofer off-axis frequency response and network combination, which is quite well behaved out to 45° off axis. This smooth off-axis response in the woofer upper frequencies is the main requirement for producing a smooth power response in the horizontal axis. A flat power response provides lateral reflections to the ear that have nearly the same timbre as the direct-radiated on-axis information.

Network components for the low-pass section are of high quality and include an 18-gauge air-core inductor, a Mylar® capacitor, and a 10W wire-wound resistor. The Mylar caps in both the crossover sections are Signet brand, custom made for the company. The network is assembled on a single PC board with separate grounds for each section.

THE TWEETER

The SL280's SEAS tweeter is one of its 1" aluminum dome series. The H398 uses the mesh screen and Mylar ⅜" diameter insert for a response-correcting phase plug. The impedance (not shown) is very flat, with the resonance damped by magnetic fluid.

This series, which includes the cavity version H400, has proved popular with many designers and is used in a number of well-reviewed speakers, including the Totem Model 1 and Counterpoint speakers. Tweeter response is shown in *Fig. 11*, with the off-axis response out to 45° in *Fig. 12*. This curve is quite smooth and rises slightly with increasing frequency, but within 2dB from 2–10kHz. The 25kHz peak is the typical breakup mode for aluminum domes. Off-axis response is also typical, although the dome's response drops off rapidly above 15kHz at 30° off axis. Not perfect, but since it takes nearly 8dB of change to be noticeable at 20kHz, the consequence is not really serious.

With the high-pass network in place, the tweeter's frequency response, on and off axis, is depicted in *Fig. 13*. The 6dB down point is approximately 2.5kHz, the company's stated crossover frequency. The response is tilted somewhat upward, rising by 4–5dB from 5–10kHz. The response attenuates about 33dB over the two-octave distance from 3kHz to 750Hz, giving it roughly a third-order response. Tweeter network components include a Mylar capacitor, an air-core inductor, and a 10W resistor. The inductor is wound on a plastic bobbin with small 26-gauge wire, producing a compact size with a high DCR. This high series resistance generally has a negligible effect on most high-pass circuits and in this case does not drastically affect the network's transfer function.

THE COMPLETE SYSTEM

Figure 14 is the speaker's full-range response, showing it fits into a ±3.25dB window from about 60Hz–20kHz. This profile is well behaved and fits a much tighter ±2dB window from 60Hz–6kHz, which is quite good. Off-axis response, shown for the horizontal plane in *Fig. 15*, is very good, with only minor deviations out to 45°. This is depicted in a "waterfall" format in *Fig. 16* (produced using the MLSSA FFT analyzer and ACO Pacific 7012 microphone) and gives a more complete picture of the horizontal polar response to 180°. This off-axis (power) response profile is essential for producing a high-quality loudspeaker. The vertical off-axis response plot is illustrated in *Fig. 17*, showing the typical cancellations that occur in that plane. *Figure 18* gives the MLSSA vertical off-axis array to 180°.

The crossover frequency can be easily detected in the graph shown in *Fig. 19*, which displays the individual driver/network responses in conjunction with the on-axis system response. The crossover occurs slightly below 2.5kHz where both driver responses are approximately 6dB down. In the crossover region from 1–4kHz, you see the typical combination of a steep tweeter rolloff combined with a shallower woofer rolloff to produce a flat summation.

Figure 20 shows the response with both drivers in phase, as in *Fig. 19*, along with the

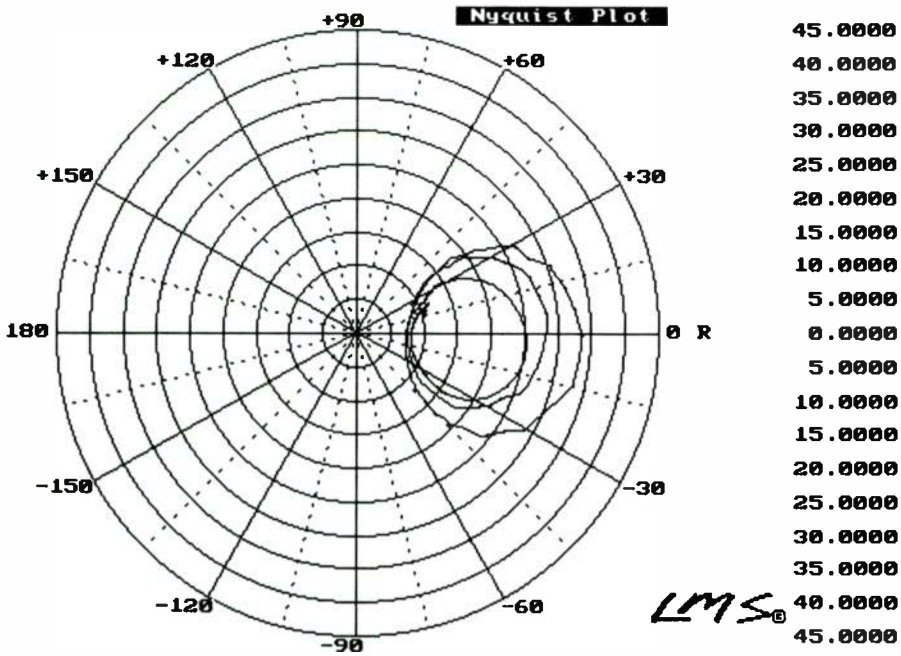


FIGURE 28: SL280 complex impedance plot.

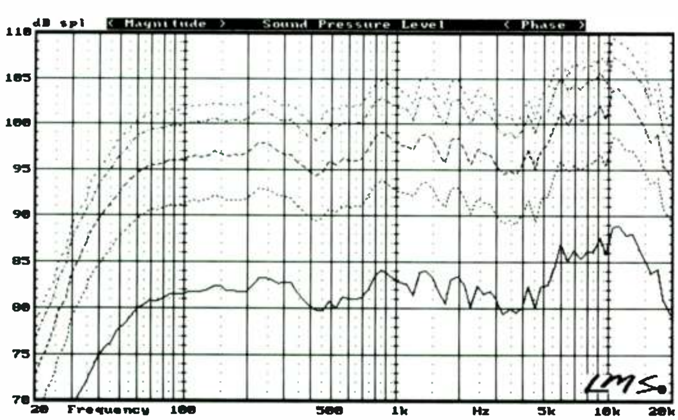


FIGURE 29: Dynamic full-range ground-plane frequency-response plot (solid = 0.9V, dot = 2.83V, dash = 5V, dash/dot = 8V, short dash = 10V).

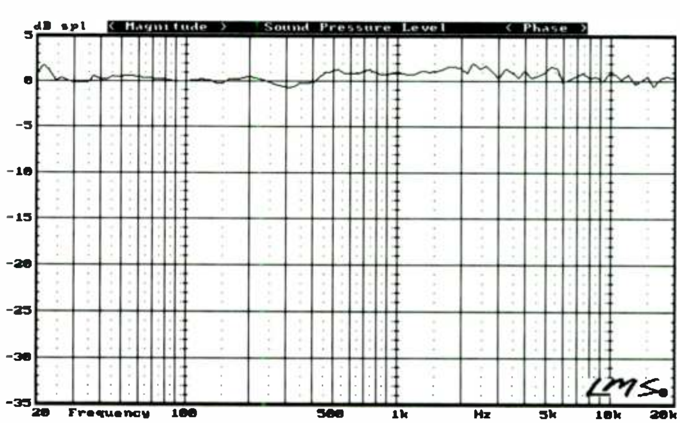


FIGURE 30: Difference curve for 10V and 0.9V curves in *Fig. 29* (10V/0.9V).

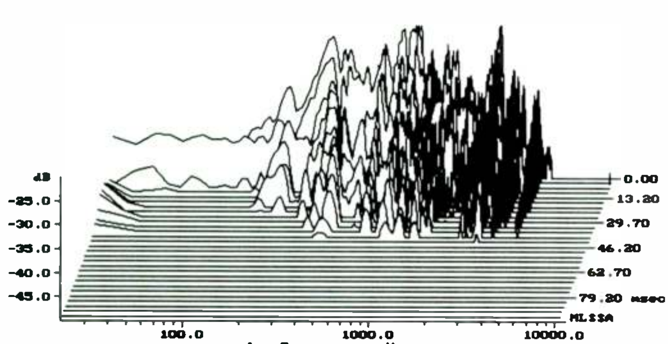


FIGURE 31: Accelerometer curve for SL280 (MLSSA waterfall plot).

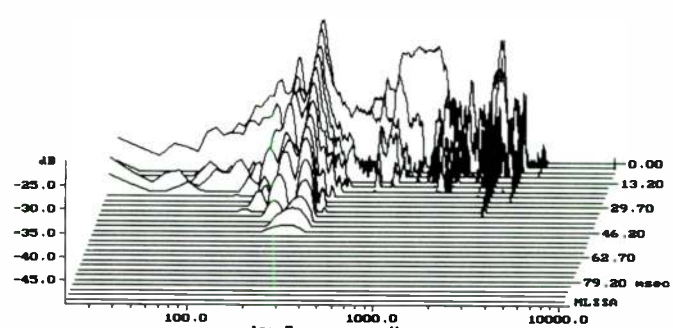


FIGURE 32: Accelerometer curve for author's prototype speaker (MLSSA waterfall plot).

FIGURE 33: Comparison of tweeter response with and without foam front baffle damping material at 0° (solid = with, dash = without).

FIGURE 34: Comparison of tweeter response with and without foam front baffle damping material at 15° off axis (solid = with foam, dash = without foam).

response with the tweeter connected electrically out of phase from the woofer. The out-of-phase null occurs almost exactly at the crossover frequency and reaches a depth of 20dB, indicating the drivers are for the most part in phase in the crossover region.

The physical delay between the woofer and the tweeter is only 110μs, or 1½". This is a small number for an 8" two-way, given the depth of the average 8" basket, and is the result of recessing the tweeter and not recessing the woofer, which removes an additional ¼" of delay between the drivers. I used this data to produce the magnitude and phase curve in *Fig. 21*. Since LMS does not measure phase, the graph's phase is calculated from the magnitude response. It is not a minimum phase curve of the system response, however, but rather the calculated phase of the low-pass section, with the added 110μs of delay mathematically corrected, summed with the calculated phase of the high-pass section. This process yields a very accurate total summed phase response—in most cases, more accurate than most test instruments can provide in a limited test environment.

The SL280 also produces a clean spectral decay curve, as illustrated in the *Fig. 22* MLSSA waterfall plot. The upper-frequency areas are attenuated quickly beyond 2ms, and are free from any major resonances. The match between the stereo pair provided for this review is portrayed in *Fig. 23*. The two speakers are very close and matched within 0.5dB except for a small area centered on 2.8kHz, where the deviation is 1.5dB (*Fig. 24*).

The SL280 grille frames are of injection-molded plastic and have a very low profile, somewhat less obtrusive sonically than the MDF types. The response with and without the grille attached is shown in *Fig. 25*, with the normalized difference curve in *Fig. 26*. Any significant deviation occurs primarily at 5.5kHz with a 5dB dip and a smaller 2dB deviation at 3.3kHz.

The impedance magnitude and phase are depicted in *Fig. 27*, along with the complex impedance curve in *Fig. 28*. Minimum impedance is 7.32Ω and occurs at 154Hz, while the maximum is 33.6Ω at 60.7Hz. The maximum capacitive load comes at a phase angle of −45.9° at 73Hz, and maximum inductive-load-phase angle of 38.4° at 51.4Hz. With low phase angles and high minimum impedance, this speaker should not be a problematic load for any amplifier. The dynamic range of any two-way speaker is generally limited by the tweeter. The SL280 is able to obtain a very good off-axis system response while crossing over the tweeter at 2.5kHz, which represents a nearly ideal situation for a two-way 8″. Looking at the full-range ground-plane measurements in *Fig. 29* and the difference curve from the 0.9V and 10V curves in *Fig. 30*, only minor changes are evident in response from a low-level signal up to nearly 102dB (96dB measured anechoically).

The SL280 cabinet is quite solid. The box is constructed of ¾″ MDF walls and incorporates a 1″ MDF front baffle. Bracing includes two horizontal types, one between the woofer and tweeter and another about 8″ from the enclosure bottom. Two pieces of egg-crate-type acoustic foam damp the box, one piece covering the walls and rear baffle directly behind the woofer, the other covering two side walls and the bottom. I used a piezo tweeter element accelerometer to analyze the effectiveness of the SL280's bracing and box construction.[3]

Results, using MLSSA, are shown in *Fig. 31* for the accelerometer placed in the center of the speaker's rear baffle, close to the lower horizontal shelf braces. For comparison purposes I have included a picture of the accelerometer placed in the same position on one of my own designs (*Fig. 32*), which is also a two-way 8″, but uses 1″ MDF for the walls and 1¾″ MDF for the front baffle plus an internal Sonotube™ baffle rear filled with sand—a very well-damped box. As you can see, the Signet cabinet is not as well damped as the other two-way 8″, but considering its price, the cabinet is very well executed.

The last objective data I took on the SL280 had to do with the unique foam damping material on the front baffle. If you have looked at the SL280 in a dealer's showroom (*Photo 1*), you undoubtedly noticed the die-cut foam placed around the tweeter with its distinctive daisy-cut pattern. *Figures 33–36* compare the tweeter response with and without the foam material at 0°, 15°, 30°, and 45° off axis. The primary effect

REFERENCES

1. *LMS Operations Manual*, LinearX 1993, 15–16.

2. L. Beranek, *Acoustics*, American Institute of Physics for the Acoustical Society of America, 1986.

3. Described in *The Loudspeaker Design Cookbook*, 4th ed., Audio Amateur Press, 1991, 138, Fig. 8.17. I have incorporated the peizo accelerometer into my work in favor of the PVDF device I described in "PSB Stratus Mini," *SB* (3/93): 48.

PREVIEW

Audio Amateur

Issue 1, 1994

- Line Stage Preamp
- More POOGE for Philips 960
- Borbely Amp Update

FIGURE 35: As in *Fig. 34*, at 30° off axis.

FIGURE 36: As in *Fig. 34*, at 45° off axis.

is off axis, and while not great, it does give the listener the subjective impression of a smoother-sounding tweeter.

SUBJECTIVE EVALUATION

The Signet SL280 is a well-designed loudspeaker and has received good marks from the review press. Its full-range musicality is about as good as it gets for a two-way speaker. The SL280B/U has enough low end not to sound like the typical anemic bookshelf, is reasonably neutral in spectral content, and has more than enough detail and definition to be considered a good value.

I agree with the *Stereophile* review of the veneered version of the SL280 done in October 1990 that the Signet SL280 is quite satisfying on a wide range of musical program material. If you are looking for a speaker to recommend (I know that *SB* readers listen only to their own designs!) in the $700/pair price range, the Signet SL280B/U is certainly worth considering.

MANUFACTURER RESPONSE

Many thanks to *SB* and Vance Dickason for the knowledgeable and detailed evaluation of the SL280B/U. We believe this model is a good example of our design approach: "affordable audiophile." Musicality and high value are our major concerns; we think they are not mutually exclusive.

As readers of *SB* know, and as this review clearly documents, speaker-system design involves selecting from among a great many interrelated factors. We are gratified that you find merit in the combination of design decisions and trade-offs chosen for this model.

Jon R. Kelly
Audio Potentials Corporation

Vintage Designs

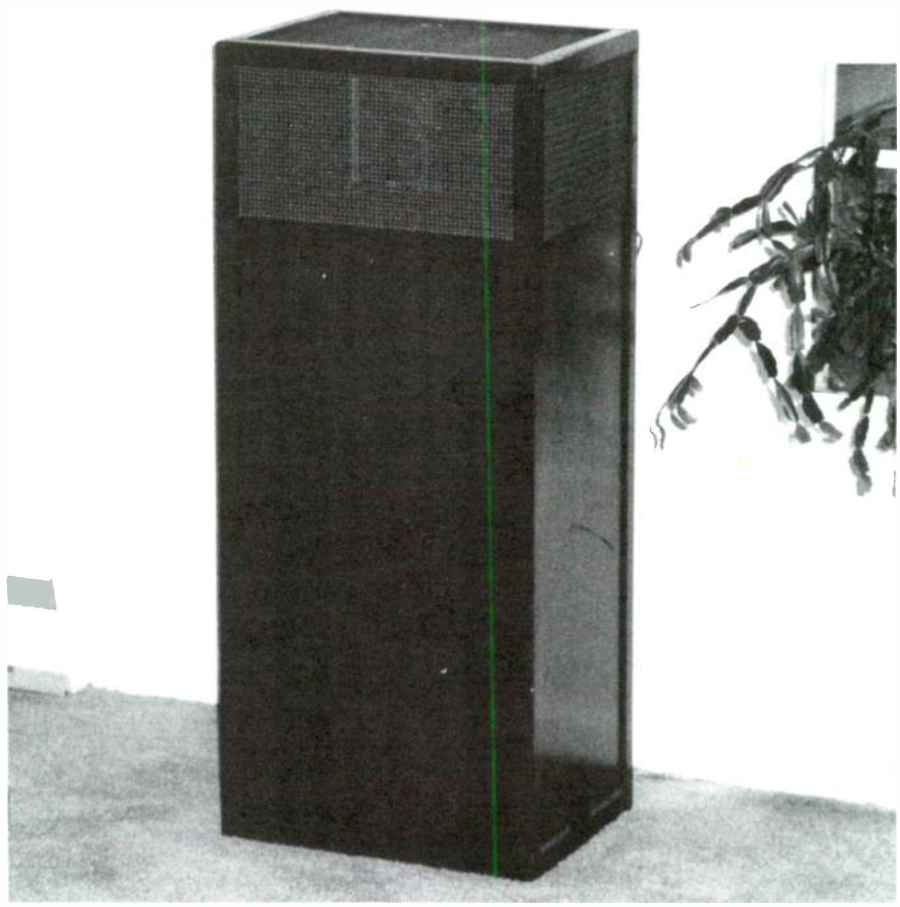
EICO Transmission-Line Enclosure, c. 1960

Courtesy of Jerome J. Klein, M.D., Norwalk, CT 06851

The photograph shows one of a pair of EICO transmission-line enclosures, c. 1960 vintage, which I bought in 1963 with burnt-out Norelco drivers. Over the years I have had many drivers installed. A vast improvement in woofer performance occurred when I installed Focal 8V416s per the three-part article on "The Brother Jon" by Robert J. Spear and Alex Thornhill ("A Prize-Winning Three-Way TL," *SB* 4/92, 5/92, 1/93).

The other drivers are a Wharfedale Super 3 used as a midrange—facing upward, open back—and an Audax composite titanium dome tweeter, all crossing over at 12dB/octave at 600Hz and 5kHz. The woofer in this enclosure faces upward. The enclosure suffers from an upper-bass peak; there is very little room around the woofer for sound-damping material, but it still sounds wonderful.

This highly popular and unusual design by Stuart Hegeman also featured a dual paper-cup-like diffuser mounted above the woofer. Like most of Hegeman's designs, it was innovative enough to attract the attention of Radio-Electronics. *It appeared on the magazine's September 1958 cover, earning kudos from J. Gordon Holt, who wrote in* High Fidelity: *"Eminently musical: would suggest unusual suitability for stereo." Eico claimed that what Hegeman had produced was "a slot-loaded, 12' split-conical bass horn." The unit, 3' high, almost 16" wide, and nearly a foot deep, sold for $139.95 in mahogany, with $5 additional for "blonde," in late 1958.—Ed.*



Ask SB

Robert M. Bullock III
Contributing Editor

CROSSOVER BLUES

In reviewing numerous crossover plans I have noticed many third-order networks that don't have the 3:1 relationship between the two inductor or capacitor values in the low- and high-pass filters that would be predicted by the Butterworth equations. Often the relationship is 2:1 or even 1:1. Could you explain the effect this has on the Q, phase, group delay, and so forth of the network in a typical two-way system?

Also, how are these same characteristics affected when a third-order high-pass filter is used in conjunction with a first- or second-order low-pass filter? What will be the polar tilt, proper polarity, response peak, and so on? Finally, are there equations for predicting third-order networks with a Q other than 0.7?

Robb Hayes
Newcastle, OK 73065

Robert Bullock responds:

There are many reasons why a third-order two-way crossover network may not satisfy the component ratio conditions of a Butterworth filter. Some of them follow.

1. The crossover network design may not be based on Butterworth filters. Sometimes designers base crossover networks on Bessel filters, Chebychev filters, or one of several other standard types.

2. The 3:1 component ratio you mention is correct only for filters whose load is constant. Many times designers alter component values in a Butterworth network because driver loads are not resistive, and the altered component values do a better job of providing the desired maximally flat (Butterworth) filter-response shape. Even when the driver loads are equalized by the use of zobels, the classical constant-resistance load requirement may not be met, and the filter component values must be adjusted to provide maximally flat response.

3. The standard Butterworth filter formulas are derived assuming that the circuit inductors have zero resistance. In practice this is not the case, so that component values often have to be adjusted to compensate.

If the crossover is three-way then the usual Butterworth filter design values are not correct even with resistive loads in all channels. In other words, maximally flat response of the crossover network is no longer provided by using Butterworth filters.

If the component values in a two-way crossover network are altered from their Butterworth values for any of these reasons, it is likely that its performance will more closely match that of the theoretical resistively loaded circuit. This is usually considered the desired goal, so the transient, phase, and magnitude response should be better.

If a first- or second-order low-pass filter is used in conjunction with a third-order crossover, the system performance will indeed be different. One or both of the channels will no longer be third-order filtered, but will be fourth- or fifth-order filtered. All other things being equal (which they won't be), the phase and transient performance will be degraded and the polar pattern will probably be altered. Whether the result is better or worse is impossible to say, because of the many variables involved.

You ask about "third-order networks with a Q other than 0.7." The notion of Q applies only to second-order filter sections, and the second-order Butterworth filter does indeed have a Q of 0.7. No such value can be associated with the third-order Butterworth filter, however. You can regard it as consisting of a first- and a second-order section in cascade and associate the Q of the second-order section to the filter, but then its Q is 1.

There are design equations for filters other than Butterworth. Bessel and Chebychev filters have design equations associated with them, for example, but the formulas are more complicated than the Butterworth formulas.

SB Mailbox

ERRATUM

My original Mitey Mike article ("Mitey Mike: For Loudspeaker Testing" *SB* 6/90, p. 10) contains a small but confusing error. On p. 16, under the section heading "Testing, Testing," all references to R5 should be changed to R3.

Joseph D'Appolito
Andover, MA 01810

HOT BUTTONS

Bill Schwefel's experiments as reported by Alan Towbin in the *Mailbox* section of *SB* 5/93 were interesting. Unfortunately, Mr. Schwefel's conclusion that "with any speaker system...the best sound will result when you lose the rear wave entirely" is unjustified from what we read of his tests. He apparently made no measurements—he just concluded subjectively, by ear, that the transmission-line and Isobarik systems he built had the least colored sound.

The problem is, we don't know what "colored sound" means to Mr. Schwefel. For example, the Isobarik and transmission-line enclosures almost surely had a lower Q at their low-frequency cutoff than did the sealed systems. Perhaps low-Q woofers sound good to Mr. Schwefel. Perhaps the sealed-box system sounded colored because of some cabinet resonance(s), because of an inadequate volume that a more careful design would have eliminated. Who knows?

This discussion brings me to one of my speaker-design "hot buttons." Most amateur speaker builders seem fascinated by transmission-line systems. Will we ever find even one article that gives a good analytical (i.e., mathematical) discussion of designing such systems along with careful experimental results to test the theory? The usual rule-of-thumb design guidelines and touchy-feelie listening tests shed no light on thissubject.

Tom Sharpe
Lexington, MA 02173

[Several authors have given serious consideration to TL design theory, including Robert Bullock. His design is available as TRANSMISSION LINE BOXMODEL from Old Colony Sound Lab. Larry Sharpe, of Mahogany Sound, offers a Lotus spreadsheet design for use with his Acoustastuf filling material that reportedly works well.—Ed.]

Bill Schwefel responds:

If my conclusion that "the best sound will result when you lose the rear wave entirely" is unjustified, then the rear wave must be regarded as something other than distortion.

But how can reflected energy that is out of phase with the initial wave front be regarded as anything other than distortion?

All design types make use of these reflections, but that is not to say that they should; it simply means that they must. Damping material can dissipate and control energy, but it cannot eliminate it. In addition to the phase problem, the rear wave will be altered as it bounces around inside and outside the enclosure.

In my experience, low Q_{TC} woofer systems generally sound better. For this reason, poorly designed drivers with small magnets and high Q_{TS} measurements are seldom used. In my test, I used a pair of Radio Shack drivers that had a single, in-box Q_{TC} of .55. This is a careful design of adequate volume.

KNOCK 'EM DEAD

I've been unhappy with the likes of the Focal Aria 5, Celestion SL600, and so forth with male quartet music, including my own DAT tapes of my quartet. And as a bass player, I'm well aware of the easy sound quality of open-back cabinets versus ported or sealed systems. I've been trying to find my copy of G.A. Briggs's book in which he describes his design of an open-baffle system Wharfedale sold commercially after World War II—so you can imagine how happy I was to read Joseph Janni and Warren Hunt's article "A Full-Range Open-Baffle System" (*SB* 1/94, p. 30).

I'm interested in your further notions as to utilizing a 15" or even 18" subwoofer. After a quick look at your passive low-pass correction circuit, I think the 20kΩ input R of my pair of Aragon 4004s won't dramatically alter what goes on. My Aragon 24k preamp output R is quite low—I'll put the filter together—I may have to change C1—Why is C1 there?

Briggs wrote of the advantages of dissimilar (10" and 12") woofers on an open baffle in parallel connection. He advised "a little bass lift" to compensate for baffle size, which if my memory is correct was about 30" × 34".

I have test gear, scopes, THD, microphones, and so on, so can go along as time permits me. Thanks for the great article.

Roger Cox
Chino Hills, CA 91709

Joseph Janni and Warren Hunt respond:

We are pleased that you enjoyed our article, and agree with you that a properly designed open baffle usually sounds better than ported or sealed enclosures. We believe it is because the internal reflections and timbre degradations due to "cabinet talk" are minimized in an open-baffle design. We also think the acoustical radiation pattern of an open baffle minimizes some undesirable wall reflections and allows the music to be heard more accurately at the listening position.

*The low-pass correction circuit (*Fig. 4 *in our article) this circuit can easily be matched to almost any preamplifier or amplifier. The preamplifier-matching part of the circuit depends only on the sum of R1 and R2, which must be 4.9kΩ. As an example, if R1 were to be increased to 2kΩ by using a different preamp, then R2 would simply have to be decreased to 2.9kΩ to maintain a total resistance (R1 + R2) of 4.9kΩ to assure correct circuit performance.*

We probably should have explained that C1 has two purposes. First, it provides DC isolation between the preamp output and the amplifier input. (This isolation is usually not necessary, but we provided it as a good design practice.) Second, C1 forms an RC network with the amplifier-input impedance R4 so that extremely low frequencies below audibility are rolled off, at least to some extent. The value of C1 depends only on the amplifier-in-

put impedance R4. If an amplifier is used with an input impedance lower than ours (100kΩ), such as yours with 20kΩ, then C1 must be increased from 0.22μF to 1.1μF. The key point is to always select C1 so that the product of C1 and R4 is constant. If R4 decreases by a factor of 5, as in your case, then C1 must be multiplied by a factor of 5. It is also a good idea for R4 to be at least ten times larger than R3. Since R3 is 1.5kΩ and your R4 is 20kΩ, this criterion is easily met and the circuit should work well.

We have Briggs's book Loudspeakers, The Why & How of Good Reproduction *(4th ed.), but can't find the sections or comments you mention on open baffles. Perhaps they are in his other book,* Sound Reproduction.

We encourage you to build our open-baffle design. We put a lot of effort into it, and everyone hearing the Tombstones says they sound great. One individual said, "The Tombstones really rock!" and another said, "They knock you dead!" We have made some improvements since the article was published, which we want you to know about before you begin. Be sure to adhere closely to our design—we want you to obtain the same excellent results that we did.

We replaced the Audax HD13D34H cloth-dome tweeters with SEAS 25TAC/D H535 6Ω 1" aluminum dome tweeters primarily because they have a resonance—660Hz—much lower than the 900Hz resonance of the Audax tweeters. The resonance of the tweeters should be as low as possible in our design, and we were unaware of this specific SEAS tweeter when we made our first tweeter selection. After listening to the SEAS we think they have a slightly more pristine sound, probably because of the aluminum dome performance. Their sensitivity is 91dB/W/m instead of 92.5dB/W/m, but the 1.5dB drop actually provides a slightly better high-frequency match to the midranges (as you can see in Fig. 1 of our article). The crossover capacitor and zobel inductor values did not need to be changed.

Our article mentioned that we were considering replacing the Polydax HD30P45TSM 12" drivers with 15" drivers. After comparing specifications and prices for more than 20 different 15" drivers, we chose the Madisound 15258DVC. We enlarged the baffle subwoofer holes, mounted the new 15" drivers, and wired the voice coils in parallel so that they would be better matched acoustically to the midranges. Because of the interaction between the subwoofer and the resonant crossover below 70Hz, the impedance still does not drop low enough to be a problem for a good solid-state amplifier.

The resonant crossover had to be changed, so we experimented with several different capacitor and inductor values. We measured the output with warble tones and listened carefully to the effect of each component change. Finally, the inductor remained unchanged at 18mH, but we doubled the capacitor to 660μF. We observed no real improvement in the sound quality by replacing the 12" subwoofer (which was an excellent driver) with the 15" subwoofer, with one exception; the system can now play even louder than it could before without showing signs of subwoofer strain. We like that.

MIKE'S MEASUREMENTS

This letter is prompted by Gary Galo's recent review of Mitey Mike ("Three Affordable Measurement Microphones," *SB* 4/93, p. 70). We have about two years' experience now with Mitey Mike kits and with the calibration of Panasonic mike cartridges in particular. Having calibrated approximately 100 cartridges at this point, I'd like to pass along a summary of our results, and some recommendations.

First, to update Gary's article, calibrations are now performed using an ACO Pacific 7012 condenser microphone with an ACO 4012 preamp and PS9200 precision power supply. The 7012 is a ½" precision laboratory microphone with a measured free-field frequency response that is flat within 0.5dB from 20Hz–30kHz.

The Mitey Mike cartridge calibration is a normal-incidence, free-field calibration. The calibration is accomplished by comparing the response of an acoustic source (e.g., a loudspeaker) measured with a Mitey Mike cartridge against the ACO-measured response of the same source. The difference between the two response measurements is the cartridge response error.

Because differences in microphone shapes and diameters cause differences in off-axis response, the response comparison is valid only under free-field, nonreverberant conditions (i.e., single plane-wave incidence). Reverberant off-axis data must be eliminated from the measurement. This is accomplished with the MLSSA system, which uses only the response data before the first reflection; which in my setup comes from the floor. Because the Panasonic cartridge is omnidirectional, the calibration should be accurate to within the Mitey Mike specification limits of ±2dB at angles up to 30° off axis.

All of the Panasonic cartridges tested have a response peak around 12kHz. Over the 100 cartridges calibrated to date this peak ranges from 2–5dB and averages a little over 3dB. *Figure 1* shows a typical calibration response curve. Calibration data below 1kHz is not provided because the cartridges are essentially flat below that frequency.

Current Panasonic production is such that most of the cartridges tested do not meet Mitey Mike's stated spec of ±2dB from 10–20kHz. Assuming an individual cartridge is close to the average, you can use *Fig. 1* to correct Mitey Mike response to within its specified accuracy of ±2dB. You can also bring Mitey Mike into spec by changing C6 from 68pF to 330pF. With 68pF, the 548 op amp has an 80kHz, –3dB response point. The 330pF cap pushes the –3dB point down to about 22kHz and puts the overall response within the ±2dB envelope. If you desire greater accuracy than this you should use the calibration service. The electret mike cartridges are very stable so the calibration data will be good indefinitely.

We've had a small problem with reflections off the Mitey Mike case, which cause response ripples of a few tenths of a decibel between 2 and 6kHz. This is not a problem with warble-tone testing, because the warble averages out the ripple. The ripples are noticeable, however, with high-resolution systems such as MLSSA and IMP. You can use the Sonex foam solution described in Gary's article.

To eliminate the case reflection problem, we have developed a version of Mitey Mike with a 6 wand and an attached 6 flexible cable. Of course, you can still get reflections off any structure used to support the mike.

Joseph D'Appolito
Andover, MA 01810

MLS FOR MACS?

As an amateur speaker builder, I am certainly grateful to Bill Waslo and *SB* for having brought to this community a useful measurement device like IMP. I am even more favorably impressed by the related MLS module.

I would really like to see this same device offered for Macintosh computers. This would probably require an I/O stage redesign to fit the Macintosh ports, and surely a rewrite of the existing PC code—which are, of course, major issues.

But I find it unthinkable that such a good and useful tool isn't accessible to the number of people owning Macs or Amigas (which incidentally have built-in, although limited, sound capabilities) just because its I/O stage was designed around a Centronics port instead of a serial RS-232. I hope there will be the opportunity to address this little inconvenience in the near future, so that Mac users like me will be able to make affordable loudspeaker measurements.

Silvio Sancese
65121 Pescara, Italy

Bill Waslo responds:

Thanks for your comments on my IMP system. I'm glad you find it useful and I appreciate your support.

I'm afraid that a Macintosh version of IMP would indeed be a major undertaking. One attractive feature of using a parallel printer port for an interface bus is its sheer simplicity. The lines are basically TTL compatible (with a little help) and are easily manipulated. Additional interface hardware would be required on the IMP board to support the serial ports, and such operation would greatly slow down data transfers as well.

Another factor that has impeded development of a MacImp is the apparently small number of Mac users in the technical community. For better or worse, most hardware hackers (computer or audio) seem to be more attracted to the open architecture of the PC than to the Mac. The PC seems more friendly to the introduction of foreign wires, boards, chips, and so forth and the Mac's intuitive user interface and icons don't mean that much to those who work at the volt or bit level—so I'm not sure that would be enough MacImp users to justify the expense and effort. But if I get a deluge of letters to the contrary, I could be convinced otherwise.

TIME-DELAY RELAY

Kindly tell me where I can buy the time-delay relay with solid-state adjustment timer—Syracuse TIR115A1002, referred to in Mark Seymour's article "An Evolving Magnepan MG-1" (*SB* 1/94, p. 44).

John Bediako
Corona, NY 11368

Mark Seymour responds:

I purchased the Syracuse relay more than ten years ago from Hughes-Peter, a distributor of electric/electronic components for industrial applications. If the Syracuse TIR115A1002 is no longer in stock, I'm sure other devices exist that will perform the same function.

Please make sure, however, that the relay contacts are rated for the current demands of your power supply. The use of a 10 surge-protection resistor limits the peak current to under 12A: or about 10A when we factor in the impedance of the transformer itself. The relay used in my setup had two sets of 10A contacts, which I wired in parallel to give me 20A overall.

Also make sure you choose a timer operated by the 120V AC line source. This overcomes the need (and space required) for a separate power supply and timer.

Finally, ask about the time sequence. You want the contacts closed in the "Power Off" mode and to remain closed until nominally 100 (20) seconds after the power (120V AC) is applied ("Start" position), at which time the contacts remain open until the power is switched off. Then the device must return to the "Before Start" position (closed contacts).

SPEAKER DAMPING REDUX

I thank Gary Galo for his answer to my letter on speaker damping, which appeared in *SB* 6/92. Mr. Galo wrote (in part): "If the loudspeaker is connected to a high source impedance, the cone will make a great deal of extraneous motion. If the source impedance is low, the loudspeaker motion will more closely mirror the signal from the amplifier, with minimal extraneous movement."

To which I replied (in part): "This is true, but the implications are wrong. Consider the loudspeaker voice-coil resistance in series with the source impedance. This resistance is several ohms; it makes no significant improvement to lower the source impedance below an ohm or so."

To which he responded (in part): "Mr. Crawford raises an interesting question: whether the loudspeaker impedance is in series with the amplifier source Z or parallel with it. This, in turn, raises the question of whether or not the voice coil's DC resistance must be added to the source impedance when computing damping factor. A search through various reference books raises some contradictions, adding to the confusion."

Some of the confusion, I suggest, arises because there are two damping factors to consider: that of the loudspeaker and that of the amplifier. It is the damping of the loudspeaker that governs the ability of the amplifier to control the extraneous motion of the loudspeaker. However, the amplifier-damping factor can affect the loudspeaker damping.

In a loudspeaker we speak not of damping factor, but of the Q. To retain this terminology is wise, for two reasons: the loudspeakers are specified in terms of Q by the manufacturer, and the definitions of loudspeaker damping and amplifier damping are different. The lower the Q the higher the loudspeaker damping. Q_{TS} is the combination of the electrical Q (Q_{ES}) and the mechanical Q (Q_{MS}). Usually Q_{ES} is more significant than Q_{MS}. Small gives Q_{ES} as:

$$Q_{ES} = [(2)(\pi)(f_S)(R_E)(M_D)] / [(BL)(BL)]$$

R_E in the above equation is the total series resistance in the loudspeaker circuit. This includes the loudspeaker voice-coil resistance, the series resistance of the crossover network (if any), the resistance of the speaker wires going from the amplifier to the loudspeaker, the resistance of any speaker fuses in the circuit, and the output resistance (impedance) of the amplifier. Manufacturers specifying a Q_{ES} for a loudspeaker usually assume that the only resistance to be considered is that of the voice-coil winding, and that all other resistances are 0. The voice-coil resistance of an 8Ω loudspeaker is usually about 6Ω.

Mr. Galo quotes several sources to the effect that the amplifier-damping factor is the nominal load impedance (usually 8Ω) divided by the output impedance of the amplifier. With most modern-day amplifiers the damping factor is large (greater than 16), because the output impedance of the amplifier is less than 0.5Ω.

Let's calculate the effect of the amplifier-damping factor on the loudspeaker Q_{ES}. Assume that a loudspeaker with a DC resistance of 6 has a Q_T of 0.707 when driven from a zero-impedance source (infinite damping factor). Assume that the amplifier we are using to drive this loudspeaker has an output impedance of 0.5Ω (damping factor of 16), that we are biamping the speakers (no passive crossovers), and that we are using perfect speaker cables with no resistance at all. The Q of the loudspeaker when driven by the above amplifier increases to 0.766. The difference of Q of from 0.707–0.766 is the difference made by the amplifier-damping factor going from infinity to 16.

This is a small difference: smaller than the manufacturer's tolerance in making a batch of these loudspeakers; smaller than the change that occurs when the voice coil heats up. It is smaller than the changes wrought by a few years' aging of the ceramic magnet—smaller than the changes that would result from decreasing the speaker magnet size by 10% or winding the speaker voice coil with one size smaller wire.

The message is plain: improving the amplifier damping factor from 16 to infinity improves the loudspeaker damping by only a few percent. Thus, a large improvement in amplifier-damping factor contributes only a small improvement in the ability of the amplifier to control the extraneous motion of the loudspeaker cone.

Don't get me wrong. I'm all in favor of biamping, and I'm all in favor of using amplifiers with high damping factors. Let's use every trick in the book to get as much as we can from these speakers. But let's be aware how much—or how little—we can gain from each trick.

Dick Crawford
Los Altos, CA 94024

SONIC BOON

Marc Bacon's excellent article "Sanctuary Sonics" (*SB* 1/94, p. 12) was a great treatment in doing real justice to worship music. Having worked extensively with both home and church music systems, I know firsthand how bad most public-address speakers are. Mr. Bacon's report raises only one question in my mind: the Focal tweeter. Although it certainly has excellent sonics, I wonder whether its power-handling capability is up to the task. One good burst of acoustic feedback will vaporize most home hi-fi tweeters in a second.

In a system I built several years ago, my solution was to use six Audax TW51A tweeters in phased-line array, as shown in *Fig. 1*. Alternately, as most readers probably know, the Morel MDT28 is pretty rugged. VIFA also has some very good horn-loaded domes with high-output capability.

Perry Sink
Berwyn, IL 60402

Marc Bacon responds:

Thank you for your kind comments regarding my article. I totally agree with you—Morel tweeters can take more power, and VIFA

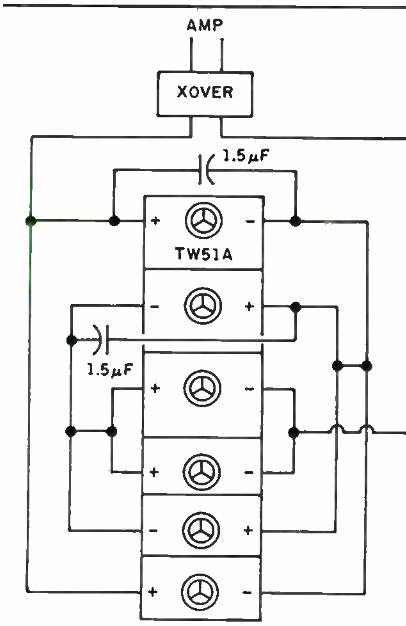


FIGURE 1: Tweeters in phased-line array.

horn-loaded tweeters have higher sensitivity. Your line array is another way of handling higher power and controlling vertical dispersion. For even more output, you could try the Polydax bullet tweeters.

I chose the Focal T90TiO2 tweeters for their sonic neutrality and wide dispersion pattern, which results from the concave titanium dome. Use of ceiling-mounted low-level speakers requires far less power than two high-SPL units in front of the auditorium, and the fourth-order L-R (Linkwitz-Riley) crossover characteristic provides adequate protection up to 75W music power/unit, or a total of 600W for the eight units. With 200W available from the power amp, the Focal tweeter performs very well. For those who want to get the sound quality available from Focal drivers with higher power, consider the T120, T122, or T130 series.

The little drivers are actually quite rugged, as an interesting anecdote demonstrates. After installation of the system, a construction worker accidentally separated the lead-in wires from one unit, and (Murphy's Law operating well) hooked the woofer up to the tweeter input from the main crossover and vice versa. Other than a horrible buzzing on low notes, the tweeter lived through a test at nearly full volume by a console operator!

In my opinion, the main destroyers of hi-fi tweeters are:

- *Use of an undersized amplifier, which generates harmonic distortion at high levels.*
- *Choosing too low a crossover point or slope.*
- *Playing pure test tones through the tweeters. A friend of mine destroyed two Accuton tweeters instantly in that fashion.*

HORNS APLENTY

I used Bruce C. Edgar's articles "The Show Horn" (*SB* 2/90, p. 10) and "The Monolith Horn" (*SB* 6/93, p. 12) to write a software program for calculating the various horn parameters. With the exception of V_B, my calculations agree with yours for the JBL driver in the Monolith article, but I am unable to duplicate several of the results you achieved in the Show Horn article. And I have some other horn-related questions.

1. On p. 12 you provide two formulas for computing S_T. I can make the results of these formulas equal only if I multiply the result of the first formula by 144. I believe your first formula produces a figure measured in square yards.

2. On p. 13 you give a formula for computing α. I then calculated V_B using V_{AS}/α. Is there another way to calculate V_B and then derive α from V_{AS}/V_B? Using the formulas provided, I cannot derive the V_B values of 1,227 and 1,197 that you got on p. 14.

3. The question above on α may also be

related to this question concerning the calculation of M. In the discussion of the M parameter you mentioned that a flare rate anywhere near the driver resonant frequency will result in an M value of 0.5. Using the formulas provided for both α and M, and substituting any value of f_O into the M formula (changing no other variables), I always get an M value of 0.503. Any insights?

4. You seem to prefer rectangular cross-sections for both throats and mouths. Are they better than square? How do you decide on the length-to-width ratios for throats and mouths?

5. Is there an optimum throat size for a given driver-radiating area, or is calculated S_T optimum by definition?

6. Is there a formula for calculating optimum f_O for a given driver, or is it a design choice?

7. I think fiberglass or other material in the back chamber would be unnecessary within the low-frequency region where bass horns work. Your thoughts?

I am thinking of building a bass horn for my next project, followed by a midrange horn. I am not certain why I want to, except that I am intrigued by horn design, and it is a type of speaker I have yet to build. When I have completed something, I will send you information on my project. I look forward to your future articles on horns, and would especially like to see something new or an update on a midrange design.

Craig Berndt
Bloomington, IN 47404-2155

Contributing Editor Bruce Edgar responds:

To answer your questions:

1. The first formula ($S_T = 2pf_S Q_{ES} V_{AS}/c$) is the general form. If the V_{AS} and c terms are in cubic feet and feet/sec, respectively, then S_T comes out in units of square feet. In the second form ($S_T = 0.8 f_S Q_{ES} V_{AS}$), I stated in the article that V_{AS} has units of cubic feet and S_T has units of square inches. The factor of 0.8 comes from combining the sound speed (c) with the conversion factors (changing cubic feet to cubic inches). I use this formula for S_T because most US specification sheets give V_{AS} in cubic feet and I need to know S_T in square inches, not in square feet.

2. The α factor has caused nothing but confusion, and I'm sorry I ever brought it in. And Leach's assumptions in using are somewhat suspect. The back-chamber formula I use now is:

$$V_B = V_{AS} / \frac{F_O}{F_S Q_{ES} M^{-1}}$$

3. The result you mention of M = 0.5 is one of the reasons I suspect Leach's formula is misapplied to horn design. You should be allowed to use M as a free parameter to choose the low-end rolloff and adjust V_B to resonate the system and annul the throat reactance. M = 0.5 is an acceptable choice, but M = 0.6 or 0.7 is also acceptable. So a designer needs a wider range to choose design parameters.

4. My mouth and throat sizes are usually determined by folding considerations and the size of the driver. On straight horns I always use a square throat.

5. No, the optimum throat size is not related to driver size, but it is inversely proportional to mass rolloff frequency. For 12"–15" woofers with a low rolloff frequency (100Hz) the optimum throat size is usually comparable to diaphragm area. On drivers with high-mass rolloffs (400Hz), the optimum throat size is less than 25% of the diaphragm area.

6. I don't know of a general formula. For best results, I have found, the resonant frequency (f_S) of a driver needs to be within 10–15Hz of the flare frequency f_O.

7. Fiberglass in the back chamber allows you to adjust the resonance of the chamber, but the most frequency shift you can expect is only about 2–3Hz. A small amount of fiberglass does help damping in the back chamber without shifting the chamber resonance frequency significantly.

Finally, I encourage you to experiment with real horns. I spent several years working out designs on paper, but I didn't start to make real progress until I built real horns and measured the response. Then you learn the things that aren't mentioned in texts or articles.

Multitap Inductor

Continued from page 18

matter how neatly the first few layers are wound you'll find the windings becoming increasingly random. You may want to seal all the pin holes you've made when taking measurements. Either seal as you go or dip the whole thing (*not the barrier blocks*) in an enamel varnish after all the winding and terminating are completed.

Minimonitor Upgrade

Continued from page 38

At $90 each, the Elac tweeters are a sizable investment. I've tried the exceptionally smooth SEAS 25TAF/D aluminum-dome tweeters in other systems and consider them a good inexpensive alternative. They're not rated for use with such a low-crossover frequency, so you'll have to see whether they have sufficient power handling for your needs. Also, the crossover I've given is intended for use with the Elac tweeters, which have a near-constant 8Ω impedance. Fortunately, you can get similar results by padding the 6Ω (after break-in) SEAS tweeter down by about 4.5dB while maintaining an 8Ω load for the crossover. Simply replace the series 2Ω resistor in *Fig. 1* with a 4Ω value.

If you build this system from scratch, consider using a Focal 6K412L or 7K415 woofer in a larger, vented enclosure. Like the Dynaudios, these use a "sandwich"-technology cone to maintain pistonic behavior, but should allow greater low-frequency power handling. I haven't tried this, however, and would recommend a complete crossover redesign.

BUF 124 and PSpice

Continued from page 44

SPICEtalk), but that was way off in my simulation: the I_{DSS} were 2.11 and 3.59mA. The I_{DSS} can be computed from the J model by squaring VTO and multplying by BETA. If you want to try a matched pair, my professorial colleague who is expert in these matters tells me to change BETA, not VTO. That's because BETA is a function of silicon area (difficult to control) while VTO is a function of silicon thickness (relatively easy to control).

The evaluation PSpice comes with an excellent book on the subject: *SPICE: A Guide to Circuit Simulation and Analysis Using PSpice®* (Prentice Hall), by Paul W. Tuinenga. This book-and-software package is available as #BKPH2/S from Old Colony Sound Lab for $35.95 plus $3 S/H in the USA. (Software is available for the IBM PC, the IBM PS 2, or the Macintosh II—please specify.) Another new beginner's book, which also teaches a lot about elementary circuit analysis, is *PSpice with Circuit Analysis* (Macmillan 1993), by Franz Monssen. I also recommend Muhammad H. Rashid, *SPICE for Circuits and Electronics Using PSpice* (Prentice Hall 1990).

Don't forget: the evaluation PSpice is limited to ten transistors, so don't expect to analyze any more than a third-order Sallen-Key stage.

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Matched MOSFETS for Pass/Thagard A75 amplifier available. Set includes all 37 (1 spare P-channel and N-channel output) IRF devices. $175(US) per channel plus shipping. Deadline for receiving your order is June 1, 1994. Anyone interested, please contact Nick Mastrobuono, 2086 Lakeshore, Sarnia, ON, Canada, N7T 7H6, (519) 542-0964.

Transmission-line D'Appolito 5¼s with Vita aluminum dome, 6" wide, 40" high, matching 12" dual-voice coil subwoofer, nice sound stage, beautiful finish. Designed by Larry Hunsicker of Mariah Acoustics. Good clean dynamic. $450 plus shipping. (512) 926-2769.

Altek Model 14, $340 pair; pair KLH 17, $120; Onkyo A7 integrated, $95; Pioneer SA8500 int., $150; EV 16Ω SP12, 16Ω T35, 80Ω 1824m drivers with 8 HD horns, trade for variable electronic crossover. D. Parsons, 71 E. West Hill Rd., Winstead, CT 06098, (203) 379-6628, 6-10 PM EST.

Sgt. Pepper UHQR (sealed), $175. Select Goldbug Brier MC cartridge with matching wood-grain headshell (list $500), $350. Monster cable ALPHAI (list $500), $175. Both never out of box. LUXMAN MQ68C 30W tube amp with extra tubes (list $800), $300. L. P. Wilkins, 550 Baker, #6, San Francisco, CA 94117, (415) 567-4178.

Pioneer Spec I preamp, excellent, $200; Spec I, scratched face, $100; Eumig preamp, $150; Audire amp, $250; Audire preamp, $125; Technics receiver, $150; three Bozak pro speakers, each 4" × 4", $50 ea.; pair JBL 46120K speakers, $425; pair JBL 2425J, $250; JBL horns/lenses, $150. David (914) 688-5024.

Matching pair of vintage KEF speakers model K-2 Celeste MKIIs made in England, sold by Radio People Ltd, Kowloon, Hong Kong, with matching serial numbers, size 6½" × 18" × 11" collector units in good shape, clean, solid wood, rare, good sound, perfect condition, call now $225. I pay shipping. (305) 931-7131.

Two Dyna MKIII chassis with cages, no mounted parts, exc., $85 plus shipping; two Sylvania 6550 very low hours, matched pair, 1½" bottle, $40; new CDE Sangamo low ESR caps, ten 15K/75V, $10 ea.; four 42K/90V, $25 ea.; seven 22K/30V, $7 ea.; two Tungsol 6550 coke bottle, long hrs., test good, $30. Mike, (708) 682-8172 (IL), machine after 4th ring.

Sony CDP-X777ES CD player, mint, very low hours, original box and manual, balanced outputs, coaxial and optical digital outputs, Stereophile Class B, also an awesome transport, built like a tank, 40 lbs, absolutely spotless, $1,900 list, asking $800. (212) 980-2826, after 7 PM (NYC).

University C15W dual-impedance woofer E.C., $75; University N-2A x-overs, $25/pr.; N-2B x-overs, $25 pr.; University H-600 horn with SAHF driver, $35; Focal 8V416 drivers, $110 pr. All plus shipping. Jim (708) 425-6719.

DEX MC phono preamplifier, new in box, $55; Kenwood KM107 150W/channel stereo basic amp., $225; Yamaha B6 "The Pyramid" 200W/channel stereo basic amp., $325. All plus shipping. Stan, (216) 288-9480.

18" Orevox essence WC18125 woofer, mint, see *Speaker Builder* 5/92, pp. 64-65. $50 plus UPS. John Bundy, 6 Aspen St., Etna, NH 03750, (603) 643-5567.

Speakers: Sanders 4-cell ESL, BIAS, Xfmrs, crossovers, equalization, $700; Audiostatic ES240, $350; 10" Isobarik subwoofers, $200 pr. Drivers: Dynaudio 6" × 12", four ok, two need rebuild, $225; rebuildable carved, maple F-hole guitar, $75; electric guitar part set: Schaller gold tuners, two humbuckers, bridge, ebony board, rod, $100. (505) 783-4551 before 9 PM MST.

Selling my personal collection of Dynaco tube & transistor gear. SASE for 3-page list. No calls for list. B. Brown, RR 1, Box 50, Valley Springs, SD, 57068.

Bruel & Kjaer 2215 sound-level meter with 4165 mike, tripod, extension cable, and case. Bruel & Kjaer 4130 calibrator. $1,500 for all. Neil, (703) 533-0717, days.

Speaker Builder magazines, complete set '81 through '93, $195 or will swap for tuner, receiver, drivers, etc. Call Vincent (207) 865-1733.

NESTROVIC Model 8 subwoofers, oak, $800/pair plus freight or best offer. Excellent LF response, current model is $5,000/pair. Van Alstine Delta 250 power amp, nice, recent upgrades, $350. (206) 643-6520 (WA).

CLASSIFIED INFORMATION

Four types of Classified Advertising are available in *Speaker Builder*:

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TRADE: For any business or private party selling equipment, supplies or services for profit.

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Classified Department
PO Box 494
Peterborough, NH 03458-0494

A75 matched MOSFETS are not available through Old Colony or Pass Laboratories. I'm willing to supply these parts if sufficient number of people are interested. Quantity purchase and matching result in cost savings passed on to you. Anyone interested, please contact Nick Mastrobuono, 2086 Lakeshore, Sarnia, ON, Canada N7T 7H6, (519) 542-0964.

Four Scan-Speak 21W/8553, $50 each. Pair Scan-Speak D2905 tweeters, $35 each. BES SM275 speakers, true dipoles, $300. DBX 3BXIII range expander, $250. 15" × 15" sheets of eggcrate foam, $2.50 each. All items in excellent condition. Nick, (209) 583-6511.

CARVER amplifier, brand-new model PM 1201, 600W rms/channel @ 4Ω, 450W rms/channel @ 8Ω, 1,200W rms bridged, never used, $975, two available. PM900 amplifier, 450W rms/channel @ 4Ω, 350W rms/channel @ 8Ω, 900W bridged, $799. Three-year warranty. Bill, (215) 863-8424.

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Nearly every advertisement has a corresponding reader service number. Use the reader service card in this issue to request more information. *Please mail all cards within 60 days of receipt of magazine.*

WANTED

Service and/or operating manuals for MARANTZ 2220B receiver and MAC MC60 monotube amp. L.P. Wilkins, 550 Baker, #6, San Francisco, CA 94117, (415) 567-4178.

Need one Audax/Polydax MHD12 P25FSMSQ midrange with or without chamber. Jim Sanford, PO Box 222, La Conner, WA 98257, (206) 466-5161.

Manual and schematic for Amber 70, repairs and updates; source for wires to rewire tone arms, five-pin arm to preamp interconnects; PIB type cables with interface box. D. Parsons, (203) 379-6628, 6-10 PM EST. 71 E. West Hill Rd., Winsted, CT 06098.

Dyna Stereo 400, working or not, reasonable—or Stereo 150 with meters as above. Pair of Mark III, cheap. B. Brown, RR 1, Box 50, Valley Springs, SD 57068.

One pair Jordan modules, Heil tweeters. Larry Evers, (217) 732-6486.

Electro-Voice 8" speakers, model SP8B or SP8C. Also any manuals or information on JBL 15" Model D130 and JBL 10" model LE10A speakers—will pay copying costs, postage. Fred Sutton, 1230 S. Helberta, Redondo Beach, CA 90277.

Looking for information, schematics, hardware, manuals, etc. for Acoustat Servo-Charge tube amps. I'm interested in other ESL direct-drive designs too. I'll reimburse for photocopy charges and postage. Call first. Mark Nelson, (414) 425-2315.

WANTED: PRO-SOUND HOBBYISTS OR CURRENT EMPLOYEES IN SOUND REINFORCEMENT FIELD to correspond with USAF serviceman in England. Looking for equipment sources and contact with anyone who has a passion for quality reproduction of live music. Rick Diaz, PSC 41, Box 6912, APO AE, 09464.

CLUBS

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.

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.

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.

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

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.

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, USA.

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.

AUDIO SOCIETY OF MINNESOTA Now in its 15th consecutive year! Serving the many and varied interests of audiophiles in the upper Midwest. Monthly meetings, tours, audiophile concerts, special guests. For information and a sample of our latest newsletter, write ASM, PO Box 32293, Fridley, MN 55432, or call our 24-hour "Audio Hotline," (612) 825-6806.

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.

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.

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

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

LONDON LIVE D.I.Y. HI-FI CIRCLE meets quarterly in London, England. Our overall agenda is a broad one, having anything to do with any aspect of audio design and construction. We welcome everyone, from novice to expert. For information contact Brian Stenning, 081-748-7489.

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.

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

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.

PACIFICNORTHWEST AUDIO SOCIETY (PAS) consists of 60 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 Bob McDonald, (206) 232-8130 or Nick Daniggelis, (206) 323-6196.

THE ATLANTA AUDIO SOCIETY is dedicated to furnishing pleasure and education for people with a common interest in fine music and audio equipment. Monthly meetings often feature guest speakers from the audio manufacturing and recording industry. Members receive a monthly newsletter. Call Chuck Bruce, (404) 876-5659, or Eddie Carter, (404) 847-9296, or write A.A.S., 4266 Roswell Rd. N.E., K-4, Atlanta, GA 30342-3738.

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.

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.

SOUTHEASTERN MICHIGAN WOOFER AND TWEETER MARCHING SOCIETY (SMWTMS). Detroit area audio construction club. Meetings every two months featuring serious lectures, recording studio visits, 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. To join or subscribe, e-mail to ad282@leo.nmc.edu, phone to (810) 544-8453 and leave your name and address on the machine, or write SMWTMS, PO Box 721464, Berkley, MI 48072-0464.

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.

HI-FI COLLECTOR/HOBBYIST seeks "living letters"/audio penpals from other states to correspond via reel-to-reel tape. Noncommercial strictly; make up short monologues on subjects from vintage technology, with regional FM excerpts for background or equipment samples, from personal tales of yard-sale scavenging success, repair/restoration tactics and strategies, favorite service centers, general ways to handle the burgeoning obsession with arcane hi-fi gear. All correspondence on 3", 5", 7" reels (¼" tape) will be cheerfully answered and tapes returned via parcel post. James Addison, 171 Hartford Rd., Apt. #7, New Britain, CT 06053.

SPEAKER BUILDERS/AUDIOPHILES in the Milwaukee, WI area. I am a speaker builder with test gear looking to join or form an audio club. Anyone interested contact Kirk Rontti, (414) 355-7509, leave message.

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

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.

Ad Index

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