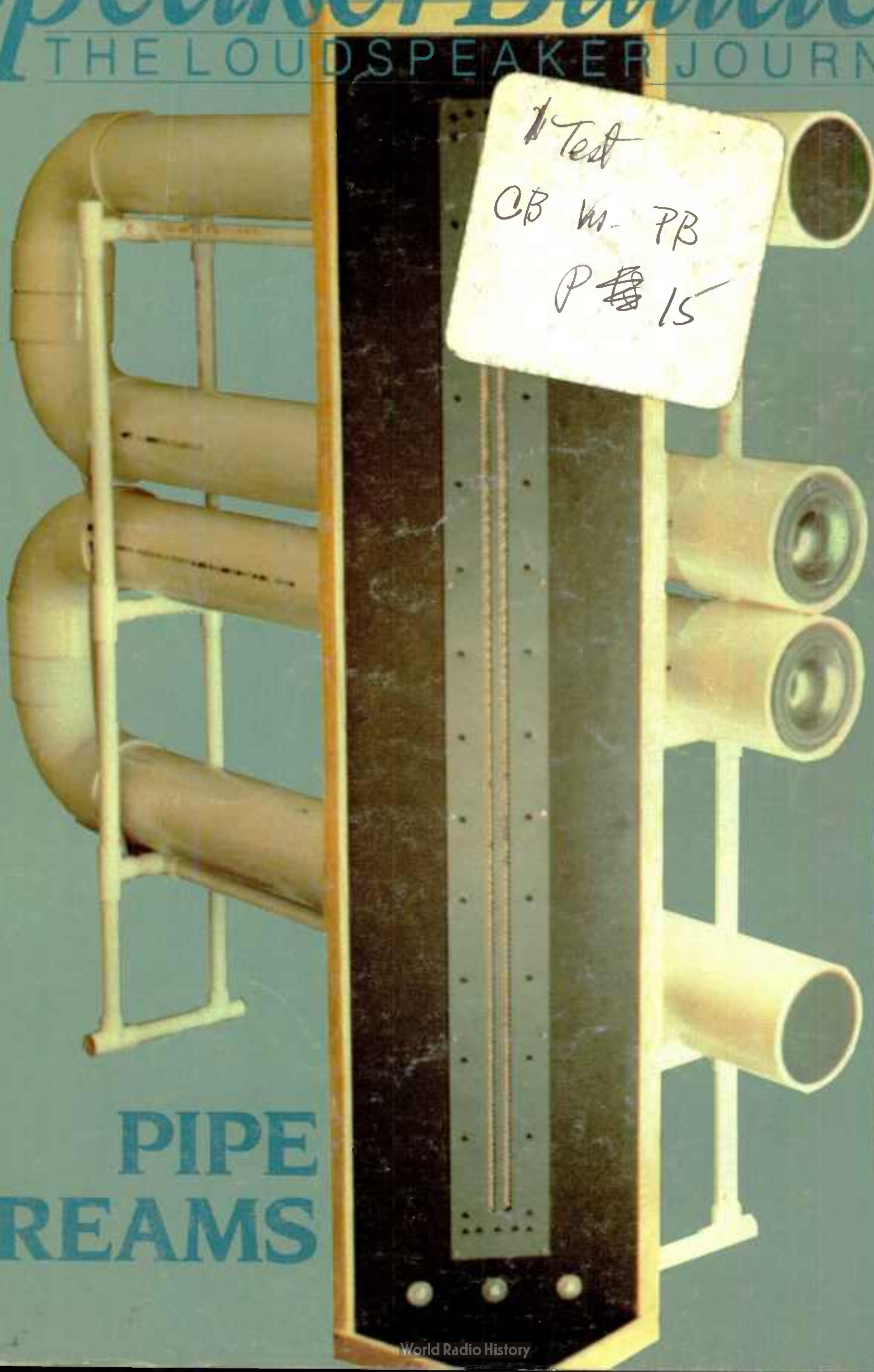
THREE/91

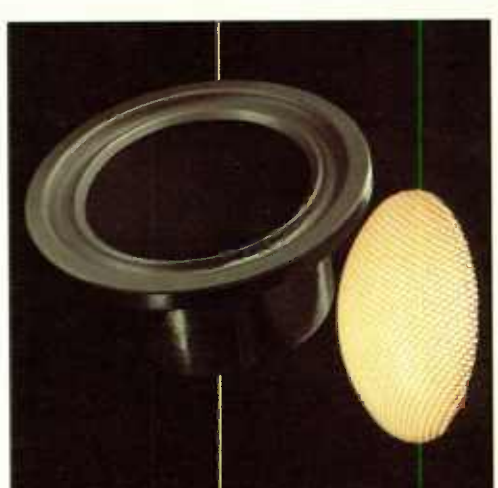
U.S. $7.00
CANADA $8.00

Speaker Builder

THE LOUDSPEAKER JOURNAL



PIPE DREAMS



Good News

To make the scoring of the real-time analysis portion of IASCA car stereo contests easier, faster, and always accurate, **AUDIO CONTROL INDUSTRIAL** has announced a scoring computer that connects to the printer interface of the SA-3050A's ⅓-octave spectrum analyzer. This computer, model HAL-100, analyzes the frequency response curve of the judged vehicle and prints out the IASCA score and the various points on the curve. The entire process using the HAL-100 is quick and eliminates hand scoring and calculations.

The unit is powered by a standard 9V battery or an external 12V power supply shipped with it. An additional parallel printer cable is necessary to connect the HAL-100 between the SA-3050A and the printer.

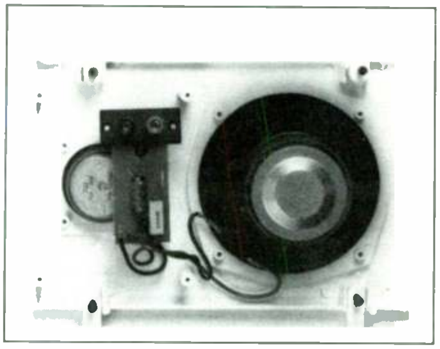
The HAL-100 AISCA sells for $350. For more information, contact Rick Chinn or Tom Walker, AudioControl Industrial, 22313 70th Ave. West, Mountlake Terrace, WA 98043, (206) 775-8461, FAX (206) 778-3166, TELEX 3711409.

Fast Reply #GF123

AUDIO ELECTRONIC SYSTEMS has announced the AES-.5 and the AES 4. The AES-.5 is a compact, "full range," two-way in-wall system designed for applications where the budget is a significant consideration, or as a high-fidelity surround channel speaker in top-of-the-line audio/video sound systems.

The AES-.5 employs a 5¼" (57mm) polypropylene woofer and 2¼" (133mm) ferrofluid-cooled tweeter with on-board thermal protection circuit and five-way binding post connections. Frequency response is 70Hz–20kHz; recommended amplifier power is 10–50W RMS per channel.

The AES-4 dual voice coil subwoofer employs a 6" by 9" (152 by 229mm) polypropylene cone, which has the same surface area as an 8" (203mm) round cone. Normally sold in single units for each stereo installation, the AES-4 complements the AES-3 subwoofer, which is sold in pairs. Both are designed to be mated with all AES full-range in-wall speakers and are shipped with dividing networks that contour the full range speaker's low-



frequency response, ensuring correct musical balance and creating a true in-wall satellite/subwoofer system. Frequency response of the AES-4 is 28–90Hz; recommended amplifier power is 10–100W RMS per channel.

The AES-.5 sells for $99.95, the AES-4 for $295. For more information, contact Jeff Myers, The J&J Myers Co., 22 Parsons Dr., Swampscott, MA 01907-2930, (617) 592-4966, FAX (617) 581-2164.

Fast Reply #GF55

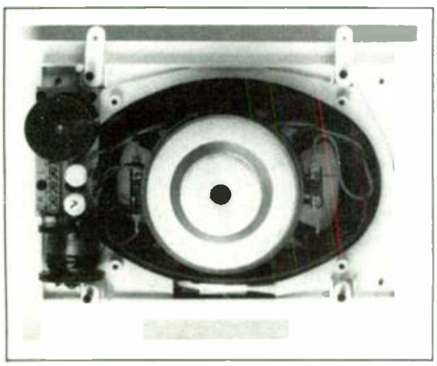
OWI has introduced the OWI-290, a multi-mini loudspeaker system that includes a weatherized® subwoofer and two weatherized two-way satellite speakers.

The subwoofer measures 6¾" by 10" by 6" and weighs 7 lbs. The satellites are 7" by 4½" by 4" and weigh less than 4½ lbs. The subwoofer is rated at 60W maximum, the satellites at 80W. The two-ways have a decibel rating of 92; the subwoofer rates at 93dB. The frequency response of the two-way units is 90Hz–20kHz. The sub's omnidirectional frequency response is 40–450Hz and satellite speakers disperse sound in a 140° radius.

OWI is also introducing the OWI-3302, a full-range weatherized speaker, and the OWI-3301, a coaxial speaker. Both have a power rating of 60W and will run in a 4Ω load. The Thindy III series speakers are 10" in diameter, but will fit into an 8½" diameter hole. Moreover, the speaker requires a depth of only 1¾" for mounting.

For more information, contact OWI, Inc., 1160 Mahalo Place, Compton, CA 90220, (213) 638-8347.

Fast Reply #GF50



← ***Fast Reply #GF29***

SUB-CALC has announced a speaker enclosure design software package for IBM compatibles. Thiele/Small transfer functions and equations have been converted to algorithms and used to create a computer program that accurately predicts speaker/enclosure performance for closed (sealed) box or vented systems.

The software is easy to use. When you enter the basic parameters supplied with the driver, the program suggests an ideal volume, tuning frequency, port diameter, and vent length. You can accept these values or enter your own. The program displays a graph showing the predicted performance and cutoff frequency. If you change any parameter, the program redraws the response curve. Also included in the package are utilities that compute box volume, box resonance, and tube equivalents.

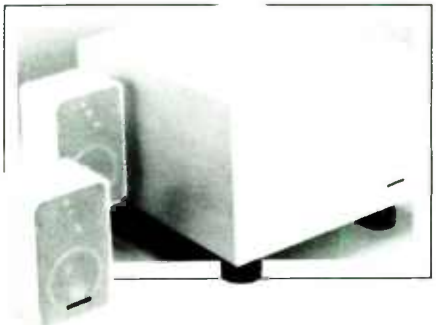
The program sells for $49, plus $3 shipping and handling. For more information, contact Dan Patton or Roger Eakin, SUB-CALC, 202 37th NE, Suite G, Auburn, WA 98002, (206) 931-0173.

Fast Reply #GF256

AUDIOSOURCE has announced the SW Three subwoofer system. The 12" passive subwoofer has a power handling range of 150W maximum, 25 minimum. The built-in passive matrix crossover is set at 150Hz. Available [illegible] or gloss white, [illegible] 15¾" by 18" by [illegible] and was designed [illegible] companion to the AudioSource LS [illegible] or LS Two/A compact loudspeakers along with the company's component separates.

The SW Three subwoofer system sells for $299. With the LS One satellite speakers, it sells for $399 or with the LS Two/A satellites for $449. For more information, contact AudioSource, 1327 N. Carolan Ave., Burlingame, CA 94010, (415) 348-8114, FAX (415) 348-8083.

Fast Reply #GF99



Low-Frequency Designer 1.0 from **SPEAKEASY** is a computer-aided design program written for the IBM PC and compatibles. It models the low-frequency performance of loudspeaker systems and offers features such as models for three types of the popular bandpass systems, System Synthesis, Curve Fitting, and Curve Families.

System Synthesis calculates a set of driver and box parameters based on specified system response. Curve Fitting alters the value of selected driver and box parameters to provide the best fit to the specified system response. Curve Families varies a parameter in steps over a given range and generates a family of curves.

LFD 1.0 is executed with a graphical user interface and on-line context-sensitive help. It requires a minimum of 512K, EGA/VGA/Hercules or AT&T graphics, and Epson or HP compatible printer support.

The program sells for $195. A free demo disk is available by contacting SpeakEasy, 46 Cook St., Newton, MA 02158, (617) 969-1460.

Fast Reply #GF68

AUDIO CONCEPTS has announced the formation of a new subsidiary, AC Components, to better serve speaker hobbyists and manufacturers, while allowing the staff of Audio Concepts to focus on the needs of mainstream audio consumers.

AC Components will supply a full line of drivers, crossover components, and accessories. Included in the catalog are comprehensive specifications featuring the company's own tests of the raw drivers. The specification sheets show the performance of actual production samples and compare drivers from different manufacturers using the same format.

To receive more information or manufacturers' spec sheets, send an SASE (for up to three spec sheets) to AC Components, PO Box 212, La Crosse, WI 54601, (608) 784-4579, FAX (608) 784-6367.

Fast Reply #GF177

MIT has introduced a metallized version of its MultiCap capacitor, designed for use in audio products to eliminate the distortions caused by parasitics in capacitors.

The metallized MultiCap is recommended for low-current applications such as preamps, amps, and other electronic circuits. They offer the same precision in design and construction, the same patented coaxial structure, and the same sonics found in the film-and-foil MultiCap. Each unit contains multiple paralleled capacitors optimized into one package.

For more information, contact Music Interface Technologies, 3037 Grass Valley Hwy., Suite 8212, Auburn, CA 95603, (916) 883-1186, FAX (916) 823-0810.

Fast Reply #GF635

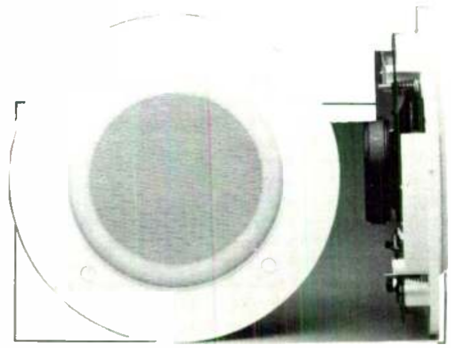
Good News

To make the scoring of the real-time analysis portion of IASCA car stereo contests easier, faster, and always accurate, **AUDIO CONTROL INDUSTRIAL** has announced a scoring computer that connects to the printer interface of the SA-3050A's ⅓-octave spectrum analyzer. This computer, model HAL-100, analyzes the frequency response curve of the judged vehicle and prints out the IASCA score and the various points on the curve. The entire process using the HAL-100 is quick and eliminates hand scoring and calculations.

The unit is powered by a standard 9V battery or an external 12V power supply shipped with it. An additional parallel printer cable is necessary to connect the HAL-100 between the SA-3050A and the printer.

The HAL-100 AISCA sells for $350. For more information, contact Rick Chinn or Tom Walker, AudioControl Industrial, 22313 70th Ave. West, Mountlake Terrace, WA 98043, (206) 775-8461, FAX (206) 778-3166, TELEX 3711409.

Fast Reply #GF123



OWI has introduced the OWI-290, a multi-mini loudspeaker system that includes a weatherized® subwoofer and two weatherized two-way satellite speakers.

The subwoofer measures 6¾" by 10" by 6" and weighs 7 lbs. The satellites are 7" by 4½" by 4" and weigh less than 4½ lbs. The subwoofer is rated at 60W maximum, the satellites at 80W. The two-ways have a decibel rating of 92; the subwoofer rates at 93dB. The frequency response of the two-way units is 90Hz–20kHz. The sub's omnidirectional frequency response is 40–450Hz and satellite speakers disperse sound in a 140° radius.

OWI is also introducing the OWI-3302, a full-range weatherized speaker, and the OWI-3301, a coaxial speaker. Both have a power rating of 60W and will run in a 4Ω load. The Thindy III series speakers are 10" in diameter, but will fit into an 8½" diameter hole. Moreover, the speaker requires a depth of only 1¾" for mounting.

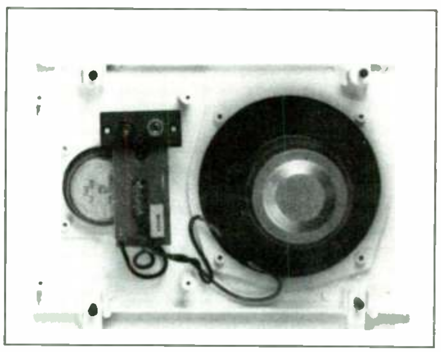
For more information, contact OWI, Inc., 1160 Mahalo Place, Compton, CA 90220, (213) 638-8347.

Fast Reply #GF50

AUDIO ELECTRONIC SYSTEMS has announced the AES-.5 and the AES 4. The AES-.5 is a compact, "full range," two-way in-wall system designed for applications where the budget is a significant consideration, or as a high-fidelity surround channel speaker in top-of-the-line audio/video sound systems.

The AES-.5 employs a 5¼" (57mm) polypropylene woofer and 2¼" (133mm) ferrofluid-cooled tweeter with on-board thermal protection circuit and five-way binding post connections. Frequency response is 70Hz–20kHz; recommended amplifier power is 10–50W RMS per channel.

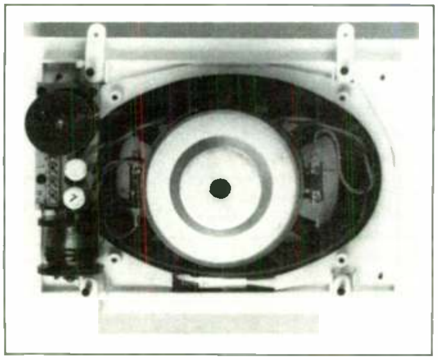
The AES-4 dual voice coil subwoofer employs a 6" by 9" (152 by 229mm) polypropylene cone, which has the same surface area as an 8" (203mm) round cone. Normally sold in single units for each stereo installation, the AES-4 complements the AES-3 subwoofer, which is sold in pairs. Both are designed to be mated with all AES full-range in-wall speakers and are shipped with dividing networks that contour the full range speaker's low-



frequency response, ensuring correct musical balance and creating a true in-wall satellite/subwoofer system. Frequency response of the AES-4 is 28–90Hz; recommended amplifier power is 10–100W RMS per channel.

The AES-.5 sells for $99.95, the AES-4 for $295. For more information, contact Jeff Myers, The J&J Myers Co., 22 Parsons Dr., Swampscott, MA 01907-2930, (617) 592-4966, FAX (617) 581-2164.

Fast Reply #GF55



← ***Fast Reply #GF29***

SUB-CALC has announced a speaker enclosure design software package for IBM compatibles. Thiele/Small transfer functions and equations have been converted to algorithms and used to create a computer program that accurately predicts speaker/enclosure performance for closed (sealed) box or vented systems.

The software is easy to use. When you enter the basic parameters supplied with the driver, the program suggests an ideal volume, tuning frequency, port diameter, and vent length. You can accept these values or enter your own. The program displays a graph showing the predicted performance and cutoff frequency. If you change any parameter, the program redraws the response curve. Also included in the package are utilities that compute box volume, box resonance, and tube equivalents.

The program sells for $49, plus $3 shipping and handling. For more information, contact Dan Patton or Roger Eakin, SUB-CALC, 202 37th NE, Suite G, Auburn, WA 98002, (206) 931-0173.

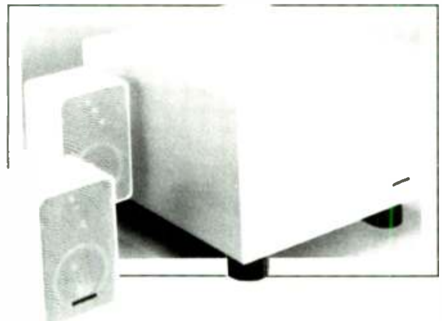
Fast Reply #GF256

AUDIOSOURCE has announced the SW Three subwoofer system. The 12" passive subwoofer has a power handling range of 150W maximum, 25 minimum. The built-in passive matrix crossover is set at 150Hz.

Available in a matte black or gloss white, the SW Three measures 15¾" by 18" by 12¼" (including feet) and was designed as a companion to the AudioSource LS One or LS Two/A compact loudspeakers along with the company's component separates.

The SW Three subwoofer system sells for $299. With the LS One satellite speakers, it sells for $399 or with the LS Two/A satellites for $449. For more information, contact AudioSource, 1327 N. Carolan Ave., Burlingame, CA 94010, (415) 348-8114, FAX (415) 348-8083.

Fast Reply #GF99



Low-Frequency Designer 1.0 from **SPEAKEASY** is a computer-aided design program written for the IBM PC and compatibles. It models the low-frequency performance of loudspeaker systems and offers features such as models for three types of the popular bandpass systems, System Synthesis, Curve Fitting, and Curve Families.

System Synthesis calculates a set of driver and box parameters based on specified system response. Curve Fitting alters the value of selected driver and box parameters to provide the best fit to the specified system response. Curve Families varies a parameter in steps over a given range and generates a family of curves.

LFD 1.0 is executed with a graphical user interface and on-line context-sensitive help. It requires a minimum of 512K, EGA/VGA/Hercules or AT&T graphics, and Epson or HP compatible printer support.

The program sells for $195. A free demo disk is available by contacting SpeakEasy, 46 Cook St., Newton, MA 02158, (617) 969-1460.

Fast Reply #GF68

AUDIO CONCEPTS has announced the formation of a new subsidiary, AC Components, to better serve speaker hobbyists and manufacturers, while allowing the staff of Audio Concepts to focus on the needs of mainstream audio consumers.

AC Components will supply a full line of drivers, crossover components, and accessories. Included in the catalog are comprehensive specifications featuring the company's own tests of the raw drivers. The specification sheets show the performance of actual production samples and compare drivers from different manufacturers using the same format.

To receive more information or manufacturers' spec sheets, send an SASE (for up to three spec sheets) to AC Components, PO Box 212, La Crosse, WI 54601, (608) 784-4579, FAX (608) 784-6367.

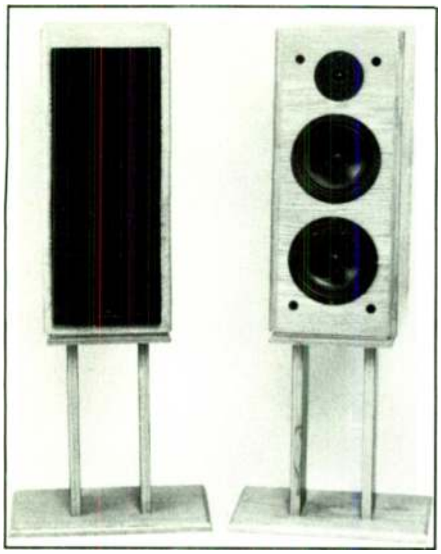
Fast Reply #GF177

MIT has introduced a metallized version of its MultiCap capacitor, designed for use in audio products to eliminate the distortions caused by parasitics in capacitors.

The metallized MultiCap is recommended for low-current applications such as preamps, amps, and other electronic circuits. They offer the same precision in design and construction, the same patented coaxial structure, and the same sonics found in the film-and-foil MultiCap. Each unit contains multiple paralleled capacitors optimized into one package.

For more information, contact Music Interface Technologies, 3037 Grass Valley Hwy., Suite 8212, Auburn, CA 95603, (916) 883-1186, FAX (916) 823-0810.

Fast Reply #GF635



M&C SPEAKERWORKS has announced the Series I loudspeaker system. The speaker incorporates two 6½" polymer woofers and one ¾" soft-dome tweeter with a sensitivity of 90dB/1W/1M and a frequency range of 37Hz–20kHz at 4Ω. It also has 24kt gold solid brass binding posts for a steady connection from your amplifier and has a maximum power rating of 150W RMS. The Series I loudspeaker system, complete with its accompanying stand, measures 12" by 9½" by 39" and has a handcrafted ported enclosure covered with real oak veneer.

The Series I sells for $800 per pair. For more information, contact M&C Speakerworks, Dept. SR, PO Box 151082, San Diego, CA 92175, (619) 281-6463.

Fast Reply #GF11

AccuPower, designed and manufactured by **EUPHONIC TECHNOLOGY**, is a patented, four-channel, isolating power-line filter. Each output channel incorporates an independent four-stage LC filter network employing 70dB of attenuation for frequencies between 10kHz and 250MHz. Independent isolating filters prevent component interactions, which can degrade audio and video performance. Each output channel can handle 1,550W of continuous power and surges in excess of 100A. Total continuous power handling is 1,875W.

All outputs incorporate six stages of spike and surge suppression with one picosecond response time. Components are fully protected from the hazards of damaging line surges and transients. The AccuPower AP-4 also includes a fifth output that provides spike and surge protection with no filtering.

For more information, contact your local audio dealer or Euphonic Technology, 19 Danbury Rd., Ridgefield, CT 06877, (203) 431-6434, FAX (203) 431-3660.

Fast Reply #GF248

THE MOD SQUAD INC. has introduced the Wonder Link digital cable, which carries the digital signal from the output of each digital source to the inputs of your outboard processor (or digital preamp). The manufacturer claims that Wonder Links' transparency reveals natural music, instead of creating artificial hi-fi sound. It cures hard glare, harsh background smear, unmusical distortion, and fuzzy defocus that often make digital fatiguing.

Wonder Link is said to be designed to match real digital systems, not paper specifications, to reveal music more accurately than 75Ω cable. It comes in two standard lengths: 0.5m and 1m. You can also custom-order the length your installation requires. Wonder Link comes with RCA phono and BNC connectors.

For more information, contact The Mod Squad Inc., 542 N. Hwy. 101, Leucadia, CA 92024, (619) 436-7666.

Fast Reply #GF640

In January, **TOROID CORPORATION OF MARYLAND** consolidated its Bowie and Salisbury manufacturing operations. As of March 18, the new corporate address and telephone numbers are Toroid Corporation of Maryland, 608 Naylor Mill Rd., Salisbury, MD 21801-9627, (301) 860-0300, FAX (301) 860-0302.

Speaker Builder

Edward T. Dell, Jr. *Editor/Publisher*
Karen Hebert *General Manager*
Contributing Editors
Joseph D'Appolito Robert Bullock
John Cockroft David Davenport
Vance Dickason Bruce C. Edgar
Gary Galo G. R. Koonce
Richard Pierce
Robin Florence *Managing Editor*
Christine Holt *Graphics Director*
Graphics Assistants
Robert Drew Karen Kellogg
Edward Roane
Katharine Gadwah *Circulation Director*
Circulation Assistants
Laurel Barlow Doris Hamberg
Helen Senechal
Maria Bane *Advertising Assistant*
Techart Associates *Drawings*
Advertising Rates & Schedules
Rally Dennis
PO Box 494
Peterborough, NH 03458
Advertising Phone: (603) 924-6710
Editorial and Circulation Offices
Post Office Box 494
Peterborough, New Hampshire 03458
(603) 924-9464 FAX: (603) 924-9467

Speaker Builder is published bi-monthly by Edward T. Dell, Jr., PO Box 494, Peterborough, NH 03458.

All subscriptions are for the whole year. Each subscription begins with the first issue of the year and ends with the last issue of the year. A sample issue costs $4 in the US, $5 in Canada.

Subscription rates in the United States and possessions: one year (six issues) **$25**, two years (twelve issues) **$45**. All sets of back issues are available beginning with 1980. Canada add **$4** per year for postage. Overseas rates available on request. Subscribers residing outside the US and Canada are served by air.

To subscribe, renew or change address in all areas write to Circulation Department or call (603) 924-9464 for MC/Visa charge card orders. For gift subscriptions please include gift recipient's name and your own, with remittance. A gift card will be sent.

A Note To Contributors

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

Speaker Builder (US ISSN 0199-7920) is published bi-monthly at $25 per year; $45 for two years, by Edward T. Dell, Jr. at 305 Union St., Peterborough, NH 03458 USA. Second class postage paid at Peterborough, NH and additional mailing office.

POSTMASTER: Send address changes to **SPEAKER BUILDER**, PO Box 494, Peterborough, NH 03458.

About This Issue

That dreaded radar that puts fear into speeding motorists can also be used to measure the low-frequency response of speaker enclosures. **Eldon Sutphin**'s treatment of the subject (beginning on page 10) will help you determine whether your enclosure is functioning as you intended.

If you desire a fast, accurate subwoofer and are not concerned with high volume levels, check out **Jerry McNutt** and **Mike Aughtman**'s article on page 18. Their subwoofer, constructed for Mike's GMC Jimmy, has the added advantage of being easily removable.

Next, **Roy Mallory** describes how to model almost any speaker system as an equivalent circuit and shows how to use PSpice to make graphs of frequency response, phase response, transient response, and cone excursion versus frequency. Turn to page 20 for the details.

A trip to the Consumer Electronics Show inspired **Scott Wolf** to create an unusual-looking speaker system using ribbons and PVC pipes. Read about his adventure on page 28.

If you prefer a system that's less flamboyant, **Ralph Gonzalez**' DELAC S10 may suit you. This elegant, unusual, and compact speaker puts soundstaging, timbral balance, musicality, and aesthetics above high power handling and bass response below 50Hz. The construction information begins on page 32.

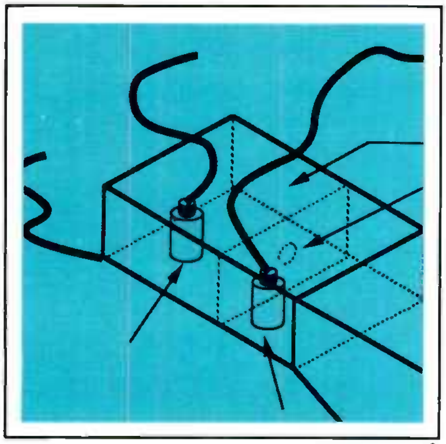
Joseph O'Connell's offering is a volume control for remote speakers. His project starts on page 40.

Contributing Editor **Gary Galo** highly praises Audio Concepts Sub-1 synthesized bandpass subwoofer, saying it complements the company's Sapphire II loudspeakers. See page 51 for his review.

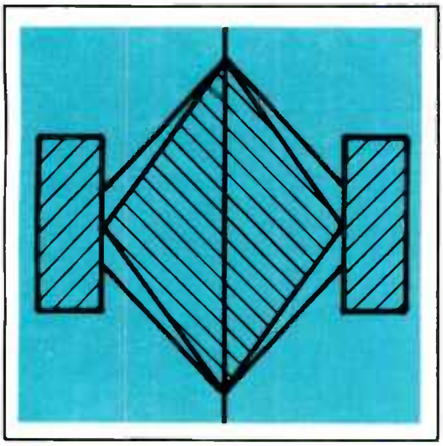
On our cover: **Scott Wolf**'s trek into the world of pipes and ribbons has produced a very atypical-looking speaker system. Photo by the author.

SpeakerBuilder
THE LOUDSPEAKER JOURNAL

VOLUME 12 NUMBER 3 JUNE 1991



10



18

FEATURES

DEPARTMENTS


FOCAL
technologie d'avant-garde
AUDIOM 15VX Polyglass 15″ subwoofer
for your free copy of the complete FOCAL catalog: call or write Focal America, Inc.
1531 Lookout Drive, Agoura, CA 91301. USA. TEL: (818) 707-1629 FAX: (818) 991-3072

Guest Editorial

RECORDING: PHOTO OR PAINTING?

By Mako Natsume

I am writing in response to your editorial "Will Loudspeakers Become Irrelevant?" from *SB* 2/91. It's important to distinguish between two recording/mixing methods. One, which has a long and distinguished history, is to re-create a musical event as closely as possible to how it sounded when performed (such as old Deutsche Grammophon recordings and new Telarc CDs).

The other style attempts to establish a balance between instruments that have inherent differences in the level of sound they can produce. Anyone who has tried to talk in the same room where a drummer is playing could see the futility of trying to *sing* in the same room without amplification.

Synthesizers generate little acoustic energy of their own—most require external amplification. To impart an illusion of space to the dry sound of DI'ed instrument, a mixing engineer will add reverb. There is little difference between amplification, the various delays, digital reverbs, flangers, and choruses and these new enhancement devices; they all alter signals in ways that may or may not be pleasing to all ears, but which will undoubtedly allow artists to express themselves in the same way a cellist will wiggle a finger to put vibrato into a note.

You cannot use a "purist" microphone placement in a modern pop session to capture the complete musical event realistically; musicians seldom play together at the same time. If these recordists and mixers are "people (who) believe sound ought to be an 'experience,' " then we are talking about matters of contextual preferences—live unamplified performance versus live amplified or multitrack studio recordings.

As for the mono incompatibility "problem," what radio station would broadcast a Q sound disc in mono save for those with a 5kHz audio bandwidth. The two Q sound discs I've heard have no dearth of "sum" (mono) signal, and on Sting's *Soul Cages*, the extreme outside-of-speaker effects are kept to a minimum.

These effects are far less noticeable on boom boxes and car stereos than on my system, which consists of home-built biamped Dynaudio loaded cabinets in a room with reflective surfaces treated with convoluted foam to minimize early reflections—something that will enhance the quality of reproduction for any recording method by eliminating room cues.

If you believe the ultimate goal in audio is to obtain total musical accuracy from playback through two spaced speaker systems, then that is a worthy aim. Saying that realism is the only valid goal of audio is like saying color photographs are better than impressionist paintings. Recorded music will *always* be an approximation of a live musical performance. Why not use this to our advantage and make the playback more fun?

Will loudspeakers become irrelevant? Not as long as new recording techniques become available and place new demands on speakers—and speaker builders.

Mr. Natsume is quite correct that many people enjoy recorded concoctions composed of synthesizers, or electrically amplified performances in a variety of locations assembled in a unified production. But these are synthetics, cleverly assembled and presented. They are cute, they have their place, and many enjoy them, myself included.

But that is not what my editorial is objecting to. I believe a performance by real people at a given time and in a particular space has a validity and wholeness which is unique. I have watched London's engineers carefully and painstakingly capture the sound of the L'Orchestre Symphonique de Montreal and attempt, as best they can, to lay it down with just as little manipulation of the signal as possible.

My editorial was not a worry directly about some "enhancers" juicing up a rock performance per se. What I and others are worrying about is permanently altering the recording of a performance to "enhance" ambience by playing games with the phase. I have no objection whatever to anyone taking the original recording of a performance and making an "entertainment" out of it in terms of ambient dynamics or whatever. I am only saying that the original, made as near as can be to the standards of either the Blumlein patent or the more recent Ambisonic definition, should be offered as well.

Pop versions of music have a long and distinguished history. But I would prefer not to have "pop" dynamics chosen for me by the engineers who get to decide they can make it better than the reality.

Like others who have attended recording technology seminars at AES, I have seen the red faces of the CBS and RCA engineers who multi-miked and multi-mixed great symphonic performances of the sixties and seventies only to have their sins hung out on the digital reprint clothesline for all to hear. The art of undoing as much of that as possible to make the performances presentable in the unforgiving CD format has become a minor industry.

I have no idea how many stations broadcast with a 5kHz bandwidth, but if the BBC is worried about broadcasting Q sound, I think we can take it as given that the system is genuinely problematical technically and their concern is not merely a fit of pique.

No present recording system can capture a live performance. The industry hasn't the will or the wit to adopt an Ambisonic technique (with one exception in England) or to invest the money needed to discover a substantial improvement on the Blumlein or Middle-Side or Ambisonic systems. They are busily outspending one another to compete in the race to make the dominant DAT as in Philips' DCC, Sony's new Micro-Dat, and whatever is next, all to reproduce the same old, same old 45/45 two channels.

I think Mr. Natsume's impressionist paintings analogy is flawed. Our "cameras" for sound are primitive and all have shortcomings. But I do not believe the Q Sound group or Hughes Aircraft's engineers are giving us the means to enable a Matisse of the recording arts. We're being offered more gimmickry rather than a better camera. I'm sure Lincoln's Gettysburg address can be edited, enhanced, and "clarified" by those who believe it could be more entertaining in a modern version. But the original deserves its integrity. So does each live musical event.—E.T.D.

MEASURING DRIVERS WITH RADAR

BY ELDON M. SUTPHIN

One major problem facing the serious speaker builder is whether a finished enclosure is functioning as intended. The greatest factor in determining an enclosure's size and shape is its low-frequency response, one of the more elusive measurement areas. Most inexpensive sound-level meters lack accuracy over this frequency range, not to mention errors associated with room acoustics or the inconvenience of outdoor measurements. I will present a reasonably low-cost approach to solving the low-frequency measurement problem.

BACKGROUND. Since building loudspeaker enclosures is a hobby for most of us rather than a profession, a good low-cost method of making reasonable measurements in the 10–200Hz range is needed. I considered calibrating my Radio Shack sound-level meter over this range and using correction factors. But what would I use for a low-cost, variable frequency acoustical standard to determine the factors?

I needed something capable of delivering a known amount of acoustical energy over the desired frequency range. First, I considered using the piston in a model airplane engine mounted on an air-tight box to create a specified pressure level; a variable speed motor could vary the frequency. This approach might create too much additional acoustical noise, making a cumbersome filter necessary.

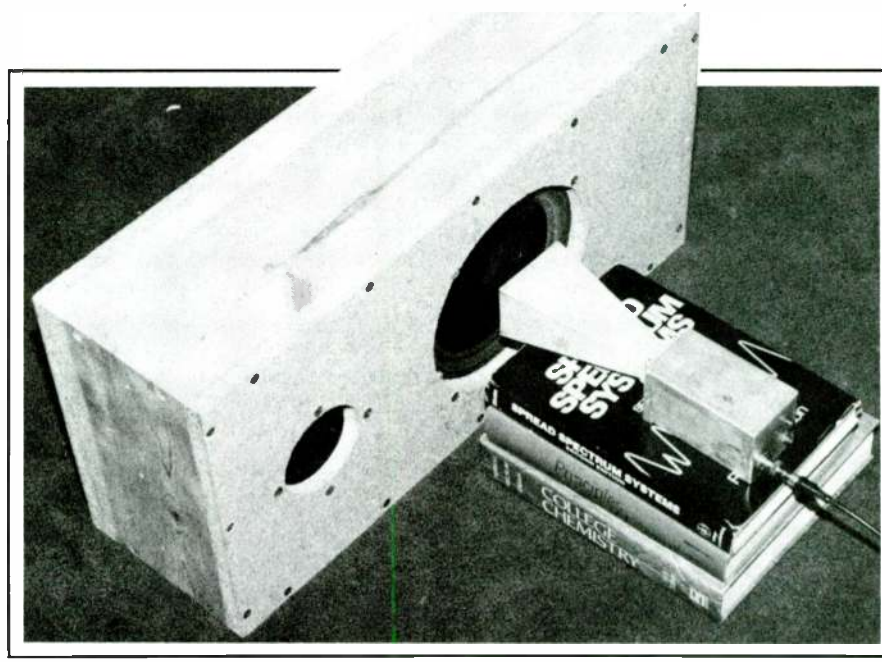


PHOTO 1: The microwave transceiver.

Next, I thought about using a speaker as the piston if I could measure its displacement. With a laser inferometer, I could paint or glue a reflector to the cone and directly measure displacement, but that seemed economically out of the question.

Finally, I decided to go to a lower frequency—microwave.

Over the years, I have experimented with X Band microwave transceivers (yes, those gadgets we dread on the highways when they are attached to a police cruiser). Using one to indicate speaker displacement is ideal because it lacks a mechanical connection to the speaker. Since these transceivers are also used for intrusion alarms and automatic door openers, they are inexpensive and are built to meet FCC specifications. They are available from several sources (such as Alpha Industries and MACOM) and cost from $50–$80.

I connected a signal generator to an amplifier and speaker and observed the output diode of the X Band transceiver. I was pleasantly surprised to see a clean signal after properly positioning the transceiver about 2–3″ from the speaker. The metal voice coil does a great job of reflecting the microwaves. By adjusting the amplifier volume and viewing the scope, I could keep the speaker displacement constant as frequency varied.

To avoid producing a distorted waveform, the speaker displacement during test must be less than a quarter wavelength (at 10.525GHz, this amounts to a displacement of less than 0.28″). Position the transceiver so when the speaker is at one extreme, the reflected microwaves add in-phase, causing a signal maximum at the detector diode. When the cone excursion is at the other extreme, the reflected microwave signals are more out-of-phase, causing a signal minimum.

ABOUT THE AUTHOR

Eldon Sutphin is a consulting engineer for a large electronics firm and is involved in the design of defense electronics systems. He is a graduate of the Pennsylvania State University College of Electrical Engineering and has been an electronics and audio enthusiast for many years. An avid speaker builder, he particularly enjoys the scientific side of speaker building and audio measurements.

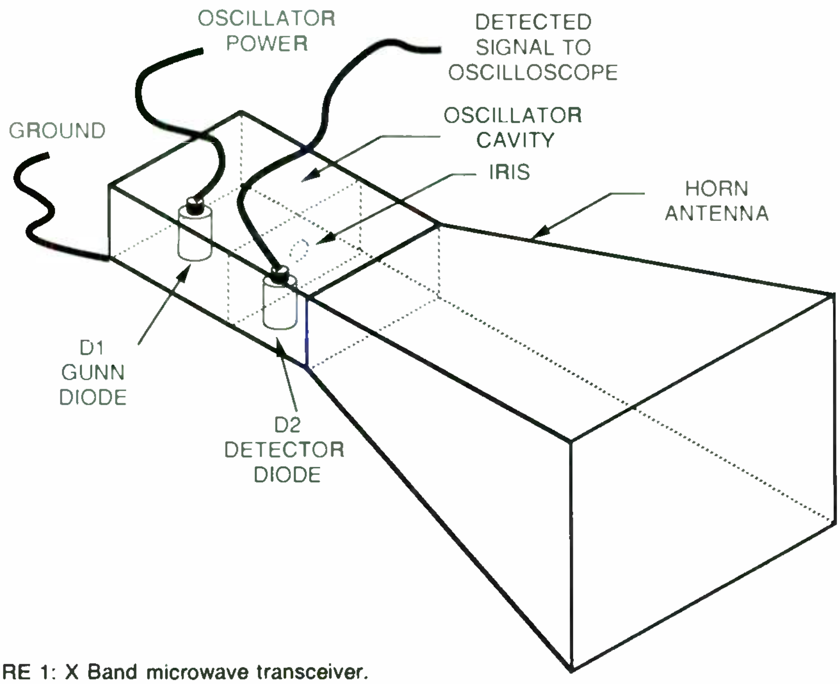


FIGURE 1: X Band microwave transceiver.

This technique works from DC to many megahertz and is quite flat in determining relative displacement over the audio range. Since you can accurately control speaker displacement by varying the drive voltage at various frequencies, you can establish a relative sound-level reference.

There is, however, one small catch. The sound level varies with the square of the frequency if the speaker displacement is held constant. In other words, if the cone deviation is constant at 0.1″, the sound level would be 16 times greater at 100Hz than at 25Hz since the frequency ratio is 4:1. You must consider this when you calibrate a sound-level meter. A computer with a spreadsheet program makes this task simple.

While experimenting with calibrating a sound-level meter in this way, I realized if I measured the speaker voltage required to keep the displacement constant as frequency varied, I could measure the speaker's frequency response directly (keeping in mind the square law relationship of displacement to frequency). The direct method is more accurate than sound-level meter calibration since it eliminates meter errors, and room acoustics are completely eliminated for closed-box designs and greatly reduced for bass-reflex designs. (You can always use the outdoor anechoic chamber if your neighbors are far enough away and extraneous noise sources such as wind and traffic are negligible.)

X BAND TRANSCEIVER. *Figure 1* is a diagram of a microwave transceiver used for speed and motion detection. It outputs a continuous signal, rather than a pulsed microwave signal used for distance-measuring radar systems. The transceiver contains two diodes: a Gunn diode (D1) and a detector diode (D2).

The Gunn diode resides in a resonant cavity tuned to the selected frequency, in this case 10.525GHz. Biased in a region of negative resistance, it causes oscillations to occur at the cavity's resonant frequency. Microwave energy is coupled to the outside through an iris, actually nothing more than a hole in the cavity.

Some of this energy is coupled to D2, which detects the returns by a mixing action. This diode is then placed in the waveguide section in a way that a small amount of the outgoing microwave energy is mixed with as much of the incoming target reflections as possible.

As a reflective target moves toward the transceiver, the outgoing and incoming signals may be in-phase with each other. As the target gets closer, the reflected signal starts going out-of-phase, causing the signal's amplitude to decrease. When the signal becomes 180° out-of-phase, a voltage minimum is created at the diode. As the target gets even closer, the reflected signal becomes more in-phase with the outgoing signal, which increases until a maximum is reached when no phase shift is present at D2.

As the target moves closer, this process of adding in-phase and out-of-phase repeats. The faster the target moves toward the transceiver, the more often the in-phase/out-of-phase condition exists, giving a direct method of measuring the target's speed. The frequency at which this in-phase/out-of-phase condition occurs is directly proportional to the speed of the target and is known as the Doppler frequency. To keep speaker displacement constant, the transceiver is not used in its normal Doppler mode, but in inferometer mode.

For this application, however, I am not interested in the speaker's speed, but in its relative cone deviation to cause the voltage at D2 to vary between a voltage minimum and maximum, but not to have so much deviation that the voltage will alternate between multiple minimums and maximums. This would produce distortion and frequency multiples rather than a clean waveform at the driven speaker frequency.

Supply voltages for these microwave transceivers usually vary between 6 and 12V, depending on the model and power output. For this application, transceivers in the 5–10mW range are fine; the larger 50–100mW units are more expensive. The target in this case is only an inch to several inches away and unless you have a different application need, you may find lower power units more economical.

The lower power units can also be battery powered; however, you should use

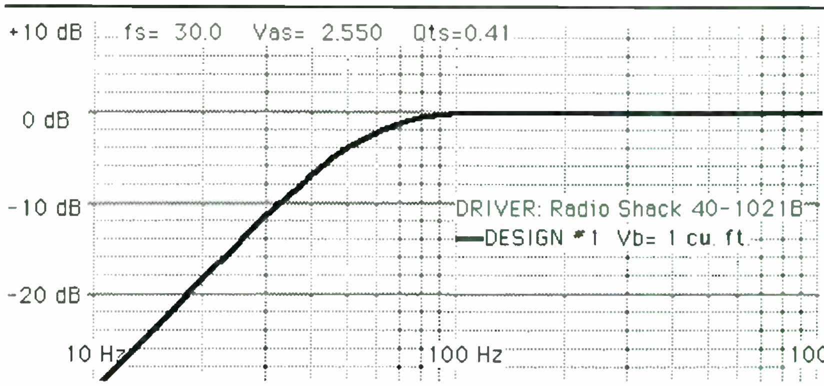


FIGURE 2: Curve for a closed-box design of 1 cubic foot using an 8″ Radio Shack speaker.

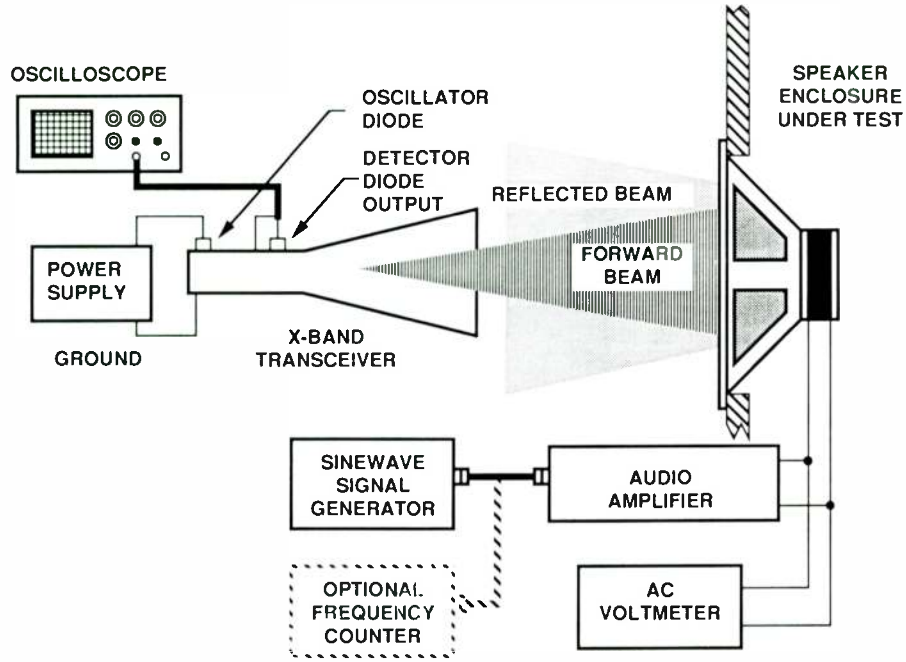


FIGURE 3: Block diagram of the test setup.


PHOTO 2: 1 cubic foot closed-box enclosure using Radio Shack's 40-1021 8″ driver.

a regulated power supply for this particular application. If the batteries are weak during the measurement, your results will be erroneous since the sensitivity of the transceiver will change with supply voltage. If you reverse the supply voltage on the transceiver, the Gunn diode will be damaged if the supply has sufficient current (the diode will be forward-biased in this case).

EQUIPMENT. In addition to the microwave transceiver, you will need other test equipment.

1. **Oscilloscope.** Since the measurements occur in the low audio range, the oscilloscope should have input AC coupling well below 10Hz or DC coupling. If you use the latter, you must provide your own AC coupling into the channel. This is necessary since the microwave transceiver's detector diode has a DC component much larger than its AC component. [*Excellent, used military Tektronix scopes are readily available on the surplus market from several suppliers, including Fair Radio Sales, Box 1105, Lima, OH 45802. New import types are also widely available.—Ed.*]

For scopes with a 1MΩ input impedance, a coupling capacitor of 1μF between the detector diode and the scope input will suffice. The scope should have a voltage sensitivity of at least 0.01V per centimeter. If it doesn't, use a preamplifier with good low-frequency characteristics—remember, the measurements you will be making are in the 10–200Hz range.

2. **Signal generator.** Almost any signal generator with a sine wave output that covers the 10–200Hz range will do. If the front panel dial is inaccurate, you may wish to connect a frequency counter to the generator output.

3. **AC voltmeter.** The AC voltmeter should be either DC coupled or capable of accurate measurements to 10Hz. Digital is preferred for better accuracy: the more accurate your measurements, the smoother the response plot. You can use a second scope input channel if a voltmeter is unavailable.

4. **Sound-level meter.** If you plan to test bass-reflex enclosures, you will need to measure the difference in decibels when the port is open and when it is closed at various frequencies. Radio Shack's low-cost meter (part number 33-2050, $31.95) works well. Low-frequency errors caused by the sound-level meter are eliminated since a difference in decibels is measured with having the port open and having it sealed at the same frequency.

5. **Computer.** A computer with a spreadsheet and plotting program will help you with data calculations and graph plotting.

CLOSED-BOX DESIGN AND TEST. Of the two designs I will present, the closed box is the simpler in construction and test. It is based on Radio Shack's 40-1021 8″ driver (now discontinued; however, the 40-1204 is similar). I used my MacSpeakerBox (available from Old Colony) to design the enclosure. The software uses Thiele/Small design parame-

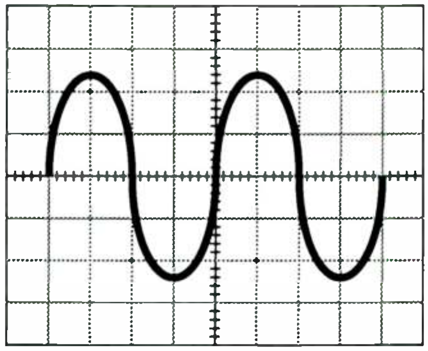


FIGURE 4a: Scope display with proper displacement and speaker/transceiver distance. Distance should be adjusted for best display.

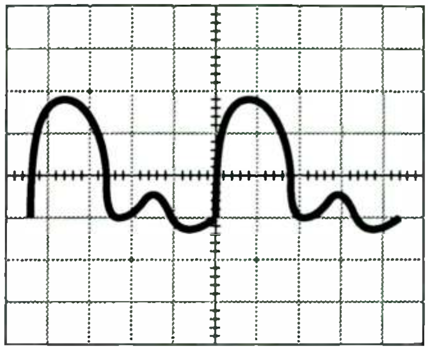


FIGURE 4b: Scope display showing improper speaker/transceiver distance or speaker being overdriven.

ters to predict speaker enclosure response and handles closed-box, infinite-baffle, and bass-reflex designs.

The graph in *Fig. 2* shows the predicted response. I built the enclosure using 9¼″ wide by ¾″ thick wood (1″ by 10″ board) for the sides and ⅝″ particle board for the front and back. This allows quick construction for experimental purposes since you can use drywall screws to fasten the front and back panels directly to the sides. I sealed the seams with RTV and damped the box by stapling ordinary building insulation fiberglass to the interior walls. *Photo 2* shows the 1 cubic foot enclosure.

Next, I tested the box's low-frequency end using the microwave transceiver test setup shown in *Fig. 3*. As the signal generator's frequency is varied, the signal amplitude is adjusted so the cone excursion represented by the waveform on the oscilloscope remains constant. *Figures 4a* and *4b* show proper and improper alignments of the microwave transceiver to the speaker cone.

I measured and recorded the voltage on the speaker terminals after I adjusted the drive signal for constant amplitude as displayed on the scope at each measured frequency. You should make the measurements going from a high frequency to a low frequency since measurement sensitivity is less at the high frequencies due to the square law effect of cone deviation versus applied voltage.

The higher frequencies will be much louder than the lower, so you can adjust the drive level for measurement sensitivity and noise tolerance level during the test. If the variation in sensitivity is too great, split the measurements into two or more groups and merge them, but keep overlapping frequency points to reference one group against the other. Be careful not to disturb the position of the microwave transceiver relative to the speaker during test since the amplitude will shift with position resulting in erroneous measurements.

Test measurements on the 1 cubic foot enclosure along with calculated data are shown in *Table 1*. Column A is the test frequency and column B is the voltage measured across the speaker terminals required to keep the excursion constant as displayed on the scope. Column C is the calculated speaker terminal voltage response of an "ideal" speaker, with the highest starting frequency and voltage from columns A and B used as references. If you used a completely flat speaker in the test setup, this column represents the voltage that the following formula would calculate:

$$\text{Ideal speaker volts} = (V_h) \times (F/F_h)^2$$

where V_h is the voltage across the speaker at the highest frequency, f_h is the highest frequency, and F is the selected measurement frequency.

Column D is the relative response in decibels obtained by comparing the measured voltage of column C to the ideal voltage of column B as a function of fre-

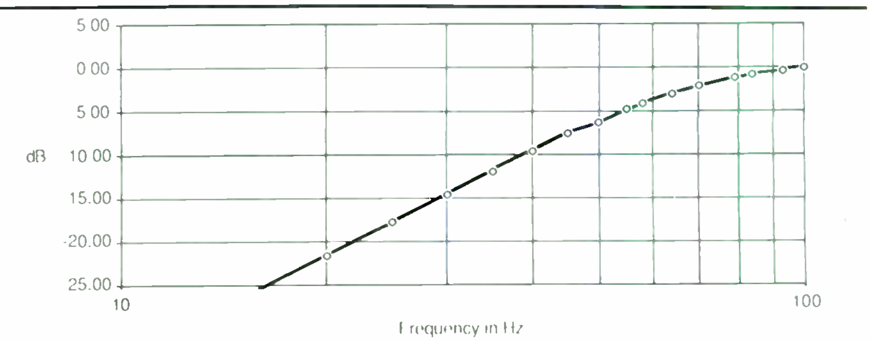


FIGURE 5: Measured response of a 1 cubic foot closed-box design using the 40-1021B driver.

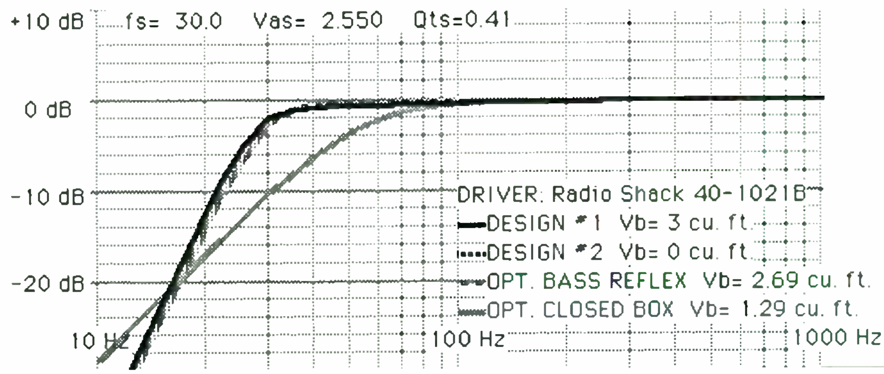


FIGURE 6: Response of optimum bass-reflex, closed-box, and 3 cubic foot enclosures using Radio Shack's 40-1021B 8″ driver.

TABLE 1

MEASURED CHARACTERISTICS OF 1 CUBIC FOOT CLOSED-BOX ENCLOSURE

Frequency	Measured Voltage Required for Constant Cone Excursion	Volts Required for Constant Cone Excursion for Ideal Speaker	dB
100	3.540	3.540	0.00
93	3.210	3.062	-0.41
84	2.758	2.498	-0.86
79	2.531	2.209	-1.18
70	2.219	1.735	-2.14
64	2.038	1.450	-2.96
58	1.911	1.191	-4.11
55	1.850	1.071	-4.75
50	1.642	0.885	-6.37
45	1.701	0.717	-7.51
40	1.701	0.566	-9.55
35	1.701	0.434	-11.87
30	1.701	0.319	-14.55
25	1.701	0.221	-17.72
20	1.701	0.142	-21.59
10	1.701	0.035	-33.63
A	B	C	D

TABLE 2

SPREADSHEET EQUATIONS FOR CLOSED-BOX DATA REDUCTION

	A	B	C	D
1		Measured	Volts Required	
2		Voltage Required	for Constant	
3		for Constant	Cone Excursion	
4	Frequency	Cone Excursion	for Ideal Speaker	dB
5	100	3.54	=B5	=20*LOG10(C5/B5)
6	93	3.21	=$C$5*(A6/$A$5)^2	=20*LOG10(C6/B6)
7	84	2.758	=$C$5*(A7/$A$5)^2	=20*LOG10(C7/B7)
8	79	2.531	=$C$5*(A8/$A$5)^2	=20*LOG10(C8/B8)
9	70	2.219	=$C$5*(A9/$A$5)^2	=20*LOG10(C9/B9)
10	64	2.038	=$C$5*(A10/$A$5)^2	=20*LOG10(C10/B10)
11	58	1.911	=$C$5*(A11/$A$5)^2	=20*LOG10(C11/B11)
12	55	1.85	=$C$5*(A12/$A$5)^2	=20*LOG10(C12/B12)
13	50	1.842	=$C$5*(A13/$A$5)^2	=20*LOG10(C13/B13)
14	45	1.701	=$C$5*(A14/$A$5)^2	=20*LOG10(C14/B14)
15	40	1.701	=$C$5*(A15/$A$5)^2	=20*LOG10(C15/B15)
16	35	1.701	=$C$5*(A16/$A$5)^2	=20*LOG10(C16/B16)
17	30	1.701	=$C$5*(A17/$A$5)^2	=20*LOG10(C17/B17)
18	25	1.701	=$C$5*(A18/$A$5)^2	=20*LOG10(C18/B18)
19	20	1.701	=$C$5*(A19/$A$5)^2	=20*LOG10(C19/B19)
20	10	1.701	=$C$5*(A20/$A$5)^2	=20*LOG10(C20/B20)

DRIVER PARAMETERS FOR SPEAKER RS 40-1021B 8"

Speaker Compliance Volume(Vas): 2.55 cu. ft. = 72.22 liters

Speaker Resonant Frequency (Fs): 30.0 Total Q (Qts): 0.41

SPEAKER BOX PARAMETERS:

	Box Design #1	Box Design #2	Optimized Bass Reflex	Optimized Closed Box
Box Volume in Cu. Ft.:	3	0	2.69	1.29
Alpha (Vas/Vb):	.85	0	.95	1.97
OPTIMUM tuning Freq. in Hz:	28.52	None	29.65	None
ACTUAL tuning Freq. in Hz:	28	None	29.65	None
Tuning Ratio h (Fb/Fs):	.93	None	.99	None
System Q:	7	0	7	.71
Duct Diameter in Inches:	2	None	2	None
Duct Length in Inches:	2.13	None	2.11	None
Approx. -3 dB Frequency:	- -	- -	29.26	51.22

FIGURE 7: Design parameters for optimum bass-reflex, closed-box, and 3 cubic foot enclosures using Radio Shack's 40-1021B 8" driver.

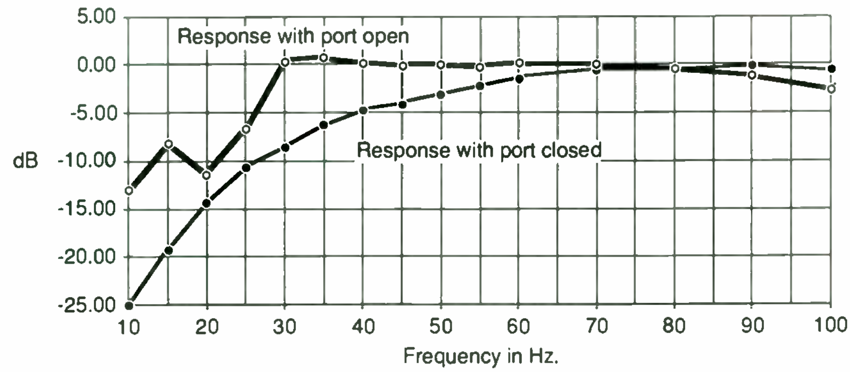


FIGURE 8: Bass-reflex response using the 40-1021B driver.

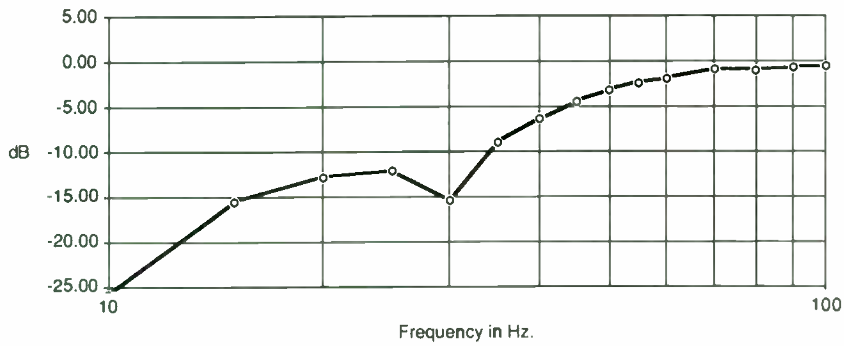


FIGURE 9: Measured 40-1021B driver response in 3 cubic foot bass-reflex enclosure.

TABLE 3

MEASUREMENTS OF BASS REFLEX*

Frequency	Measured Speaker Voltage Required for Constant Cone Excursion	Volts Required for Constant Cone Excursion for Ideal Speaker	Enclosure Response with port closed in dB	Change in dB with port open	Total Bass Reflex Enclosure Response
200	4.530	4.530	0.00	1.00	1.00
160	2.891	2.899	0.02	0.00	0.02
140	2.151	2.220	0.27	2.55	2.82
120	1.552	1.631	0.43	1.00	1.43
100	1.216	1.133	-0.62	-2.00	-2.62
90	0.932	0.917	-0.14	-1.00	-1.14
80	0.765	0.725	-0.47	0.00	-0.47
70	0.590	0.555	-0.53	0.50	-0.03
60	0.479	0.408	-1.40	1.50	0.10
55	0.444	0.343	-2.25	2.00	-0.25
50	0.402	0.283	-3.04	3.00	-0.04
45	0.367	0.229	-4.08	4.00	-0.08
40	0.314	0.181	-4.78	5.00	0.22
35	0.286	0.139	-6.28	7.00	0.72
30	0.276	0.102	-8.65	9.00	0.35
25	0.243	0.071	-10.71	4.00	-6.71
20	0.237	0.045	-14.37	3.00	-11.37
15	0.233	0.025	-19.22	11.00	-8.22
10	0.204	0.011	-25.11	12.00	-13.11
A	B	C	D	E	F

* 3 cubic foot enclosure using Radio Shack's 40-1021B 8" woofer using microwave transceiver with port closed and sound-level meter differences with port open.

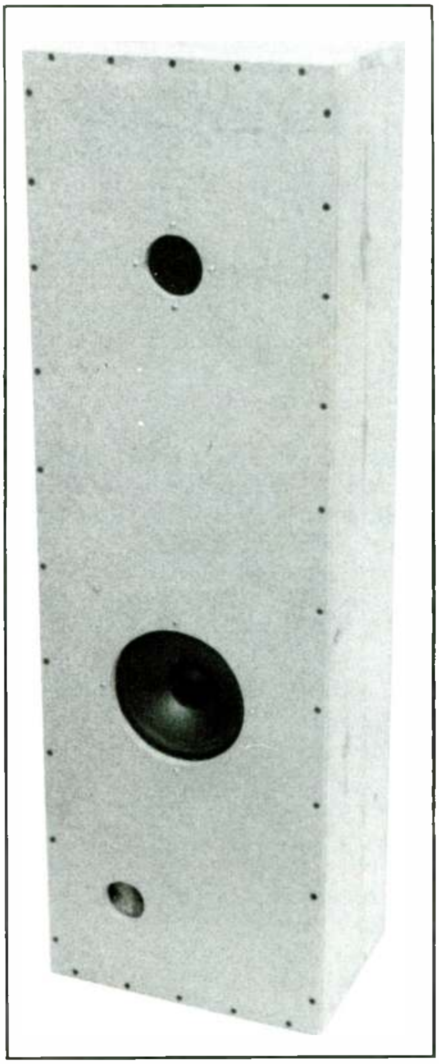


PHOTO 3: 3 cubic foot bass-reflex enclosure using Radio Shack's 40-1021 8" driver.

quency. *Table 2* shows the spreadsheet equations used in constructing *Table 1.* The equations in column C take into account the square law. *Figure 5* is a graph showing the measured response of the 1 cubic foot closed-box enclosure. As you can see, this response closely agrees with the predicted response of *Fig. 2.*

BASS-REFLEX DESIGN AND TEST.

I also designed the bass-reflex enclosure using Old Colony's MacSpeakerBox program. *Figure 6* shows the frequency response plots for optimum bass-reflex, closed-box, and 3 cubic foot enclosures based on measured Thiele/Small parameters and *Fig. 7* lists the various design parameters of each.

Photo 3 is the partially finished bass-reflex enclosure, constructed for my youngest daughter who asked for "tall" enclosures. Although you might expect enclosure resonances due to the tall construction, the boxes sound good with excellent low-frequency response as pre-

TABLE 4

SPREADSHEET EQUATIONS FOR BASS-REFLEX DATA REDUCTION

	A	B	C	D	E	F
1		Measured Speaker	Volts Required	Enclosure		Total
2		Voltage Required	for Constant	Response	Change in dB	Bass Reflex
3		for Constant	Cone Excursion	with port	with port	Enclosure
4	Frequency	Cone Excursion	for Ideal Speaker	closed in dB	open	Response
5	200	4.53	=B5	=20*LOG10(C5/B5)	1	=E5+D5
6	160	2.891	=$C$5*(A6/$A$5)^2	=20*LOG10(C6/B6)	0	=E6+D6
7	140	2.151	=$C$5*(A7/$A$5)^2	=20*LOG10(C7/B7)	2.55	=E7+D7
8	120	1.552	=$C$5*(A8/$A$5)^2	=20*LOG10(C8/B8)	1	=E8+D8
9	100	1.216	=$C$5*(A9/$A$5)^2	=20*LOG10(C9/B9)	-2	=E9+D9
10	90	0.932	=$C$5*(A10/$A$5)^2	=20*LOG10(C10/B10)	-1	=E10+D10
11	80	0.765	=$C$5*(A11/$A$5)^2	=20*LOG10(C11/B11)	0	=E11+D11
12	70	0.59	=$C$5*(A12/$A$5)^2	=20*LOG10(C12/B12)	0.5	=E12+D12
13	60	0.479	=$C$5*(A13/$A$5)^2	=20*LOG10(C13/B13)	1.5	=E13+D13
14	55	0.444	=$C$5*(A14/$A$5)^2	=20*LOG10(C14/B14)	2	=E14+D14
15	50	0.402	=$C$5*(A15/$A$5)^2	=20*LOG10(C15/B15)	3	=E15+D15
16	45	0.367	=$C$5*(A16/$A$5)^2	=20*LOG10(C16/B16)	4	=E16+D16
17	40	0.314	=$C$5*(A17/$A$5)^2	=20*LOG10(C17/B17)	5	=E17+D17
18	35	0.286	=$C$5*(A18/$A$5)^2	=20*LOG10(C18/B18)	7	=E18+D18
19	30	0.276	=$C$5*(A19/$A$5)^2	=20*LOG10(C19/B19)	9	=E19+D19
20	25	0.243	=$C$5*(A20/$A$5)^2	=20*LOG10(C20/B20)	4	=E20+D20
21	20	0.237	=$C$5*(A21/$A$5)^2	=20*LOG10(C21/B21)	3	=E21+D21
22	15	0.233	=$C$5*(A22/$A$5)^2	=20*LOG10(C22/B22)	11	=E22+D22
23	10	0.204	=$C$5*(A23/$A$5)^2	=20*LOG10(C23/B23)	12	=E23+D23

dicted and measured. As with the 1 cubic foot closed-box design, I used 9¼″ by ¾″ board for the sides and particle board for the front and back. Overall dimensions for the enclosure are 48″ by 15″ by 10½″.

This technique for measuring bass-reflex response is similar to that of the closed box, and, for the microwave measurements, the port is temporarily sealed. Measurement steps for determining bass-reflex response are as follows:

1. Seal the port so the enclosure acts as a closed box.
2. Starting with the highest frequency, adjust the signal generator or amplifier for a good signal as viewed on the scope using the test setup as illustrated in *Fig. 3.*
3. Measure the voltage across the speaker terminals and record the frequency and voltage. Be careful not to disturb the transceiver's alignment relative to the speaker once you start taking these measurements.
4. Repeat steps 2 and 3 for each frequency you will measure.
5. Position a sound-level meter 3–4′ away from the speaker. Depending on room acoustics, you may have to adjust this distance.
6. Using the same set of frequencies as in steps 2–4, measure the difference in decibels using a sound-level meter with the port sealed and with it open. A book works well for sealing the port momentarily. If the sound-level meter reads higher with the port open, record this as a positive number; if lower, as a negative number.

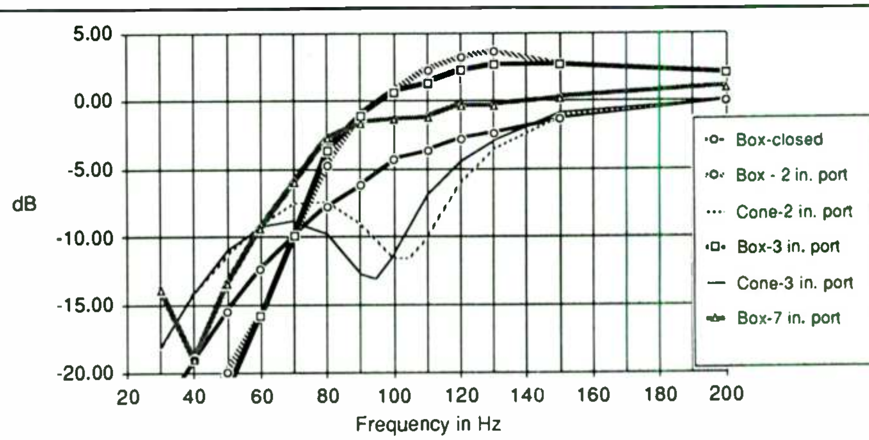


FIGURE 10: Response of 40-1284C driver for various port lengths.

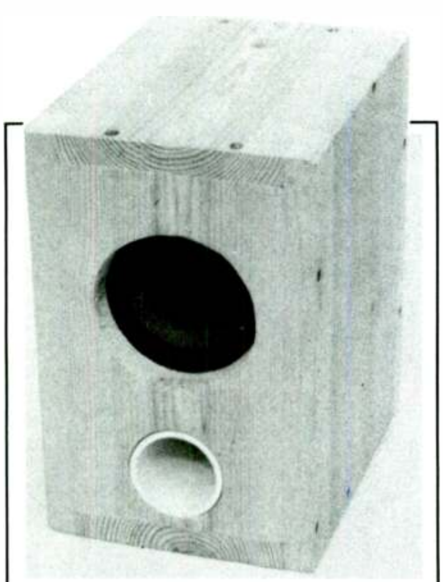


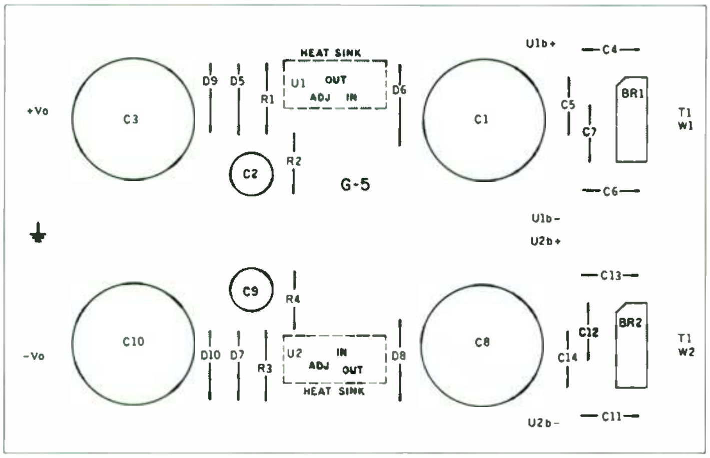
PHOTO 4: 0.22 cubic foot bass-reflex enclosure using Radio Shack's 40-1284C 5″ speaker.

Since a difference in decibels is measured at the same frequency, low-frequency response errors of the sound-level meter are cancelled. You may have to experiment with this step to cancel room effects. Fortunately, the change in decibels is smaller at the higher frequencies where standing waves are closer together.

Table 3 lists the tabulated measurements and calculations for the bass-reflex design. Between 100 and 200Hz, the measured voltage is close to the calculated voltage for the ideal speaker using the same starting voltage at 200Hz. As the frequency decreases, the measured voltage increases relative to the ideal speaker. Since the bass-reflex box with the port sealed is rolling off at a nearly constant rate, the measured voltage between 10

Continued on page 17

KIT SPECIALS FOR SPRING!



PREAMP POWER SUPPLY — **KG-5**
Gary Galo/Linear Technology — **$195**

This low-impedance, dual-polarity power supply uses the excellent Linear Technology LT1033CT and LT1085CT 3-Amp IC regulators. The regulators feature low dropout voltage, which helps maintain regulation even in situations where line voltage is low. Although the parts supplied provide ± 15 volts, the output can be varied from ± 12 to ± 18 volts by changing two resistors. An oversized Avel-Lindberg 120VA toroidal transformer will operate many KG-5 regulator boards, allowing separate regulation of the left and right channel circuitry, or separate regulation of a phono preamp and line level stages. The printed circuit board has heavy traces on the input and output lines to minimize resistance. Case not included.

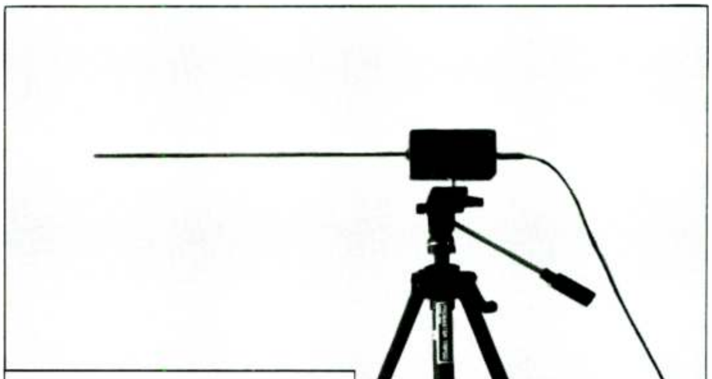
The KG-5 is ideal for many low-level applications, including preamplifiers and electronic crossovers. Gary Galo also uses the KG-5 to power the analog circuitry of his CD player. From *TAA* 4/90.

Purchasing options available:

PCBG-5 Printed Circuit Board Only	**$ 15.00**
PCB and Parts Only (less Transformer)	**$134.95**
Transformer Only	**$ 64.95**
Complete Kit	**$195.00**

AOCOM POWER SUPPLY REGULATOR — **KY-2**
Kit Ryan — **$99 per chan.**

This popular mod was designed for taming Adcom's GFA-555, but it adds sweetness and definition to just about any amp in the 80V-in, 60V-out, 10A-regulated family. Mounts in existing case; complete with PCB, custom heatsink, and Japanese transistors. Two usually needed. From *TAA* 4/89.



MITEY MIKE SPECIFICATIONS

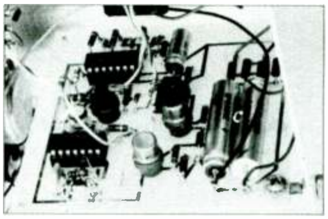
Response (rel. 1kHz)	±1dB, 20Hz–10kHz ±2dB, 10kHz–20kHz −3dB @ 3Hz and 25kHz
Sensitivity	39mV/Pa, ±2dB
Max Undistorted SPL	> 120dBA
Wideband Noise Level	< 42dBA
Power Consumption	5mW typ. / 7mW max.

MITEY MIKE TEST MICROPHONE — **KD-2**
Joseph D'Appolito — **$149**

At long last, a top-quality test mike at a great price! Mitey Mike is the answer to many an audiophile's loudspeaker testing dreams, providing flat free-field frequency response; high, undistorted SPL capability; good measurement repeatability; and guaranteed long-term stability. Plus, it's a snap to build! Complete with PCB, mike cartridge, custom brass wand, all components, and (undrilled) blue case. Typical specs are shown in the inset. For greater accuracy, a mike calibration service is available. Tripod, 9V battery not included. From *SB* 6/90.

THE WARBLER OSCILLATOR — **KK-3**
Richard Crawford — **$69**

The Warbler produces a swept signal covering any ⅓ octave between 16Hz and 20kHz. With a mike, it's a pink noise source to evaluate speaker system performance and the effects of reflection and absorption. Comes with PCB, transformer, all parts. Case not included. From *TAA* 1/79.



MITEY MIKE/WARBLER SET — **KMW-1**
D'Appolito/Crawford — **$199**

Both of the above, at a special savings of $19!

Yes! Please send me

____ # ______ @ $ ____ = $ ______
____ # ______ @ $ ____ = ______
____ # ______ @ $ ____ = ______
SHIPPING ______
TOTAL $ ______

SHIPPING: Postpaid in U.S. In Canada, please add 10% for surface; others, 20% for surface. Airmail: Please inquire.

OLD COLONY SOUND LAB

OLD COLONY SOUND LAB
PO Box 243, Dept. B91
Peterborough, NH 03458 USA
24-hour telephone: (603) 924-9464
24-hour FAX: (603) 924-9467

NAME ______ MAGIC NO ______
STREET ______
CITY ______ STATE ____ ZIP ____
MC/VISA ______ EXP DATE ____

CALL OR WRITE FOR YOUR FREE CATALOG!
DEALER INQUIRIES INVITED.

Continued from page 15

and 30Hz is more constant. *Table 3* is the same as *Table 1* for the closed-box design, but with the addition of the two columns for the change in decibels when the port is open and closed. *Table 4* shows the formulas used in the *Table 3* spreadsheet.

Figure 8 plots the data shown in *Table 3*. Notice its resemblance to the predicted response of *Fig. 6*. A curious peak in the measured response occurs around 15Hz and seems to be real since the speaker visibly makes a longer excursion in this frequency range with the port open, but not with the port sealed. I have noticed this on other bass-reflex systems, but not on closed-box systems.

If you use the microwave transceiver approach to measure driver cone response on the bass-reflex enclosure with the port open, you can directly measure the frequency to which the system is tuned since the driver cone excursion is at its minimum at the tuned frequency. *Figure 9* is a plot at 5Hz increments showing the cone deviation for the bass-reflex enclosure. The notch clearly shows the system is tuned to approximately 30Hz. Rather than make a series of measurements to check the tuning, sweep the generator about the desired range and watch for a minimum from the transceiver on the scope.

Figure 10 illustrates the effect of various port lengths on overall box response, as well as on tuning frequency as evidenced by nulls in the driver cone response. I constructed the enclosure in *Photo 4* using ¾″ thick pine with inside dimensions of 5½″ by 9½″ by 7½″, yielding an inside volume of 0.22 cubic foot. The various ports had an inside diameter of 2″. I designed the enclosure using Radio Shack's 5″ speaker, part number 40-1284C. At $10.95, it represents a low-cost, reasonable-sounding driver for small single-speaker enclosures.

The Radio Shack catalog does not provide Thiele/Small parameters, perhaps due to variation in characteristics; however, a magnet size of 9.8oz. relative to the cone diameter indicates reasonable parameters. Measured values on the speaker I tested are V_{AS} = 0.22 cubic foot, f_S = 96Hz, and Q_{TS} = 0.4. The thin solid and dashed lines of *Fig. 10* show the cone responses for 3″ and 2″ long ports, which indicate tuning frequencies of approximately 93 and 103Hz. The bold lines represent overall enclosure response as a function of various port lengths. The bold plot for the response with the 7″ port is starting to rise, giving evidence of the previously mentioned low-frequency resonance.

CONCLUSIONS. Although I have suggested using the microwave transceiver for measuring woofers, you can extend the technique to the midrange and perhaps to the tweeters if you have the measurement sensitivity available (remember the square law effect). You may need a preamplifier on the transceiver to measure the midrange.

An important point to remember is that this technique is good only for the low-frequency response of a given speaker where the cone functions as a piston. At higher frequencies, the cone may be deforming, so the voice coil is making the deviation, but the cone material may not. This is precisely why adding a whizzer cone to a speaker increases its high-frequency response.

Although I have presented only closed-box and bass-reflex systems, this technique may also be useful for transmission line and horn systems to measure cone loading.

ACKNOWLEDGMENT

I wish to thank my daughters, Heather and Michelle, for their assistance in providing the photographic work for this article.

JIMMY NEEDS DEEP BASS

BY JERRY L. McNUTT and MIKE AUGHTMAN

Mike needed a subwoofer in his GMC Jimmy—a very fast and accurate sub to augment his Soundstream/KEF system and one that was as small as possible. He was not concerned with ear-shattering volume levels. With these criteria in mind, we designed and built a dual-channel Isobarik sub.

We chose SEAS 8″ woofers because their performance-to-price ratio is very high. They have cast magnesium baskets, long-throw rubber surrounds, and high power voice coils.

We decided to give each pair of drivers 0.8 cubic feet. You cannot simply halve the optimum single woofer air space to get a corresponding Isobarik design. For best results, tight and articulate bass, we cut the volume by one-third. We also used "clamshell" or front-to-front woofer mounting because it gives the tightest coupling between the drivers and all even-order woofer harmonics cancel.

THE BOX. The partially completed enclosure, made out of ¾″ medite, is shown in *Photo 1*. The box's lid is removable so you can load and service the inner woofers. The framework on each end protects and hides the outer woofers. We left two sides of the end caps open to let the sound radiate through and we used an angled divider to break up standing waves and stiffen the enclosure.

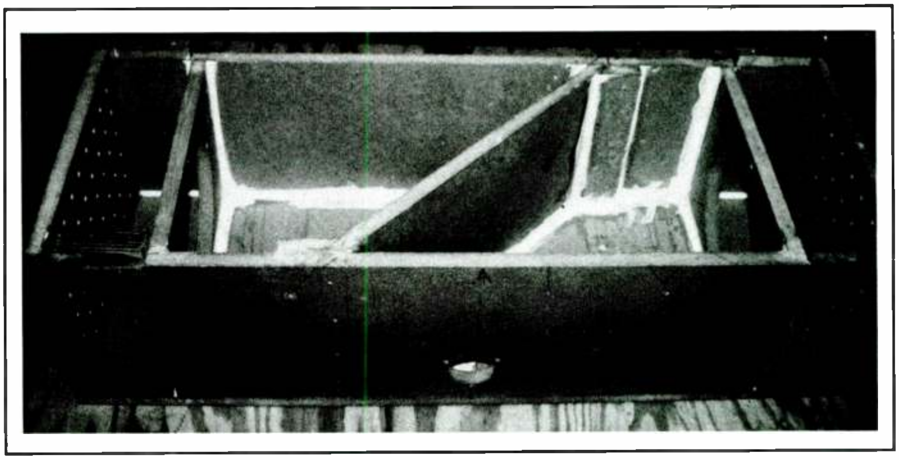


PHOTO 1: The partially completed enclosure.

After we sealed the enclosure with silicone caulk, we painted all interior panels and some of the exterior ones with four heavy coats of Magic Paint. *Photo 2* shows the enclosure ready to be sanded, before the Magic Paint arrived. After we sanded and painted the enclosure, we loaded and wired the drivers with AudioQuest wire.

To seal the lid, we used black windshield caulk—great stuff. It sticks to anything, always makes an airtight seal, and is very pliable. The best part is it will let you get back into the enclosure. You can buy it at most windshield repair shops.

DOUBLE DUTY. Mike wished to be able to remove the sub from his truck and

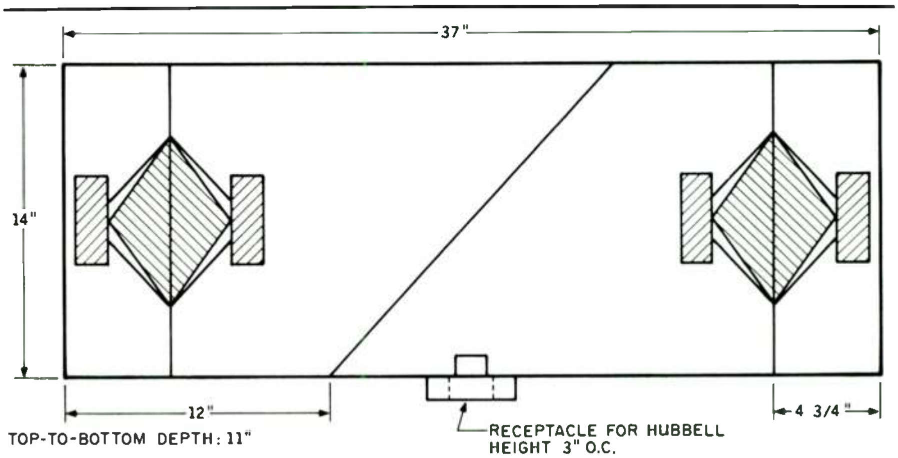


FIGURE 1: A top view of the outside dimensions of the enclosure with the lid off.

ABOUT THE AUTHORS

Jerry L. McNutt received an electrical engineering degree from Auburn University. He is employed by Choctaw Electronics Enterprises, who manufacture Premium OEM loudspeakers for Chrysler, Ford, and Peavey. He has a ten-year background in audio, including two years of running his own loudspeaker production company and six years in custom home stereo installation.

Mike Aughtman received an industrial engineering degree from Auburn University. He is employed by Mizuno America, who make the word's finest golfing equipment. He has been interested in home and car hi-fi for nearly six years. He worked at Accurate Audio (local high-end stereo shop) during his matriculation at Auburn.

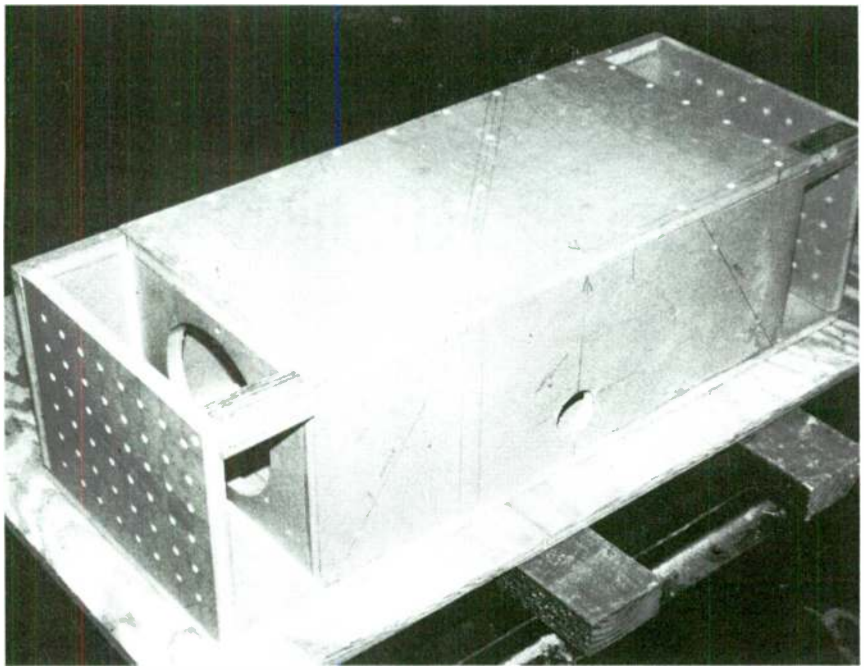


PHOTO 2: The enclosure ready to be sanded.

replace it quickly, so we had to find the right connector system. At an electrical supply store, we found a male chassis-mount, three-phase locking connector manufactured by Hubble Corporation. It has four 30A connections, making it perfect for two-channel operation. We put this on the enclosure (mounted in a 2½" diameter hole).

On the wire end, from the bass power amp, we attached an appropriate female body-locking connector. This system never vibrates loose, is easy to undo, and makes a superb electrical connection.

For the final touches, we painted the two ends flat black and wrapped the enclosure in thin black automotive carpet. The painted ends did not need carpeting because they are covered by the Jimmy's rear wheel wells. We also added a pair of musical instrument handles to help in loading and unloading the subwoofer enclosure. *Photo 3* shows the sub ready to be loaded into Mike's Jimmy.

The sub is electronically crossed over at 80Hz by a Soundstream SX-1 and is driven by a Linear Power 2202 power amp. The front and rear KEF satellite speaker pairs are crossed over at 125Hz by passive crossovers built into the modified Soundstream D-200 and Class A-50 power amps. The crossover gap is intentional to compensate for a resonance in the Jimmy of around 100Hz.

HARDWARE ROSTER. The system's source is a Soundsteam TC-308. All system interconnects are MIT's PC Squared. The KEF drivers are hooked up with Monster Cable's Power Line 3 and the subwoofer with AudioQuest Type 8. All power and ground wires are 8-gauge stranded copper by Phoenix Gold.

The music this system reproduces is stunning. The subwoofer blends in perfectly. Low bass response has been extended to well below 35Hz. The KEF speakers sound so much better now that they do not have to play low bass. This proves again that adding a subwoofer to a system adds more than just low bass; it improves the critical midrange frequencies as well.

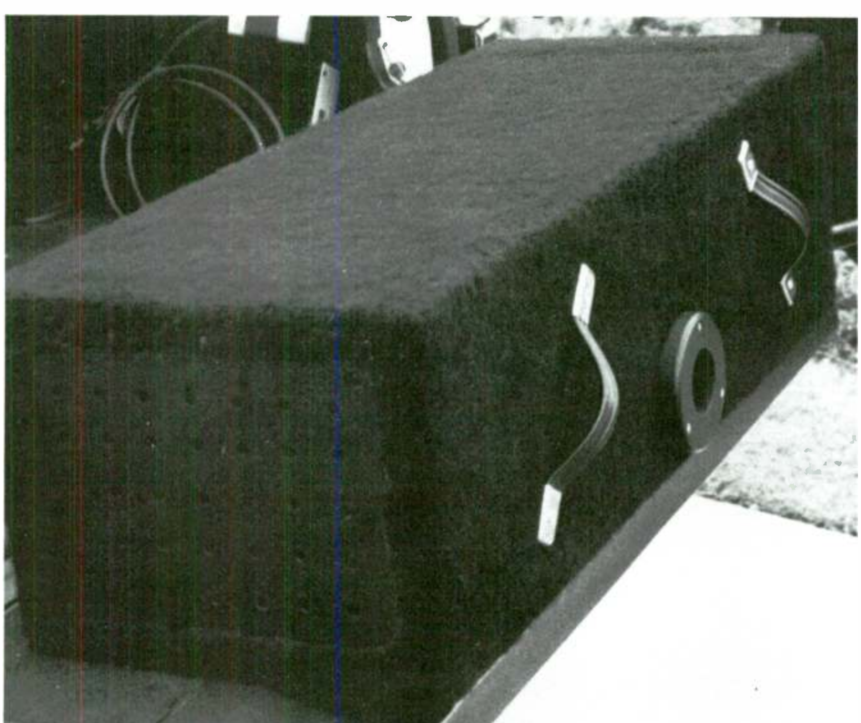


PHOTO 3: The sub about to be loaded into Mike's Jimmy.

CALCULATING LF RESPONSE WITH PSpice

BY ROY MALLORY

The standard way to analyze speaker systems' small-signal behavior is to model the speaker and its cabinet as electrical components, then use circuit-analysis techniques to determine the system's response function. It's easy to translate a speaker design into an equivalent circuit; calculating the response, however, is cumbersome.

You can derive and solve the equation for an enclosure's frequency response, but it varies for each type of enclosure. And, what if you wished to solve for the transient or phase response? The answer is simple: use a circuit-simulation software package such as PSpice (reviewed in *TAA* 1/91, p. 28). Within minutes after you describe a speaker system's equivalent electrical circuit, PSpice will provide you with high-quality graphs of frequency response, phase response, transient response, and cone excursion versus frequency.

PSpice was derived from SPICE (simulation program with integrated circuit emphasis), a public-domain software package developed in the 1970s by researchers at the University of California at Berkeley. This software was developed primarily as an aid to integrated-circuit design, hence its acronym.

Several companies sell versions of SPICE that operate on personal computers. The most successful of these programs is PSpice, which is sold by the MicroSim Corporation. PSpice is available for several computers, including IBM PC compatibles and the Macintosh II. A student version of the software is also available.[1]

You can get a more complete package with up-to-date software directly from MicroSim. They sell an evaluation version of their software for $75. The package includes the Tuinenga[1] book, the most recent revision of their software, and a complete user's manual. If you expect to use PSpice enough to justify the extra cost, this package is an excellent value. MicroSim's phone number is (714) 770-3022.

ABOUT THE AUTHOR

Roy Mallory is a graduate of Tufts University with a degree in electrical engineering. He is employed as a Principal Engineer at a small high-tech company in Massachusetts where his focus is analog circuit design, but where he also contributes in the areas of transducer and software design. He has had a long-standing interest in audio and has completed many projects including preamps, amps, test equipment, and a few speaker systems.

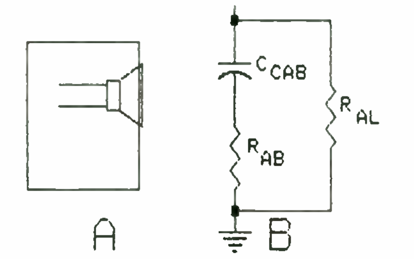


FIGURE 1: The simple enclosure (a) diagram and (b) circuit diagram with losses.

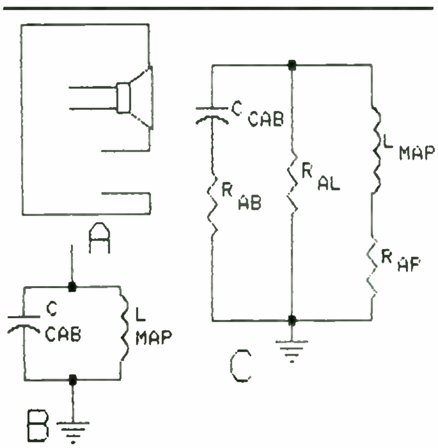


FIGURE 2: The ported enclosure (a) diagram, (b) circuit diagram, and (c) circuit diagram with losses.

CIRCUIT ELEMENTS. To predict loudspeaker performance with PSpice, you must be able to model a loudspeaker system as an electrical circuit. I will describe the simple rules and equations that enable you to do so. For an understanding of the theory behind the models, read Beranek's[2] thorough discussion of the subject.

First, let's look at how we model each component in a loudspeaker system. Then we will see how to combine these components to make most of the usual types of enclosures: acoustic-suspension, ported, passive-radiator, infinite-baffle, Isobarik, and bandpass (two types). In *Table 1* I summarize this information; the formulas use MKS units (meters, kilograms, seconds), but you can use the values in *Table 2* to convert to more familiar units. Also, *Table 3* is a glossary of the symbols I used.

Square One. A simple enclosure is a closed box with a single hole for mounting the loudspeaker (*Fig. 1a*). For low-frequency analysis, you can model such an enclosure as a pure compliance (as having springiness). The electrical analog is a capacitor, one of whose terminals is connected to ground. The larger the enclosure, the larger the compliance and the capacitance will be.

An infinite baffle (an infinitely large enclosure) is a special case of the simple enclosure. As such, you can model it as an infinite capacitance—a short to ground. The following equation relates the capacitance to the enclosed volume:

$$C_{CAB} = V_{AB} / (P_o C^2)$$

Where P_o is the density of air and c is its velocity.

To model a simple enclosure with losses requires the addition of two resis-

TABLE 1

ELEMENTS, CIRCUITS, AND FORMULAS

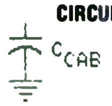
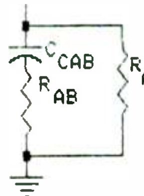
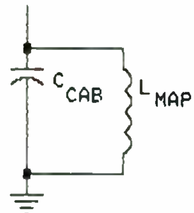
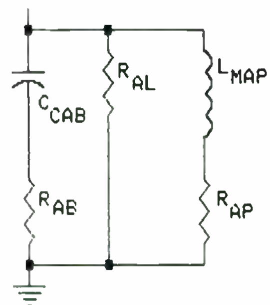
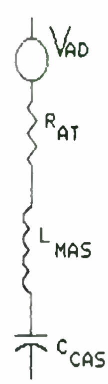
ELEMENT	EQUIVALENT CIRCUIT	FORMULA (MKS UNITS)
Simple enclosure		$C_{CAB} = \frac{V_{AB}}{P_o C^2}$
Simple enclosure with losses		R_{AL} = Loss due to leakage R_{AB} = Loss due to internal damping
Ported enclosure		$f_B = \frac{1}{2\pi\sqrt{L_{MAP} C_{CAB}}}$ $L_{MAP} = \frac{P_o (L + 0.85r + 0.61r)}{\pi r^2}$ 0.85r is the effective length of the flush end, 0.61r is the effective length of the free end, and r = $\sqrt{2/\pi}$ for noncircular ports
Ported enclosure with losses		$Q_L = R_{AL} \sqrt{C_{CAB} / L_{MAP}}$ $Q_A = \frac{1}{R_{AB} \sqrt{C_{CAB} / L_{MAP}}}$ $Q_P = \frac{1}{R_{AP} \sqrt{C_{CAB} / L_{MAP}}}$ $Q_B = \frac{1}{\frac{1}{Q_L} + \frac{1}{Q_A} + \frac{1}{Q_P}}$
Speaker		$C_{CAS} = \frac{V_{AS}}{P_o C^2}$ $L_{MAS} = \frac{1}{(2\pi f_S)^2 C_{CAS}}$ $R_{AT} = \frac{1}{2\pi f_S Q_{TS} C_{CAS}}$ V_{AD} normalization: $V_{AD} = 2\pi L_{MAS}$ (AC analysis) $V_{AD} = L_{MAS} / 2$ (transient analysis) $V_{AD} = 1 / 2\pi C_{CAS}$ (cone excursion)
Passive radiator		(Use same formulas as for speaker.)

TABLE 2

CONVERSION FACTORS

Length
- 1 inch = 0.00254 meter
- 1 foot = 0.305 meter

Area
- 1 square foot = 0.0920 square meter
- 1 square inch = 6.45×10^{-4} square meter

Volume
- 1 liter = 0.001 cubic meter
- 1 cubic foot = 0.0283 cubic meter
- 1 cubic inch = 1.64×10^{-5} cubic meter

TABLE 3

SYMBOL DEFINITIONS

c	Velocity of sound in air (345m/sec)
C_{CAB}	Acoustic compliance of enclosure (m^5/newton)
C_{CAR}	Acoustic compliance of passive radiator suspension
C_{CAS}	Acoustic compliance of speaker suspension
f	Frequency (Hz)
f_r	Resonant frequency of passive radiator
f_s	Resonant frequency of speaker
l	Length (m)
L_{MAP}	Acoustic mass of port (kg/m^4)
L_{MAR}	Acoustic mass of passive radiator cone including air
L_{MAS}	Acoustic mass of speaker cone including air load
Q	Ratio of resistance to reactance (parallel circuit) or ratio of reactance to resistance (series circuit)
Q_A	Q of cabinet due to internal energy absorption
Q_B	Total Q of cabinet
Q_L	Q of cabinet due to leakage losses
Q_P	Q of cabinet due to port losses
Q_{TR}	Total Q of passive radiator
Q_{TS}	Total Q of speaker
R_{AB}	Acoustic resistance of enclosure losses due to internal energy absorption (newton-sec/m^3)
R_{AL}	Acoustic resistance of enclosure losses due to leakage
R_{AP}	Acoustic resistance of port losses
R_{AR}	Acoustic resistance of passive radiator losses
R_{AT}	Total acoustic resistance of speaker losses
S	Area (m^2)
V_{AB}	Enclosure volume (m^3)
V_{AD}	Normalized acoustic pressure output of speaker
V_{AS}	Volume of air having same acoustic compliance as driver suspension
P_o	Density of air ($1.18kg/m^3$)

tors as shown in *Fig. 1b*. R_{AL} represents losses due to leaks in the enclosure and R_{AB} represents losses within the enclosure, typically due to stuffing material. (I will discuss adding stuffing to a simple enclosure when I describe acoustic suspension speaker models.)

With Holes. A ported enclosure has two openings: one for the speaker and one for a vent or port (*Fig. 2a*). It can be represented as a compliance (the enclosed volume) and a mass (the port). The compliance is modeled exactly as for a simple enclosure and the mass is modeled as an inductor. The two components are in parallel; one connection goes to ground as shown in *Fig. 2b*.

As described in Beranek's book,[2] the equivalent inductance of the port is:

$$L_{MAP} = (P_o l') / \pi r^2$$

r represents the port's radius and l′ the effective length, which includes not only the port's length, but end effects as well. Beranek notes that the free end of the port adds effective length by the formula:

$$l = 0.61r$$

The end flush with the cabinet (the so-

FIGURE 3: The speaker circuit diagram.

called flanged end) adds effective length by the formula:

$$l = 0.85r$$

Thus, for most enclosures whose ports have one free end and one flush end, the equation for port length would be:

$$L_{MAP} = (P_o\ (l + 0.85r + 0.61r)) / \pi r^2$$

Solving for l gives you:

$$l = ((\pi r^2\ L_{MAP}) / P_o) - 1.46r$$

As with the simple enclosure, losses in the ported enclosure can be modeled by adding the same two resistors and an additional one to model port losses as shown in *Fig. 2c*. I will discuss these three resistances further when I describe bass-reflex enclosures.

Driver. A speaker is modeled as a voltage source in series with a resistor, capacitor, and inductor, as shown in *Fig. 3*. Either one of this circuit's free terminals represents the cone's back surface; the other represents its front.

The capacitance, C_{CAS}, represents the speaker suspension's acoustic compliance and is calculated by the formula:

$$C_{CAS} = V_{AS} / (P_o\ C^2)$$

The inductance, L_{MAS}, represents the speaker's acoustic mass, including the air load, and is calculated by the formula:

$$L_{MAS} = 1 / (C_{CAS}\ (2\pi f_s)^2)$$

The resistor, R_{AT}, represents the acoustic resistance of all driver losses and is calculated by the formula:

$$R_{AT} = 1 / (2\pi f_s\ Q_{TS}\ C_{CAS})$$

The voltage source, V_{AD}, represents the acoustic pressure the speaker's magnetic circuit provides. Since our concern is only the speaker system's output relative to frequency (not efficiency), the value of V_{AD} is unimportant. However, it is customary to normalize the graphs to a simple value. Small,[3] when deriving the efficiency expression for direct-radiator speaker systems, broke it into two main terms, one containing the frequency-invariant information and the other the frequency-response function. Small manipulated his expression so the frequency-response term equals 1 in the speaker's passband. To follow this precedent, use the formula:

$$V_{AD} = 2\pi\ L_{MAS}$$

You can test the transient response by feeding the modeled speaker system a step. The formula shown below causes a critically damped or overdamped direct-radiator speaker's output to be a pulse of one unit:

$$V_{AD} = L_{MAS} / 2$$

Normalizing cone excursion versus frequency is more problematical; in no frequency band does the cone excursion reach a steady-state value that is the same for all direct-radiator enclosure types. Two approaches seem sensible: not to normalize at all or to normalize to 1, the steady-state value that an unloaded speaker (infinite baffle, bass reflex) reaches at low frequencies. The first suggestion obviously requires no further discussion. To accomplish the second, use the formula:

$$V_{AD} = 1 / (2\pi\ C_{CAS})$$

The Interconnected. A passive radiator's enclosure consists of a cabinet with two holes (*Fig. 4*). One contains the driver, and the other the passive radiator. The box is modeled as a compliance, as is the simple enclosure. The passive radiator itself is modeled like a speaker with no voltage source. You calculate the values of its equivalent capacitance, inductance, and resistance exactly as those for a speaker. The cabinet compliance and passive-radiator equivalent circuit are in parallel, and one end of the circuit is connected to ground.

THE PROGRAM. These instructions on using PSpice apply to the IBM PC version. Usage is similar for other computers, but may differ in some details. Also, I assume you have a basic familiarity with your computer, so I will only address issues directly related to using PSpice.


FIGURE 4: The passive radiator enclosure.

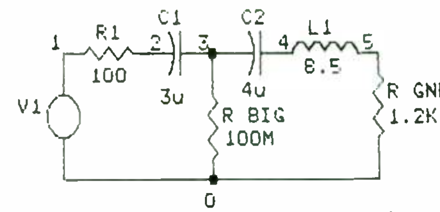


FIGURE 5: Sample circuit for using PSpice.

TABLE 4

EXAMPLE.CIR

```
example 1
*Comment lines start with an asterisk
V1 1 0 AC 1 PULSE(-1V 1V 0SEC 10U
  10U .2SEC .4SEC)
R1 1 2 100  ;Appended comments start
  with a semicolon
C1 2 3 3U
C2 3 4 4U
L1 4 5 8.5
RGND 5 0 1.2K
RBIG 3 0 100MEG
.AC DEC 20 1 1000
.IC V(1)=0
.TRAN .05 .5 UIC
.PROBE
.END
```

To use PSpice, you must have a text editor capable of producing standard ASCII files. Any programmer's editor is ideal; most spreadsheets can also make text files, although not as efficiently.

PSpice is a complex program that would take many pages to describe fully. Therefore, I will describe, mostly by example, only those features needed to model speaker systems.

For PSpice to analyze your circuit, you must describe it to the program. Use the circuit in *Fig. 5* as an example. You must give all components a unique label: resistors (RXXX), capacitors (CXXX), inductors (LXXX), and independent voltage sources (VXXX), where XXX represents alphanumeric characters.

Also, you must number all nodes (a junction between two or more components). These numbers are not required to be contiguous, but the ground node must be 0.

Next, make a text file (EXAMPLE.CIR in *Table 4*) that contains the description of this circuit. (Unless you tell it otherwise, PSpice expects circuit-description files to have a .CIR file extension.) The first line, the title line, may contain anything you wish. The second is an example of an optional comment line.

The next seven lines describe the circuit. The first describes a voltage source named V1 that connects from node 1 to node 0 and is a source for the AC (frequency response) and the transient analyses. "AC 1" tells PSpice the AC analysis should be done with a source whose output is 1.

The next block of text, starting with "PULSE," describes information used in the transient analysis, performed by making V1 a low-frequency square wave. The remaining entries on the line (from left to right) specify the wave: the output starts at −1 and goes to +1, with no delay; it has a rise time of 10μS and a fall time of 10μS, a pulse width of 0.2S, and a period of 0.4S.

The next six items in the circuit file describe passive components. Each line follows the general format of "name node node value." For example, the first describes a resistor named R1 that connects from nodes 1 to 2 and has a value of 100. This line also demonstrates how to add a comment to a line. The value designation can have a multiplier at the end; the most common ones you might wish to use are *MEG* for mega (1×10^6), *k* for kilo (1×10^3), *m* for milli (1×10^{-3}), *u* for micro (1×10^{-6}), and *p* for pico ($1 \times {}^{-12}$).

Following the component descriptions are lines that tell PSpice the kinds of analyses you wish done. The line that starts ".AC" specifies that the frequency response be calculated. "DEC" indicates the frequency is swept logarithmically by decades, the last three numbers specifying a calculation of 20 points per decade and a response from 1Hz–1kHz.

The next line, ".IC", specifies the initial conditions for the transient analysis; that is, the program should consider V(1) to be at 0V when starting the transient calculations. The next line specifies a transient-response analysis of 0.5 seconds, with the results printed every 0.05 seconds (if you wish to have them printed). "UIC" tells PSpice to use the initial conditions specified in the .IC statement.

The next line, ".PROBE," tells PSpice to write a file the Probe graphics post processor will use. This program lets you "probe" various parts of the circuit and graph their voltages and currents. As such, it is much like an oscilloscope and spectrum analyzer rolled into one. The final line in the file simply marks the end of the circuit.

PSpice has a few quirks you may encounter. First, every node in the circuit must have a DC path to ground. If any circuit has a "floating node," placing an arbitrarily large resistor from it to ground will solve the problem. In *Fig. 5*, node 3 shows this problem and its solution.

Also, PSpice will not let you place two inductors in parallel. Placing an arbitrarily small resistor in series with either inductor will solve the problem. Finally, if you have an old version of PSpice and place a carriage return after the ".END" statement, it will give you an error message, but continue to analyze your circuit correctly.

Save the example circuit file in the same directory that contains PSpice, then type *PSPICE EXAMPLE* to run the analysis. After much ruminating, PSpice asks if you wish to see results of the AC or the transient analysis. Choose AC. When the next screen comes up, you will note several optional actions listed at the bottom of the screen. Choose "add trace," then type *v(3)*. If all goes correctly, a plot of the voltage versus frequency at this node will appear.

It is important to note that you can enter a mathematical expression when you add a trace. If you type *v(3)*3*, *v(3)/4.6*, *v(3)/v(2)*, or *v(3)*frequency*, for example, PSpice will graph the result of the expression for you. This feature is important for showing speaker response. Now, press 0 (zero) for older versions of PSpice or E for newer versions to return to the screen that allows a choice between AC and transient analysis.

This time, choose transient. Once again, choose "add trace" and type *v(3)*. A plot of the voltage at node 3 versus time will appear. Now is a good time to familiarize yourself with some of the Probe program's other options. Incidentally, it takes PSpice much longer to perform a transient analysis than an AC analysis. Therefore, if you are interested only in frequency response, omit the ".TRAN" statement in your circuit file or put an asterisk before it. You may also leave out the part of the "V1" statement referring to the transient analysis.

When analyzing this circuit, PSpice created a file called EXAMPLE.OUT. If your circuit file contained errors, error statements would be listed here, as would other information PSpice calculates.

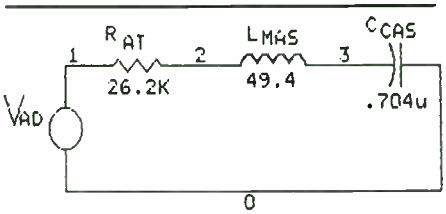


FIGURE 6: Infinite-baffle design data.

TABLE 5

INFINITE.CIR

```
INFINITE BAFFLE DESIGN EXAMPLE
VAD 1 0 AC 310 PULSE(-25V 25V .1SEC
  10U 10U .5SEC 1SEC)
RAT 1 2 26.2K
LMAS 2 3 49.4
CCAS 3 0 .704U
.AC DEC 20 1 1K
.TRAN .05 .2 UIC
.IC V(1)=0
.PROBE
.END
```

Frequency response:
$I(R_{AT})$*FREQUENCY
Phase response:
$IP(R_{AT})+90$
Transient response:
$D(I(R_{AT}))$
Cone excursion:
$I(R_{AT})$/FREQUENCY

OTHER SYSTEMS. In this section, let's look at the equivalent circuits for most of the popular enclosure types and also their PSpice circuit files. For all equivalent circuits, the current through any component represents the so-called volume velocity of the air movement in the part of the speaker system that component represents.

Sound pressure level, however, is proportional to acceleration, not velocity, so the current PSpice calculates must be differentiated to show frequency or transient response. For the frequency response, you can differentiate by multiplying by frequency. Relative cone excursion is the integral of the volume velocity, so the current in the speaker part of the equivalent circuit must be integrated, which can be accomplished by dividing the current by frequency.

For each example, I give the mathematical expressions necessary for graphing the various speaker responses. For the first example, the infinite baffle, I also provide instructions that, for reasons of brevity, I will not repeat for the other examples.

In several of the following examples, the equivalent circuits have floating nodes, as described in the section "The Program." In all such cases, the large resistor connected to the node that provides a DC path to ground is named R_{BIG}. These resistors, of course, are not an inherent part of the speaker's equivalent circuit.

To make the examples more concrete, all represent real alignments for a mythical 8″ woofer with the following parameters: f_s = 27Hz, Q_T = 0.32, and V_{AS} = 99 liters. Let's start by converting these manufacturer's specs into the values needed for this speaker's equivalent circuit. First, convert liters into cubic meters using the conversion factor in *Table 2*.

$$V_{AS} = 99\ (0.001) = 0.099\text{m}^3$$

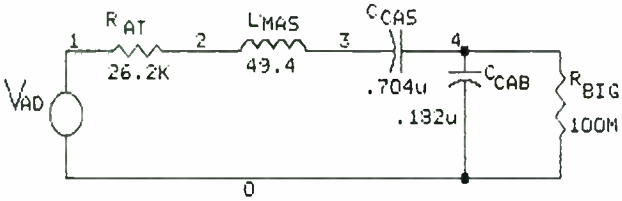


FIGURE 7: Acoustic-suspension design data.

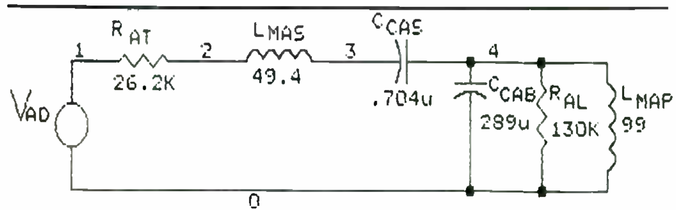


FIGURE 8: Bass-reflex design data.

TABLE 6

PASSIVE.CIR

ACOUSTIC SUSPENSION DESIGN EXAMPLE

```
VAD 1 0 AC 310 PULSE(-25V 25V .1SEC
  10U 10U .5SEC 1SEC)
RAT 1 2 26.2K
LMAS 2 3 49.4
CCAS 3 4 .704U
CCAB 4 0 .182U
RBIG 4 0 100MEG
.AC DEC 20 1 1K
.TRAN .05 .2 UIC
.IC V(1)=0
.PROBE
.END
```

Frequency response:
I(C_{CAB})*FREQUENCY
Phase response:
IP(C_{CAB})+90
Transient response:
D(I(C_{CAB}))
Cone excursion:
I($_{RAT}$)/FREQUENCY

Using the formula from *Table 1*, calculate the acoustic compliance:

$C_{CAS} = (0.099 / (P_o C^2)) = (0.099 / (1.18 (345)^2)) = 0.704 \times 10^{-6}$

Again using *Table 1*, calculate the acoustic mass:

$L_{MAS} = 1 / ((0.704 \times 10^{-6}) (2\pi 27)^2) = 49.4$

Next, calculate the acoustic resistance:

$R_{AT} = 1 / (2\pi 27 (0.32) (0.704 \times 10^{-6})) = 26.2k$

Finally, calculate the equivalent voltage:

$V_{AD} = 2\pi (49.4) = 310$ (AC analysis)

and

$V_{AD} = 49.4 / 2 \approx 25$ (transient analysis)

Now let's start with the simplest example, an infinite baffle.

Big Ones. In an infinite-baffle enclosure, the speaker's front and back are effectively connected to an infinitely large enclosure and are thus shorted to ground, as the circuit in *Fig. 6* shows. To analyze this enclosure, make a text file exactly like the one shown in *Table 5* and name it INFINITE.CIR. Now run PSpice by typing *PSPICE INFINITE* at the DOS prompt.

When PSpice has finished, choose AC analysis, select "add trace," and, as indicated in *Table 5*, type *i(rat)*frequency*. The program will graph the loudspeaker system's frequency response. Now select "remove trace," "all," and "add trace" and type *ip(rat)+90* to graph the phase response. Next, select "remove trace" and "all." Then select "add trace" and type *i(rat)/frequency*1e6* to graph cone excursion versus frequency. If you had not selected "remove trace," PSpice would have graphed the responses on top of each other, an outcome you may occasionally desire.

To view the transient response, back out of this screen by pressing "0" or "e" (depending on which software revision you have) until the screen that allows a choice between the AC and transient analysis appears. Choose transient. Then choose "add trace" and type *d(i(rat))* (which means take the derivative of the current in component R_{AT}).

When you are interpreting your simulated speaker system's transient plot, remember that the equivalent circuits you are building do not accurately predict any loudspeaker's high-frequency response. Consider that for any direct-radiator system, the models presented here predict perfectly flat frequency response extend-

TABLE 7

REFLEX.CIR

BASS REFLEX DESIGN EXAMPLE

```
VAD 1 0 AC 310 PULSE(-25V 25V .1SEC
  10U 10U .5SEC 1SEC)
RAT 1 2 26.2K
LMAS 2 3 49.4
CCAS 3 4 .704U
CCAB 4 0 .289U
RAL 4 0 130K
LMAP 4 0 99
.AC DEC 20 1 1K
.TRAN .05 .2 UIC
.IC V(1)=0
.PROBE
.END
```

Frequency response:
I(C_{CAB})*FREQUENCY
Phase response:
IP(C_{CAB})+90
Transient response:
D(I(C_{CAB}))
Cone excursion:
I(R_{AT})/FREQUENCY

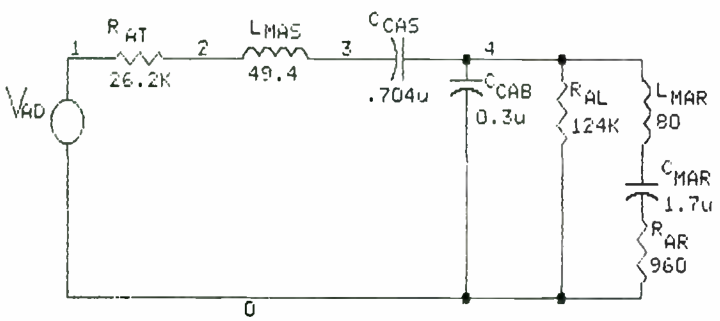


FIGURE 9: Passive-radiator enclosure design data.

TABLE 8

PASSIVE.CIR

PASSIVE RADIATOR DESIGN EXAMPLE

```
VAD 1 0 AC 310 PULSE(-25V 25V .1SEC
  10U 10U .5SEC 1SEC)
RAT 1 2 26.2K
LMAS 2 3 49.4
CCAS 3 4 .704U
CCAB 4 0 .3U
RAL 4 0 124K
LMAR 4 5 80
CCAR 5 6 1.7U
RAR 6 0 960
.AC DEC 20 1 1K
.TRAN .05 .2 UIC
.IC V(1)=0
.PROBE
.END
```

Frequency response:
I(C_{CAB})*FREQUENCY
Phase response:
IP(C_{CAB})+90
Transient response:
D(I(C_{CAB}))
Cone excursion:
I(R_{AT})/FREQUENCY

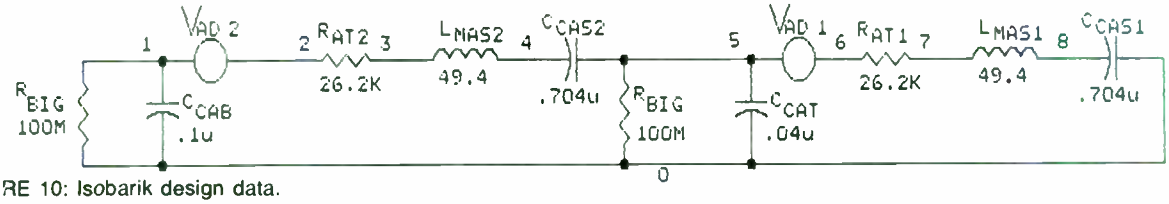


FIGURE 10: Isobarik design data.

TABLE 9

ISOBAR.CIR

ISOBARIK DESIGN EXAMPLE

```
VAD1 5 6 AC 310 PULSE(-25V 25V
  .1SEC 10U 10U .5SEC 1SEC)
VAD2 1 2 AC 310 PULSE(-25V 25V
  .1SEC 10U 10U .5SEC 1SEC)
RAT1 6 7 26.2K
LMAS1 7 8 49.4
CCAS1 8 0 .704U
CCAT 5 0 .04U
RAT2 2 3 26.2K
LMAS2 3 4 49.4
CCAS2 4 5 .704U
CCAB 1 0 .1U
RBIG 1 0 100MEG
RBIG2 5 0 100MEG
.AC DEC 20 1 1K
.TRAN .05 .2 UIC
.IC V(1)=0 V(7)=0
.PROBE
.END
```

Frequency response:
$I(R_{AT1})$*FREQUENCY
Phase response:
$IP(R_{AT1}) + 90$
Transient response:
$D(I(R_{AT1}))$
Cone excursion:
$I(R_{AT1})$/FREQUENCY (external speaker)
$I(R_{AT2})$/FREQUENCY (internal speaker)

ing out forever. Thus, the transient-response plot displays sharp edges that no woofer can really reproduce. Therefore, it is best to use the transient-response graphs to look for low-frequency phenomena such as ringing and try to ignore the impressive crispness of the graphs.

Inner Spring. *Figure 7* and *Table 6* show the pertinent information for the acoustic suspension system. In this type of enclosure, one side of the driver (its front) connects to an infinite baffle, the room, and so is shorted to ground. The other side connects to the enclosure and therefore to an acoustic compliance (capacitance) as shown.

I determined the capacitor's size by selecting V_{AB} from an alignment table and calculating its equivalent compliance from the equation in *Table 1*. I chose a total system Q of 0.707 (maximally flat) for this example. Making the enclosure smaller by reducing the size of C_{CAB} shows how increasing the system Q causes a response peak just before rolloff. Note that in this example, enclosure losses, R_{AB}, have been ignored.

The effect of stuffing on the response and efficiency of acoustic suspension speaker systems is not a trivial topic and is well beyond the scope of this article. For those interested in learning more about it, Small[4] and Bradbury[5] provide some good insights.

Bass Reflex. As *Fig. 8* shows, one side of the speaker connects to the room and is thus shorted to ground, while the other side connects to a ported enclosure whose equivalent component values have been calculated from *Table 1*. Note that the only loss associated with the enclosure that has been modeled is R_{AL}.

Thiele[6] assumed no cabinet losses in deriving his alignments, but experiments by Small[7] indicate that the Q of most bass-reflex speakers lies in a range from about 5 to 20, and that despite its somewhat unexplained (at least by Small) origin, the loss acts as if it were caused by a leak. Thus, R_{AL} is included in the equivalent circuit, and its value is calculated using *Table 1*, assuming a cabinet Q of 7.

A bass-reflex speaker has four branches where you could measure the current or volume velocity. Both Small[7] and Beranek[2] show that the system's sound output is proportional to the current in the circuit's cabinet leg—the current through C_{CAB}. Cone excursion, however, is proportional (not surprisingly) to the current in the driver part of the circuit.

Passive Radiator. As *Fig. 9* shows, the equivalent circuit for a passive-radiator loudspeaker system looks much like a bass reflex, except the passive-radiator cone takes the port's place. As *Table 1* shows, the passive-radiator cone itself is modeled much like an ordinary driver. In fact, you can use an ordinary driver as a passive radiator if you leave its terminals unconnected and calculate its equivalent circuit using Q_{MS} instead of Q_{TS}. Component R_{AL} is once again included, for a cabinet Q, exclusive of the R_{AS} of the passive-radiator cone, of 7.

Isobarik. As *Fig. 10* indicates, an Isobarik speaker system consists of two cones driven in parallel; one is internal, with one surface connected to a simple enclosure, while its other surface is connected to the exterior driver by another, typically small, simple enclosure. In the equivalent circuit, the drivers are represented by two voltage sources with their respective components.

In the PSpice circuit file (*Table 9*), these sources are listed with identical parameters. Thus PSpice will consider the voltages to be identical in amplitude and phase, as the output would be for two

Continued on page 27

TABLE 10

BPASS1.CIR

BANDPASS1 DESIGN EXAMPLE

```
VAD 5 1 AC 310 PULSE(-25V 25V .1SEC
  10U 10U .5SEC 1SEC)
RAT 1 2 26.2K
LMAS 2 3 49.4
CCAS 3 4 .704U
CCAB1 4 0 .075U
RAL1 4 0 160K
LMAP1 4 0 40
CCAB2 5 0 .16U
LMAP2 5 0 110
RAL2 5 0 184K
.AC DEC 20 1 1K
.TRAN .05 .2 UIC
.IC V(1)=0
.PROBE
.END
```

Frequency response:
$(I(L_{MAP1}) + I(L_{MAP2}))$*FREQUENCY
Phase response:
$P(I(L_{MAP1}) + I(L_{MAP2})) + 90$
Transient response:
$D(I(L_{MAP1}) + I(L_{MAP2}))$
Cone excursion:
$I(R_{AT})$/FREQUENCY

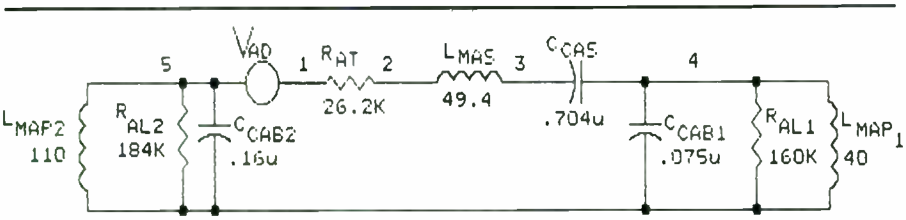


FIGURE 11: Bandpass 1 design data.

OLD COLONY'S USER-FRIENDLY SOFTWARE

Speaker Designer™

Release 1.2 by Stuart E. Bonney

A loudspeaker system design aid and modeling tool for use with both closed and vented systems over the frequency range from 10 to 300Hz. Computes and displays system frequency response, power handling capabilities, and relative sound pressure level (SPL) outputs for each of 26 discrete frequencies over this range. Includes one year support by the author when the user registers this Shareware product. Each **$19.50**

IBM 5¼" 360K DS/DD SPD-1B5

Loudspeaker Modeling Program

by Ralph Gonzalez (*SB* 1, 2, 3/87)

LMP produces a full-range frequency response prediction for multi-way loudspeakers, including the effect of the crossover, driver rolloffs, interdriver time delay, "diffraction loss," etc. (Includes author support.) Each **$17.50**

Specify:

Apple II 5¼" SS/DD (DOS 3.3) LMP-1A5
Apple Macintosh 3½" SS/DD LMP-2M3G
IBM PC/XT/AT 5¼" DS/DD LMP-2B5G
Commodore 64 5¼" DS/DD LMP-1C5

"Souped-Up" LMP provides professional-quality graphics and a fast, friendly, user interface. Remains compatible with LMP data files and is available for IBM PC (CGA, EGA, VGA, or Hercules graphics) and Macintosh computers. The Macintosh version also provides square wave analysis with audible output.

Each **$49.50** (upgrade price: $39.50*)

Specify:

IBM PC/XT/AT 5¼" DS/DD LMP-3B5G
Apple Macintosh 3½" SS/DD LMP-3M3G

*Original LMP disk or sales receipt must be included w/ order.

Driver Evaluation and Crossover Design

by G. R. Koonce (*SB* 5/88)

Disk 1 evaluates the suitability of drivers for closed, vented and passive radiator enclosures, and allows detailed designs of vented boxes.

Disk 2, in addition to driver evaluations, allows the design of first-, second-, and third-order crossovers. 5¼"IBM: 360K, DS/DD. Each **$12.50**

Specify:

Driver Evaluations DRI-1B5
Crossover Design CRS-1B5

Active Filter Design

by Fernando Garcia Viesca (*SB* 4/88)

Calculates component values for Butterworth filters in four configurations: High- and low-pass in second- and third-order. (Includes author support.) Each **$17.50**

IBM 5¼" 360K DS/DD ACT-1B5

Two-Way Active Crossover Design

by Gary Galo (*SB* 5/88)

Performs the calculations for the eight two-way active crossover designs described by Bob Bullock using formulas exactly as given in the articles; plus a program to calculate V_{TH}. (Includes one year user support.) Each **$20**

Specify:

IBM 5¼" 360K DS/DD TWO-1B5
IBM 3½" 720K DS/DD TWO-1B3

Stepped Volume Controls

by Joseph O'Connell (*TAA* 4/88)

These ready-to-run Mac programs come on a 3½-inch SS/DD disk initialized as a 400K disk for compatibility with all machines. Also included are the Pascal source codes, should you wish to customize them for your own use. Program A. Precisely matches the resistor values to the measured or estimated source and load impedances, yielding great accuracy. Your volume control can have 3 to 99 positions. The program will ask you how many dB each step should be attenuated and has provisions for a standard audio taper or any other taper you devise. Program B. Calculates the taper that will result with your actual resistor values, because you are limited to standard values or with series and parallel combinations. It can also show the effects of different source and load impedances on the taper. Both programs (contained on the same disk) allow you to save their output to a text file and include author support via mail. Each **$25**

Apple Macintosh 3½" SS/DD SVC-1M3

BULLOCK & WHITE

BOXRESPONSE

Model-based performance data for either closed-box or vented-box loudspeakers with or without a first- or second-order electrical high pass filter as an active equalizer (*SB* 1/84). *The program disk also contains seven additional programs as follows:*

Air Core: This program was written as a quick way of evaluating the resistance effects of different gauge wire on a given value inductor. The basis for the program is an article in *Speaker Builder* (1/83, pp. 13-14) by Max Knittel. The program asks for the inductor value in millihenries (mH) and the gauge wire to be used. (NOTE: only gauges 16-38.)

Series Notch: Developed to study the effects of notch filters in the schematics of some manufacturers. Enter the components of the network in whole numbers (i.e., 10 for 10μF and 1.5 for 1.5mH) and indicate whether you want one or two octaves on either side of resonance. Output is frequency, phase angle and dB loss.

Stabilizer 1: Calculates the resistor-capacitor values needed to compensate for a known voice coil inductance and driver DC resistance.

Optimum Box: A quick program based on Thiele/Small to predict the proper vented box size, tuning and −3dB down point. It is based only on small signal parameters, therefore, it is only an estimate of the response at low power (i.e., limited excursion).

Response Function: Calculates the small signal response curve of a given box/driver combination after inputting the free-air resonance of the driver (f_S), the overall "Q" of the driver (Q_{TS}), the equivalent volume of air equal to the suspension (V_{AS}), the box tuning frequency (f_B), and the box volume (V_B). Output is the frequency and relative output at that frequency.

L-Pad Program by Glenn Phillips: Appeared in *Speaker Builder* (2/83, pp. 20-22). It is useful for padding down a tweeter or midrange while still retaining the same load as the driver itself.

Vent Computation by Glenn Phillips: Calculates the needed vent length for 1, 2 or 4 ports of the same diameter. Input box volume in cubic feet and required tuning frequency (f_B), output is vent length and vent area for each case.

Medium: 5¼" SS/DD Disk. Price, **$25**.

Specify:

BOXRESPONSE

Apple BOX-1A5
Commodore 64 (Disk) BOX-1C5
Commodore 64 (Cassette) BOX-3CC
IBM BOX-1B5

IBM Plus Graphics
BOXRESPONSE BOX-2B5G $50

PASSIVE CROSSOVER

This disk is a result of Mr. Bullock's extensive research concerning first-, second-, third-, and fourth-order passive crossovers in *Speaker Builder* 1, 2 & 3/85; **$25**

Specify:

PASSIVE CROSSOVER CAD

Apple PAS-1A5
Commodore 64 (Disk) PAS-1C5
IBM PAS-1B5

IBM Plus Graphics
CROSSOVER CAD PAS-2B5G $50

CALSOD

Computer-Aided Loudspeaker System Optimization and Design

by Witold Waldman

CALSOD is a new entry into the field of crossover network optimizing software available for the IBM PC desktop computer. It combines the transfer function of an LC network with the acoustic transfer function of the loudspeaker, by using some form of iterative analysis. CALSOD creates, through the process of trial-and-error curve fitting, a suitable transfer function model which it can then optimize. The program is the subject of CALSOD author Witold Waldman's research paper "Simulation and Optimization of Multiway Loudspeaker Systems Using a Personal Computer" which appeared in the Audio Engineering Society *Journal* for September 1988, pp. 651-663. CALSOD differs considerably from other software since it models the entire loudspeaker output of a multiway system, including the low-end response, and the summed responses of each system driver.

The program performs a lot of tricks. One of the more spectacular of these allows the designer to specify the location of the driver acoustic centers using an XYZ coordinate system. Thus, if the designer expects to mount a driver combination on a flat baffle, the summed response can be optimized to compensate for rearward displacement of a woofer's acoustic center with respect to a tweeter. CALSOD can model up to seven drivers at a time in a four-way system giving the summed response and acoustic phase response of the entire system.

The CALSOD program comes on a single 360K floppy, and requires one directory and two subdirectories in installation, plus access to the DOS GRAFTABL file, which it uses for a couple of special symbols. The 133-page User Manual, provided on a second disk, is well written, adequately describes the various program functions, and contains an excellent tutorial example, which demonstrates the use of the program.

Specify:

IBM 5¼" 360K DEMO .... CAL-2B6D $ 5.00*
IBM 2 x 5¼" 360K DS/DD .. CAL-2B6 $65.00
IBM 3½" 720K DS/DD CAL-2B4 $67.50

*** Usable as credit toward later purchase of full CALSOD.**

NOTE: OLD COLONY SOFTWARE SHOWN ON THIS PAGE IS NOT RETURNABLE FOR REFUND OR CREDIT. ALSO, BECAUSE OF UNREASONABLE TAXATION REQUIREMENTS BY THEIR STATE, WE REGRETFULLY ARE UNABLE TO SELL SOFTWARE TO RESIDENTS OF CONNECTICUT.

POSTAGE & HANDLING: In US, please add $2.00 for first disk and 75¢ each additional disk. In Canada, please add $4.50. Overseas, please include 20% of total order for shipping.

DEALER INQUIRIES INVITED

CALL OR WRITE FOR YOUR FREE CATALOG!

OLD COLONY SOUND LAB

PO Box 243, Dept. B91, Peterborough, New Hampshire 03458 USA
(603) 924-6371 / FAX: (603) 924-9467
Answering machine for credit card orders only: (603) 924-6371 before 9:00 a.m., after 4:00 p.m. and weekends.
Have information plus MC/VISA available.

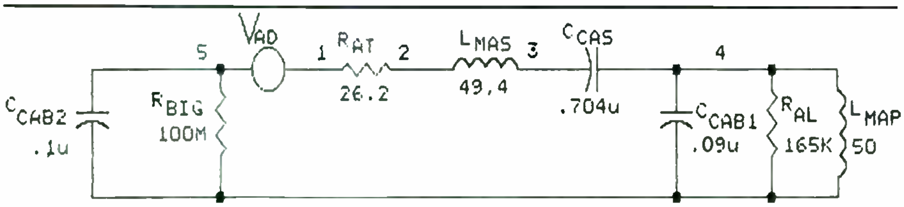


FIGURE 12: Bandpass 2 design data.

Continued from page 25

identical speakers connected in parallel. The small volume connecting the two speakers is labeled C_{CAT} (T for tunnel). The analysis of the speaker system reveals two interesting facts: (1) that the system's response is not independent of the rearmost enclosure's size (the exterior speaker is not truly in an Isobarik environment) and (2) that the small volume connecting the two speakers can cause minor frequency-response anomalies, especially as the rear enclosure becomes smaller.

Bandpass (Type 1). In this type of bandpass-speaker system, *Fig. 11* shows that both surfaces of an internal speaker are connected to a ported enclosure (like the B--e Acoustimass system). If you experiment, you will discover that for any given speaker, an infinite number of alignments are possible that differ in their passband frequency and efficiency.

Bandpass (Type 2). *Figure 12* shows a bandpass speaker system with one surface of an internal driver connected to a simple enclosure, and the other to a ported enclosure. Jean Margerand wrote articles on this type of speaker system in the 6/88 and 1/89 issues of *SB*.

CONCLUSION. PSpice offers a quick way to plot the response of virtually any kind of speaker system. Because results are available so quickly, you do not have to stick to known alignments, but can easily experiment with all manner of crazy ideas. It is so easy to play with cabinet, port, and loss parameters, you can design and simulate *many* speaker systems in an hour or two. The general methodology is quite simple; follow the step-by-step procedure below.

1. Choose a driver and calculate its equivalent circuit values by using *Table 1.*
2. Choose a basic cabinet arrangement (bandpass, bass reflex, and so on) and draw its equivalent circuit.
3. Using dimensions retrieved from an alignment table, or from educated guesses, use *Table 1* to calculate the circuit values of the cabinet components.
4. Make a circuit file.
5. Run PSpice.
6. Most importantly, play with some of the circuit values and see what kind of unusual designs you might be able to build.

TABLE 11

BPASS2.CIR

```
BANDPASS2 DESIGN EXAMPLE
VAD 5 1 AC 310 PULSE(-25V 25V .1SEC
  10U 10U .5SEC 1SEC)
RAT 1 2 26.2K
LLMAS 2 3 49.4
CCAS 3 4 .704U
CCAB1 4 0 .09U
RAL 4 0 165K
LMAP 4 0 50
CCAB2 5 0 .1U
RBIG 5 0 100MEG
.AC DEC 20 1 1K
.TRAN .05 .2 UIC
.IC V(1)=0
.PROBE
.END
```

Frequency response:
$I(L_{MAP})*FREQUENCY$
Phase response:
$IP(L_{MAP})+90$
Transient response:
$D(I(L_{MAP}))$
Cone excursion:
$I(R_{AT})/FREQUENCY$

REFERENCES

1. Old Colony has available a how-to book on PSpice, titled *SPICE: A Guide to Circuit Simulation and Analysis Using PSpice* ($19.95, BKPH2), and a student version of the software. You can also purchase the book packaged with the software ($27.95, BKPH2/S, specify software). Software: PC version ($12.95, SOF-SPC1B5GD), Mac II ($12.95, SOF-SPC1M3GD), PS/2 ($12.95, SOF-SPC2B5GD).
2. Beranek, L.L., *Acoustics*, McGraw-Hill, 1954.
3. Small, R., "Direct-Radiator Loudspeaker System Analysis," *JAES*, Vol. 20 #5, p. 383 (June 1972).
4. Small, R., "Closed-Box Loudspeaker Systems, Part 1: Analysis," *JAES*, Vol. 20 #10, p. 798 (December 1972).
5. Bradbury, L.J.S., "The Use of Fibrous Materials in Loudspeaker Enclosures," *JAES*, Vol. 24 #3, p. 162 (April 1976).
6. Thiele, A.N., "Loudspeakers in Vented Boxes: Part 1," *JAES*, Vol. 19 #5, p. 382 (May 1971).
7. Small, R., "Vented-Box Loudspeaker Systems, Part 1: Analysis," *JAES*, Vol. 21 #5, p. 363 (July 1973).

PIPE AND RIBBON ODYSSEY

BY SCOTT WOLF

PHOTO 1: A 164 mounted in a coupler. I used Mortite to form an unbroken seal between the MDF ring and the 164. It was more effective and much easier to work with than silicon rubber, and also many times less expensive.

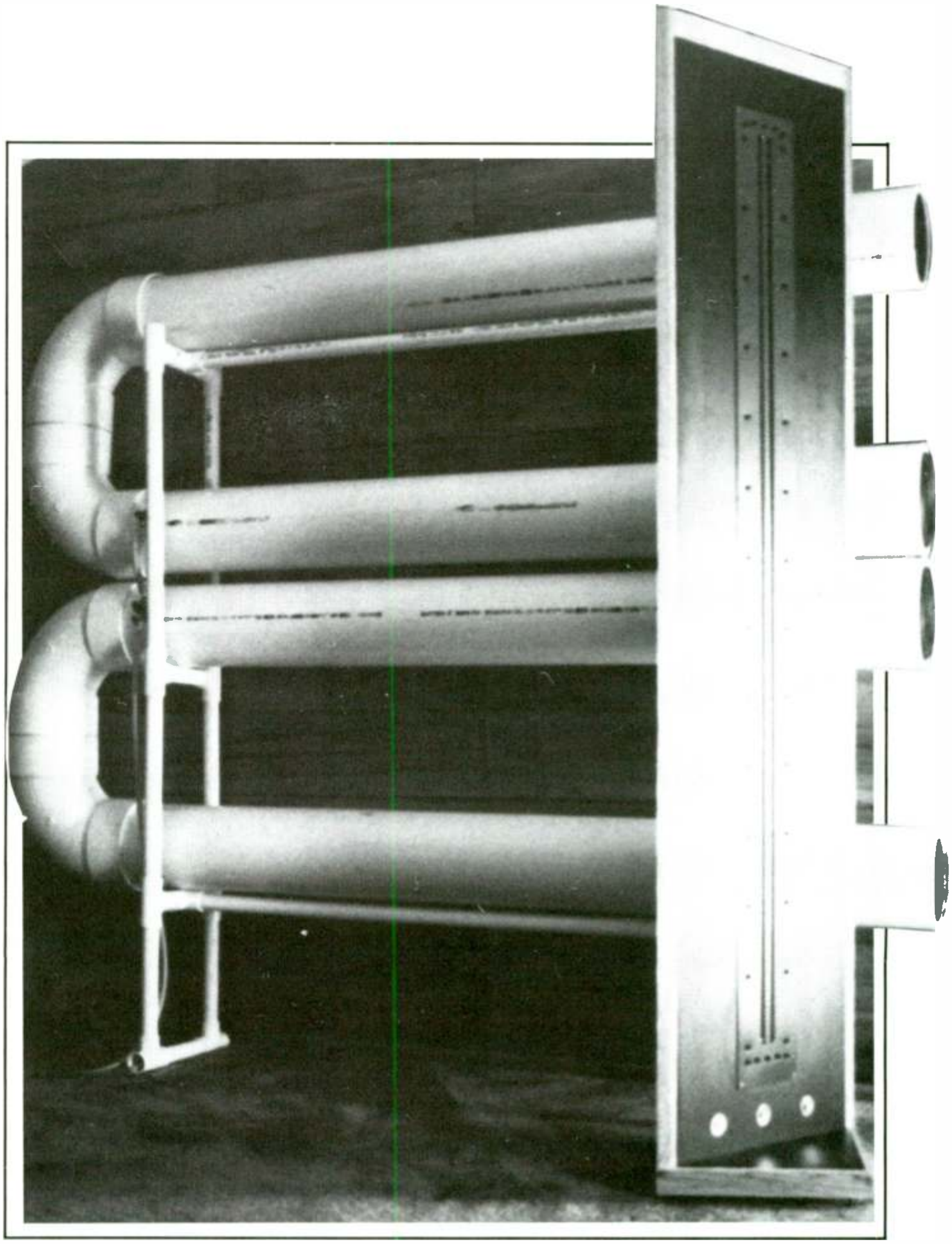


PHOTO 2: While the PVC lines are 72″ deep by 62″ tall, they take up little floor space when against side walls. The ribbon panels are 68″ by 17″ by 0.85″. The base is 12″ by 12″.

In January of 1987, I attended the Consumer Electronics Show (CES) in Las Vegas, always a marvelous learning experience. Before that time, I had believed the Dynaudio transducers were the most cost-effective method of reproducing a musical event.

I found several enjoyable loudspeakers at the Riviera. The speaker with perhaps the most realistic sensation of a live musical event was the Duntech Crown Prince PCL-1000 driven by Rowland gear. I listened to several planar-ribbon types also. The Apogee Caliper powered by Lazarus was a close second, yet the same speakers driven by Carver equipment in a stereo store sounded terrible—a very amplifier-dependent loudspeaker.

THE GOLD RIBBON. I had seen the Gold Ribbon in several issues of *SB* and was very curious about it. I believe it was operated in a sealed enclosure with two 7″ Focal bass drivers in transmission lines. This setup was crossed over to a single Velodyne subwoofer. The deep bass was non-musical. I've never enjoyed most vented alignments with the inherent ringing imposed on the driver from such an enclosure. The midbass wasn't bad, but you had to sit close to on-axis or the high-frequency performance suffered.

Yet, something about the Ribbon mesmerized me; it had a sense of immediacy. The company claimed the Ribbon could

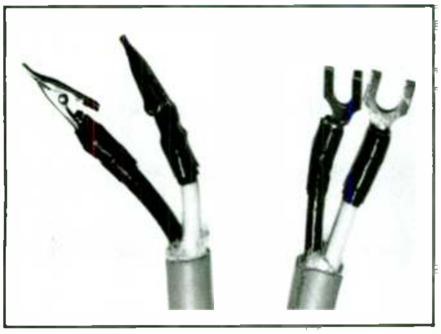


PHOTO 3: Monster Cable power line 2 is used for internal and external wiring for the woofers. The solid copper alligator clips allow the driver-coupler to be removed with a twist-and-pull motion. The Old Colony spade lugs terminate to Old Colony five-way binding posts on the PVC pipe.

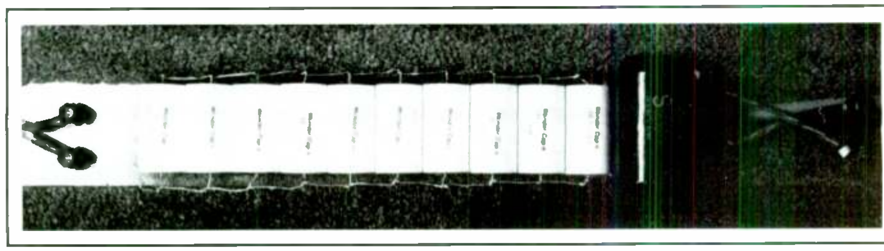


PHOTO 5: Capacitors mounted on ¼″ plexiglass using Old Colony binding posts, 12-gauge Monster wire, 10 IAR $10\mu F$ WonderCaps, and a $100\mu F$ Chateauroux, purchased from Audio Concepts.

resolve differences as small as $3\mu S$. I asked a salesman what that meant. He said it was far too complicated to explain. I lucked out, though, and talked to Allen Hulsebus, a company engineer. He drew a μ-shaped curve in the air and said the time between the peaks was $3\mu S$. This driver could start and stop very quickly. I listened to the speaker for 30 minutes. For the $1,600 asking price and the performance limitations, however, I couldn't justify grabbing a pair.

ALTERNATE SOLUTIONS. When I read "A Low-Cost Two-Way Ribbon System" by Richard Painter (*SB* 4/88, p. 38), I became intrigued. For $57, I had 4″ wide, ¼″ thick steel cut to a length of 72″, and had ½″ holes drilled with centers 1″ apart. The pole pieces are 2″ wide by ⅛″ thick. A friend who does vapor deposition for optical instrumentation told me about a company that would do vapor deposition onto Kapton. The problem: a minimum order of $500, but I still planned to go ahead. When Edmund Scientific ran out of 4 lb./ft. Alnico V magnets, I checked out Permag's price. Each magnet cost $4, quite a bit more than Edmund's price. Project terminated.

I first saw the RD-57s in *SB* 3/89 when I flew to the west coast to audition them. They were driven with solid-state Harman-Kardon amps, with a single subwoofer up against the rear wall. The ribbons were operated as dipoles and were placed in the middle of a fairly large room. The driver had the sense of speed that the Gold's had, yet could resolve an acceptable stereo image anywhere in the room.

I listened to the Duntech Sovereigns at Christopher Hansen, Ltd. Those 2001s were the earlier version using the D-76 midrange unit. The amplification was in four large boxes per channel, manufactured by Cello. The RD-57 yielded a more musical experience than Duntech's use of the Dynaudio line, with far less costly amplification. Also the RD-57 is made in America.

I accumulated information from several sources to guide me. I've been reading Richard Heyser's articles in *JAES* Loudspeakers Anthology, Vol. I since 1987. I don't claim to understand most of the math. It's been eight years since I had a class in Laplace transforms and the Convolution Integral. If I had to solve a problem now, I'd probably burn out a frontal lobe. Heyser's articles, however, convinced me of the need to adhere to the idea of a loudspeaker as a minimum phase network, where the phase response is a Hilbert Transform of the amplitude response.

THE SPEAKER BASICS. My one requirement in building these speakers was for them to be powered by a single stereo amp. To accomplish this and approximate minimum phase behavior, I needed to use a first-order 6dB/octave passive filter. Ralph Gonzalez's article (*SB* 1/87, p. 20, *Fig. 4*) illustrated the minimum phase nature of the first-order filter.

My crossover is very simple. I used Bob Bullock's program to generate the values on an Apple IIe. It spat out $198.9\mu F$ and 3.18mH for a crossover point at 200Hz. I used ten $10\mu F$ IAR caps and one $100\mu F$ Chateauroux in parallel and a 3.18mH Perfect Lay inductor from Solen.

The ribbon or electromagnetic transducer was invented by E. Gerlach and W. Schottky about 60 years ago. It is an elegant use of the Lorentz force, $F = qv \times B$. Problems of rise time and perhaps more importantly time to cease motion associated with cone and dome designs are greatly reduced.

The RD-57 uses a four-turn motor structure with three rows of ceramic V magnets in front of and behind the voice-coil diaphragm for a push/pull setup. The aluminum voice coil, polyamid (Kapton) diaphragm is convoluted. If viewed edge on, it would resemble a square wave.

I mounted the ribbons in panels of ¾″ MDF type 44 and laminated black formica on front and back. The panels are bolted to bases with three ¼″ bolts each 2″ long. The bases have a groove routed out about ¼″ deep into which the panels fit. I used red oak as a trim material and ran the oak strips on the panels through a table saw with the blade set at 23°. This allowed me to bevel both sides of the strips to approximate a rectangular truncated pyramid and parallel-piped (*SB* 1/80, p. 29, *Fig. 5*), figuring in dipole operation. To obtain the correct width of the panels, I used B. Berg's technique (*SB* 2/84, p. 30) to avoid cancellation above 200Hz.

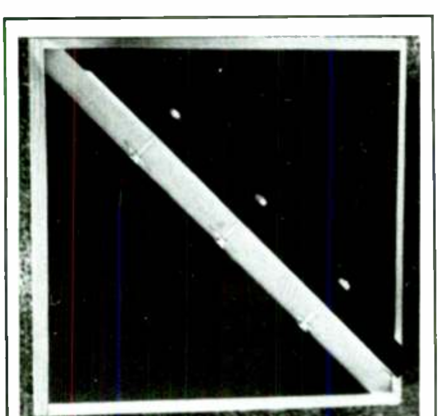


PHOTO 4: Top view of the ribbon's base.

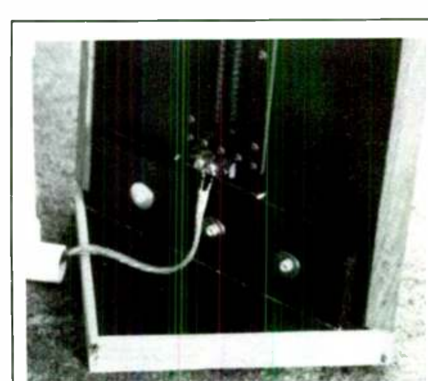


PHOTO 6: The wiring for the ribbons is 12-gauge Monster cable, run through the 1.1″ ID PVC pipe frame. The ribbon base "floats" on brass insert-machine screws.

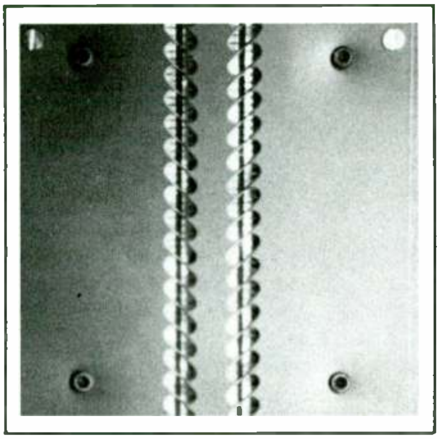


PHOTO 7: Close view of the ribbon's front.

PERFORMANCE. While the RD-57s sounded quite good with a single subwoofer, using a passive filter with such shallow slopes makes such a setup unsuitable. I've been pleased with the low-frequency performance of the Morel Integra Is in a 7.84′ PVC line (*SB* 5/89, p. 51). I had some spare pipe, approximately 7′. I removed the lower part of the u-shaped structure, about 34″ in length, and substituted the 7′ length. The bass was considerably deeper.

I bought four Morel 164s from Zalytron for $194 including shipping. These dual-magnet beauties have 75mm diameter voice coils and doped paper cones. They remind me of the plastiflex-treated paper Dynaudio 21W54s, except the 164's dust cover is paper instead of cloth, the surround is neoprene instead of foam, and the voice coil diameter is 75mm. Morel claims an f_S of 37Hz with a flat response out to 6kHz.

In 1936, Ben Olney reported that a driver with a resonance at 50Hz dropped to 40Hz when placed in a labyrinth. I've been using the 164s in 12.3′ long pipes with an allowed fundamental of about 23Hz. I placed Acousta-Stuf in the final 5′ of the lines, using only enough to eliminate any resonances stimulated by my talking into the open end. Besides extremely deep bass, the internal reflections found in so many enclosures are greatly reduced, if not eliminated. This allows the driver to cease motion sooner after an impulse has been reproduced.

Front-panel resonances are also eliminated. Examples of these problems may be seen in the third edition of *High Performance Loudspeakers* by Martin Colloms. On page 13 of this book, *Fig. 2.3* is the directivity pattern for a rigid circular piston in the end of a long tube. At 200Hz, Ka for my speakers is 0.10 and even at 5kHz, Ka = 2.6. The 164s with no crossover sound fairly three-dimensional (polar response plot, Θ in degrees, r in decibels).

Tapping the cone lightly and placing your ear at the open end results in an uncomfortable sensation due to low-frequency propagation. There is considerable output from the pipes while reproducing a 20Hz, ⅓-octave warble tone.

Because I left the PVC in the stands and lines unglued, I was able to observe the behavior of the 164s in much longer lines. I disassembled the lines in which the Integras were mounted and used the elbow joints to increase the line length to 28′.

By using a block of wood and a mallet, I pushed the MDF rings about 3″ into the couplers. This allowed me to experiment with the 28′ line in a conventional single driver and push/pull alignment. In this setup, the 164s could not handle the excursion forced on them by air movement they themselves stimulated, even at quite low volume.

To couple the bases holding the ribbon panels to the floor, I used threaded brass inserts and 2″ long brass machine-screws (*Photo 4*), not so much for vibration reduction as for compensating for old floors. By varying the length of each screw combination in each of the base's four corners, I got the panels to appear orthognal to the floor.

I've listened to many planar-ribbon and electrostatic-type loudspeakers at quite a few high-end audio stores. The performance of the RD-57 is not found wanting. Depending on the low-frequency alignment and associated woofers chosen, it is possible to construct a loudspeaker capable of holding its own with any speaker I've heard regardless of price.

I prefer transmission-line labyrinth types. The low-frequency performance is tight, clean, and 3-D. I'm using Carlos Bauza's (*SB* 4/85, p. 32) "near-field" listening position (*Photo 8*). The performance of the 164s in the 12.3′ long lines has convinced me it will be worthwhile to load my four Dynaudio 30W54s into 18′ long, 10″ internal diameter pipes. My speakers cost me $2,400, less than half the retail price of the Seas woofer/Focal tweeter combination of the Wilson Watts.

PHOTO 8: "Near-field" listening position with ears about 0.9 meters from the ribbons, while the woofers are about 1.5m from the side walls. Note the lack of oak trim. I spent about seven months listening in this position.

ZALYTRON

YOUR ONE STOP SHOPPING CENTER

IF YOU HAD THESE ZALYTRON CABINETS, YOU'D BURN YOUR OLD ONES!

For the Audiophile who wants the best!!

YOU HAVE NEVER SEEN CABINETS LIKE THESE BEFORE

These cabinets are the latest of Zalytrons efforts to bring the finest cabinets to the market. They are made with 11/8" MDF to start. then the side walls are laminated with 1/2" flake board (thus giving 1 5/8" sidewalls!!). Next the sides are rounded to give 1 1/2" radius curve. We then wrap Nevamar brand laminate around the cabinet. The result is a strong attractive cabinet with less diffraction loss. The thick walls improve the bass remarkably. They simply have to be heard and seen to be believed and appreciated.

Pictured are the Aria 5 with Accuton tweeter and the brand new Time-aligned models of Focal 033 and 133 systems.

ZALYTRON INDUSTRIES CORP.
469 JERICHO TURNPIKE, MINEOLA, N.Y. 11501
TEL. (516) 747-3515 FAX (516) 294-1943

Our warehouse is open for pick-up 10AM to 6 PM daily, Monday thru Saturday
UPS orders shipped same day • Minimum order $50.00

FOR ALL YOUR SPEAKER NEEDS OR INQUIRIES - WRITE OR CALL ASK FOR ELLIOT

Fast Reply #GF1131

THE DELAC S10

BY RALPH GONZALEZ

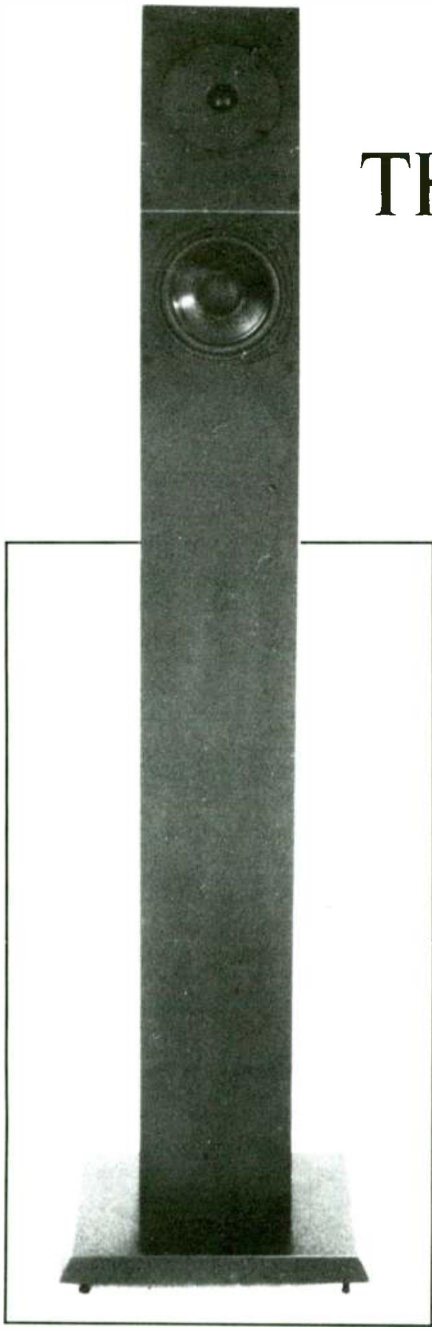


PHOTO 1: The DELAC S10.

The S10 loudspeaker was my attempt at designing a system that did not compromise important performance characteristics (soundstaging, timbral balance, musicality, aesthetic acceptability) at the expense of those I could do without (high power handling, bass response below 50Hz).

The result, based on my own and other owners' observations, is an elegant, unusual, and compact (5″ by 3.5″ by 40″) loudspeaker that produces a 3-D yet focused soundstage, a tight and surprisingly deep bass, an extended yet mellow treble, and a realistic vocal and string tone. Due to power limitations, it is best suited to small and medium-size rooms, where the dynamics can be very lifelike. As an unexpected benefit of the design, the sound quality remains musical even when you are positioned well off-axis or in another room.

DELAC (Delaware Acoustics) was an ill-fated attempt to bring the S10 loudspeaker to a wider audience (see the sidebar "What Went Wrong?"). While I would not have attempted to introduce another "box" loudspeaker to the already saturated market, I believed the unusual aesthetics of the S10 would help differentiate it.

This article presents construction details on the DELAC S10, as well as several design features that can be incorporated into your own designs.

REVERBERANT RESPONSE. At low frequencies, dynamic loudspeakers radiate sound in all directions. At a given frequency, typically 300–800Hz, the baffle begins to become an obstacle to the sound waves and limits the radiation of higher frequencies to the forward hemisphere. If you make no attempt to correct this, the speaker will exhibit a gentle 6dB rise in its frequency response—termed the "response step"[1] or (viewed as a 6dB falloff at low frequencies) "diffraction loss." This phenomenon is often visible in the manufacturer's frequency response graphs for woofer or midrange drivers, when the measurement conditions involve a small baffle. It is also responsible for the "forward" sound of poorly designed small speakers.

Most good loudspeakers eliminate the response step by means of the crossover design to achieve a flat anechoic response. Thus, the forward radiation is made flat, while the rearward radiation is limited to the bass and lower midrange. As a result, the overall energy the loudspeaker radiates is unbalanced. In my opinion, this discrepancy causes loss of a convincing soundstage, particularly in large speakers.

PHOTO 2: SEAS drivers.

I designed the S10 to be a loudspeaker whose reverberant response (the soundfield which reaches the ears only after reflecting from one or more room surfaces) was as well-balanced as its on-axis response. The approach involved placing a second bass/midrange driver on the rear of the enclosure so the forward and rearward responses are the same up to the tweeter crossover frequency (around 3–4kHz). The intent was NOT to increase

TABLE 1

SPECIFICATIONS

Power	Recommended for amplifiers from 15–100W/channel
Nominal power rating	80W
Sensitivity	85dB/W/m (2.83V), though the perceived sensitivity is somewhat higher due to the enhanced reverberant response.
Nominal impedance	4Ω, with a minimum of 3.5Ω

the overall ratio of reflected/direct sound energy (as in some B_ _e loudspeakers), but to improve the frequency *balance* of the reflected sound.

ENCLOSURE CONSTRAINTS. This seemingly simple solution introduced constraints that determined several other areas of the design.

The output from the rear-mounted driver is omnidirectional at low frequencies and is limited to the rear hemisphere at higher frequencies. This helps "fill in" the bass and lower midrange response on the *forward* axis to compensate the aforementioned response step. However, computer simulations indicated that the rear-mounted driver's time delay caused its output to interfere with the forward-facing driver's on-axis response at certain frequencies.

To minimize this partial cancellation, I made the time delay as small as possible. This necessitated a shallow enclosure, made unusually narrow to raise the frequency of the effect to the 600–800Hz region. The crossover helped compensate the remaining effect. To maintain internal volume, then, required a long vertical dimension, leading to the columnar floor-standing design of the S10.

Rather than using a second, rear-mounted tweeter to maintain the balance of the reverberant response above the crossover frequency, I mounted the tweeter angled partly away from the listener. In this way, extra high-frequency energy could be fired into the room without producing a bright on-axis response. As a side effect, high-frequency beaming is reduced. (This technique is also seen in some Spica designs.)

DRIVER CONSTRAINTS. The narrow enclosure, in turn, suggested a small bass/midrange driver. Such drivers have the advantage of having better upper-midrange performance than larger drivers. This allows a highish crossover frequency to a small tweeter whose high-frequency response is especially smooth and well-dispersed.

I chose the 4.5" SEAS 11FG-X bass/midrange driver (*Photo 2*) for its exceptional bandwidth (50Hz to 6kHz) and quality construction, including a rigid cast magnesium frame and well-damped diaphragm. Its magnet is so large it determines the size of the driver's mounting hole. This driver has found its way into commercial designs by ProAc and Thiele, for its absence of coloration, wide bandwidth, and long-throw design. The 0.75" SEAS H225 dome tweeter is a ferrofluid-cooled design with an amplifier-smooth response not found in larger tweeters.

PHOTO 3: The crossover/backplate.

What attracted me to the Scandinavian SEAS line was the completeness of the driver specification sheets. These included graphs of each of the following: on- and off-axis response, impedance, and second- and third-order distortion products. Most importantly, they provided details on the measurement conditions.

MINIMUM-PHASE CROSSOVER. A rarely attempted design goal is a multi-driver system with a flat frequency response *and* a phase response near zero throughout the crossover region. Unlike conventional speakers, such systems pre-

PHOTO 4: The passive bass equalizer.

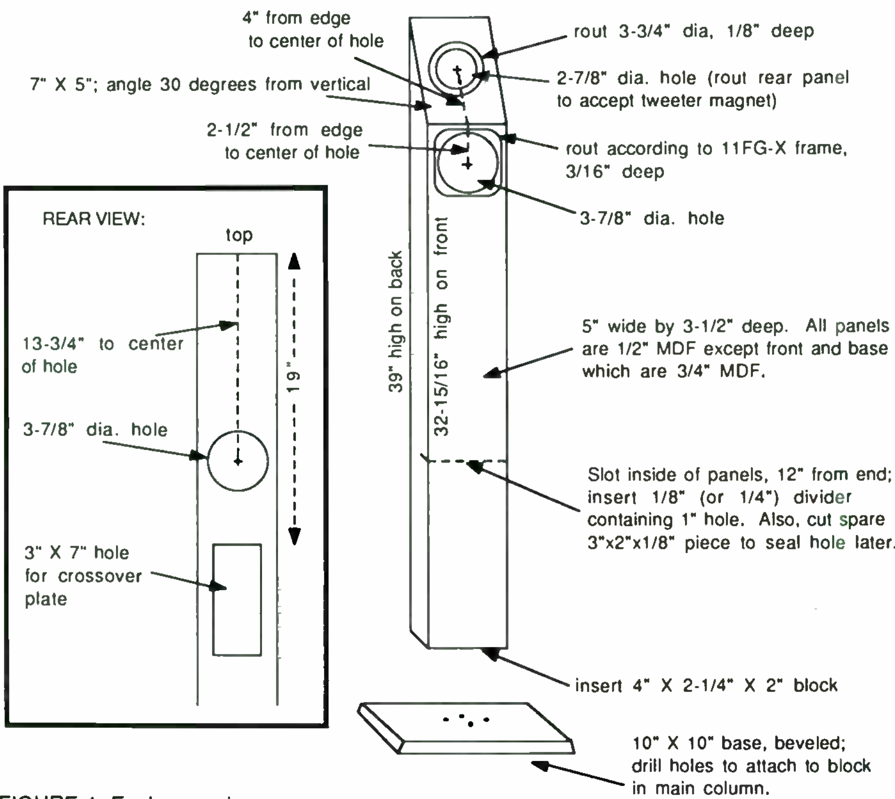


FIGURE 1: Enclosure plans.

serve the *shape* of the musical waveform in the same way as does the rest of the reproduction chain (although it has not been conclusively demonstrated that this has an audible effect on typical music signals). In theory, a first-order crossover produces this effect; however, due to the additional phase shift real-world drivers produce, most commercial systems using first-order crossovers are anything but minimum phase.

One technique (pioneered by Spica) for approaching the minimum phase ideal with real-world drivers requires that the tweeter be delayed slightly with respect to the woofer.[2] The S10 uses this technique, along with a computer-optimized crossover. If you listen *above* the axis, however, you will experience a response dip near the crossover frequency.

Apart from its minimum phase aspirations, I designed the S10 crossover to compensate for the falling response usually produced by the angled mounting of the tweeter.

BASS EQUALIZATION. Unfortunately, the SEAS 4.5″ woofer is so well-damped by its oversized magnet that its response is −6dB at 100Hz. Even with a vented loading, this will produce an overly "lightweight" sound (as noticed in the ProAc Tablette minispeaker).

Instead of using a vent, the S10 tackles this problem with a simple line-level equalizer inserted between preamp and power amp or in a tape loop. Since the driver's natural rolloff is rather gentle, a simple first-order equalization serves to extend the −6dB point to 50Hz. The combined response remains well-damped and integrates well with typical listening rooms. The response continues to fall naturally below 50Hz, unlike the sudden cutoff of vented designs.

The circuit to produce this equalization was designed with no active components (transistors or tubes), which might otherwise influence sound quality. The price paid for this passive circuit is a 10dB loss of preamp gain, compensated for by higher volume control settings. Adding a single op amp will restore this gain, if necessary.

Unfortunately, bass equalization reduces the low-frequency drivers' power handling. The S10's dual 4.5″ units have a combined surface area equivalent to a single 6.5″ woofer. This is sufficient for satisfying levels in average-size rooms. However, the speakers cannot produce live sound levels in large rooms. Fortunately, the SEAS 11FG-X drivers tend

TABLE 2

DELAC S10 PARTS LIST

Crossover Parts (each channel)
(5) 0.47μF film capacitors (or one each: 1μF, 1.5μF)
(1) 15μF film capacitor
(1) 0.4mH air-core inductor, DCR > 0.4Ω
(1) 10Ω, 10W resistor
(1) 5Ω, 10W resistor
(2) binding posts (four for optional bi-wire connection)

Equalizer Parts (each channel)
(1) 7.5kΩ, ¼W resistor
(1) 5.1kΩ, ¼W resistor
(1) 1.6kΩ, ¼W resistor
(1) 0.33μF film capacitor
(1) 0.068μF film capacitor
(2) phono jacks
(1) DPDT switch to "bypass" both channels

Overall Parts, less enclosures*
(4) SEAS 11FG-X bass/midrange drivers (8Ω)
(2) SEAS H225 tweeters
(2) crossover boards plus crossover parts
(1) bass equalizer box plus parts for two channels
(2) speaker grilles, if desired
(8) carpet spikes and inserts (T-nut or threaded)
weatherstrip tape, polyester fiberfill, sand, drywall screws, and so on

Parts Available from the Author**
SEAS 11FG-X 4.5″ bass/midrange driver ($21)
SEAS H225 0.75″ ferrofluid dome tweeter ($8.25)
Fully assembled bass equalizer, low-impedance version only ($58)
Punched and silk-screened metal equalizer enclosure ($13)
Etched and silk-screened backplate/crossover board ($12)
5″ by 12″ by ⅜″ acoustically transparent foam grille ($2.75)
Set of four multi-way binding posts ($2)
Four-way patch cord for equalizer in/out ($6)
1½″ by ¼″ threaded carpet spike (40¢)
0.4mH Sidewinder inductor: DCR = 0.2Ω ($2.68)
15μF Mylar capacitor ($1.48)
0.47μF polypropylene capacitor (56¢)
5 or 10Ω, 10W wirewound resistor (25¢)
DELAC S10 owner's manual, including bi-amping instructions ($2)

* If you buy from a speaker component distributor, the total cost not including enclosures is about $200–$250.

** To save money or if you need parts for another project, I have these parts available in limited quantities which I'll sell at my cost. All are new. Due to quantity purchases, these prices are considerably lower than retail. Please send orders or questions to Ralph Gonzalez, PO Box 54, Newark, DE 19711. Add 7% for shipping.

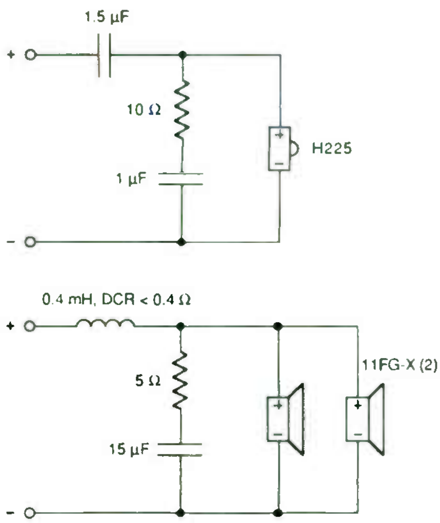


FIGURE 2: Crossover diagram.

to compress the sound when overdriven, rather than failing catastrophically.

BASS CORRECTION. I've also experimented with a small subwoofer, to *replace* the line-level equalizer. A simple but effective design consists of a Madisound 81524DVC 8.5" dual voice-coil woofer, mounted in a 28-liter sealed box. I added weights to the cone to reduce its sensitivity by 3 or 4dB (try three or four size 5 fishing weights) and to lower the resonant frequency and raise the Q. The first-order crossover consists of a 6mH iron-core inductor (DCR > 0.5Ω) in series with each voice-coil connection. (That is, you need two such inductors.) By using a down-firing arrangement, the effective crossover slope becomes more like a second-order one.

I did not use a high-pass crossover with the S10 (a series capacitor would be ineffective due to the bass-midrange driver's impedance peak), since the S10's natural response is equivalent to a second-order, high-pass filter at 100Hz. Its power handling is increased significantly by the removal of the bass equalization circuitry. The overall response with this subwoofer is extended to 35Hz (−6dB), and the system may be used with amplifiers up to 150W/channel.

MORE BENEFITS. Some features of the S10 design served several purposes. For example, in addition to minimizing the rear driver's time delay, the narrow enclosure has another beneficial effect: using ½" and ¾" medium density fiberboard eliminates the need for additional wall bracing. Rapping on the enclosure produces the same sound as rapping on a solid block of wood.

Internal damping consists only of polyester fiberfill, since the frequencies at which standing waves develop are high enough to be readily absorbed. I partially attribute the speaker's soundstaging qualities to the absence of enclosure colorations, which would otherwise inform the ear that the sound is indeed coming from the speaker. The tall, narrow enclosure also eliminates the need for costly speaker stands.

An important side-effect of the rear-mounted driver is to cancel residual fore/aft enclosure motion produced by the front-facing driver, which might otherwise produce a form of intermodulation distortion. The angled top surface provides the desired off-axis tweeter listening angle and the time delay required for the near minimum phase crossover. Although somewhat tricky to construct, this angle contributes to the aesthetic appeal of the slender enclosure.

FINE-TUNING. I obtained measurements with a calibrated microphone in a semi-anechoic room, which validated the computer-designed crossover. While computer simulation and measurements help in obtaining a crossover that integrates the drivers properly, they can't judge the ear's sensitivity to certain as-

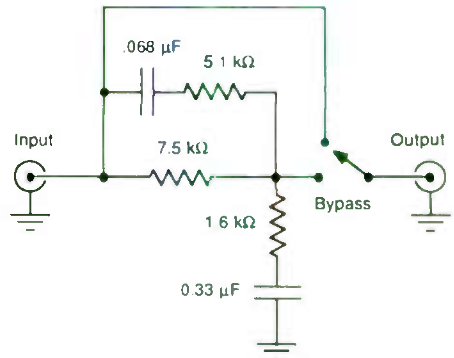


FIGURE 3: Equalizer diagram (one channel).

pects of musical sound or to the subtle effects of listening-room acoustics.

I spent many lengthy listening sessions tweaking the tweeter level by 1dB at a time to find the one that sounded neither dull nor bright in a real-world listening room. Similarly, I experimented seemingly endlessly with slight changes in the bass equalization to obtain the most musical balance. As noted below, my initial choice was probably a little lightweight in the bass, due to the absence of a true full-range system to serve as a reference.

ENCLOSURE PLANS. *Figure 1* is a plan

What Went Wrong?

Delaware Acoustics was a one-man operation aimed at selling S10s (as well as a smaller, more conventional speaker and subwoofer). All manufacturing (enclosures, boards, and equalizer) was contracted out, save for the final assembly of speakers. I created not-quite-glossy brochures and an owner's manual using desktop-publishing software. Sales were factory-direct, since retail sale would have nearly doubled the $629/pair list price.

My marketing plan for the S10 was (too) simple: get a good review in a national magazine and wait for orders. I managed to get a few press releases and finally a review in *Stereophile* (April 1989). The reviewer was "stunned" by the precision of the vocal image and thought the soundstage was "holographic," but ultimately withheld a recommendation on the grounds of a perceptible midrange coloration and lightweight bass response.

In later listening tests and comparisons, I discovered both of these effects were room- and placement-dependent. As a compromise, I moved the bass equalization frequencies slightly to "fill in" the lower midrange and to raise the bass response by 2dB in the 50–100Hz range. By this point, however, the fun had gone out of the project and (not wishing to invest in advertising) I let the business fall aside.

Only a handful of S10s were sold. In hindsight, the mail-order speaker business has limited potential since audiophiles naturally prefer to see and hear speakers before buying. DELAC was limited by my lack of marketing know-how, as well as by my failure to devote the kind of money and effort required to start a new business. On the bright side, my startup costs were low, making it a relatively inexpensive failure.

Since then I've applied a few of the design ideas to consulting projects, but haven't done anything with the S10 design, save use one for my own listening. I still think the design has commercial potential if marketed effectively, sold through retail outlets, and perhaps re-engineered for easier manufacture. A fully active version of the bass equalizer could eliminate the gain and loading requirements which may have scared off non-technical purchasers.

From this experience, I'd advise potential entrepreneurs to seek outside financial and managerial backing. No matter the product, success requires commitment and outlay as well as planning.

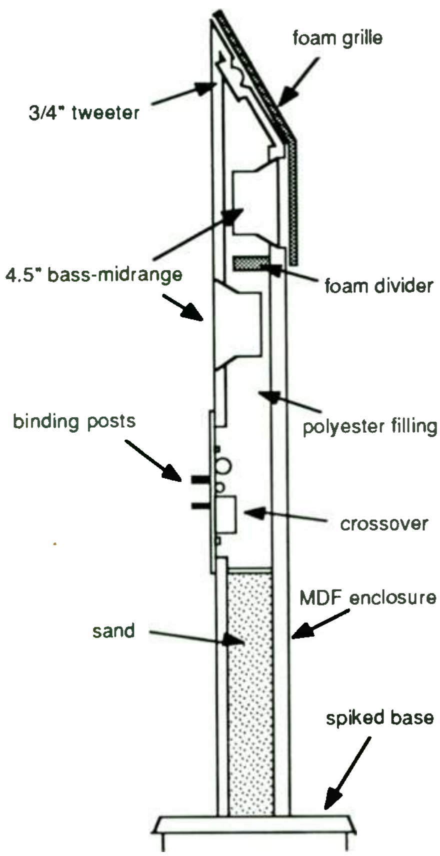


FIGURE 4: Cutaway view of the S10.

for the S10 enclosure. All pieces are ½″ medium density fiberboard (MDF), except the front and bottom panels which are ¾″. Since the dimensions are small, butt joints will suffice. A few tricky tasks are cutting the sharp top edge, routing a crescent shape inside the back panel to accommodate the tweeter magnet, and correctly sealing the subenclosure for sand filling. Don't substitute thicker MDF, since the bass/midrange magnets will not fit.

Not being a woodworker, I had a local craftsman construct the enclosures, at $200-$250/pair including finishing. For finishes, I tried Braewood[3] laminates, which consist of stained wood (the Rosewood is especially attractive) on an easily applied backing. Dull or glossy formica was also attractive, durable, and economical. You may wish to paint the enclosures, though this will require careful sanding and several coats due to the sharp edges.

CROSSOVER PLANS. *Figure 2* is the circuit diagram for the S10 crossover. In appearance, this is a simple first-order design using impedance compensation, with different corner frequencies for the high-pass and low-pass sections. However, by combining the natural rolloffs of

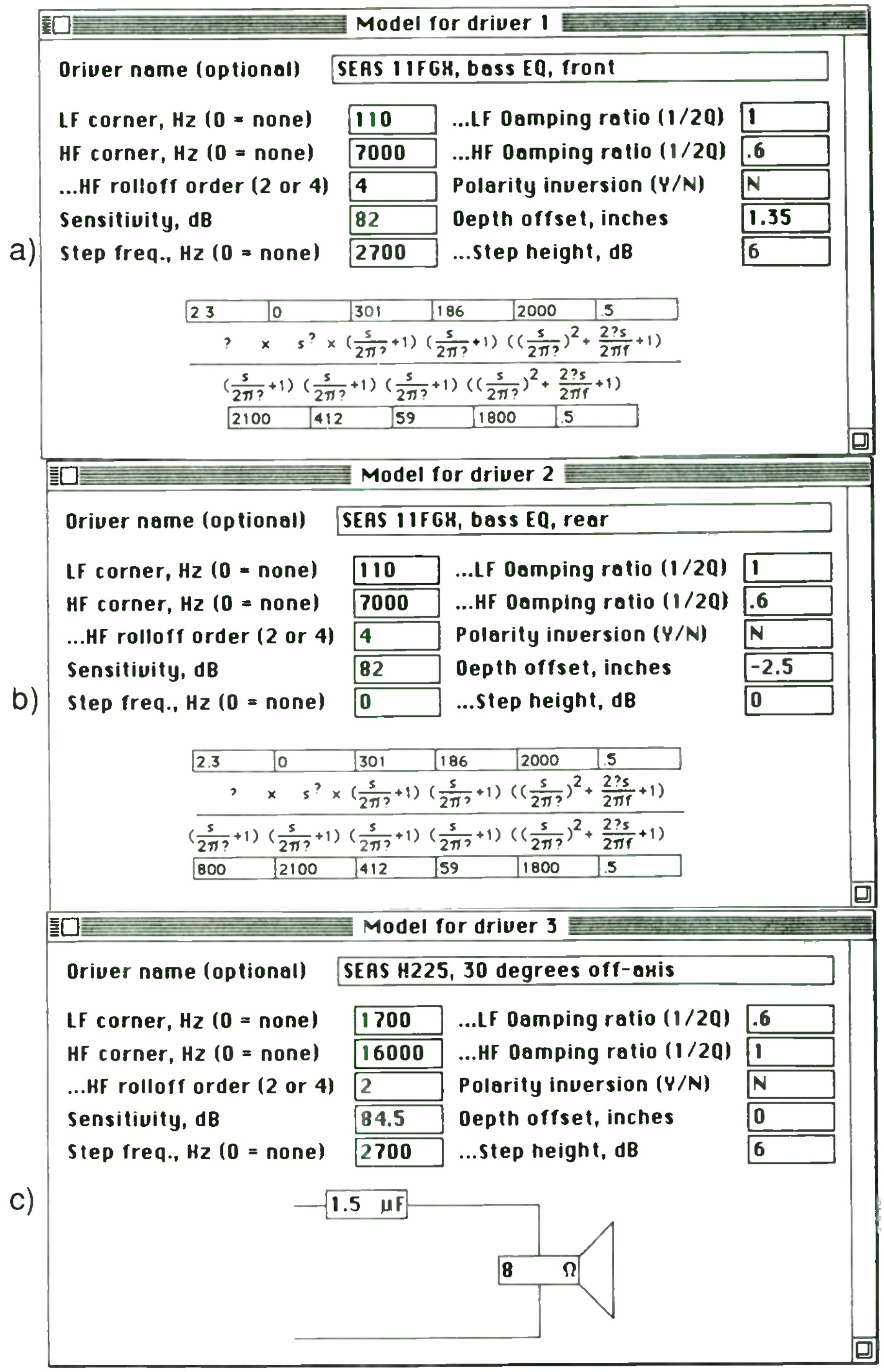


FIGURE 5: LMP Professional model (a) for the front woofer, (b) for the rear woofer, and (c) for the tweeter.

the selected drivers and the enclosure geometry, you get a near minimum phase design with acoustical slopes varying from 6-18dB/octave (see "LMP Modeling" below).

For easy wiring, I used heavy-duty printed circuit boards, though you may prefer to glue the components to a piece of 4¼″ by 8¼″ by ⅛″ fiberboard. The binding posts are attached directly to this board, which mounts on the back of the enclosure. *Table 2* shows each speaker's parts.

BASS EQUALIZER PLANS. *Figure 3* is the bass equalizer circuit diagram. Its effect is a 6dB boost at frequencies below 100Hz, relative to higher frequencies.

Due to its passive nature, the circuit imposes requirements on the input and output impedance of its associated power amp and preamp. The power amp input impedance should be greater than 10kΩ and the preamp output impedance less than 1kΩ. This will generally be the case with transistor systems.

However, you can multiply all the resistors by 10 and divide all capacitors by 10 to use the equalizer with power amps whose input impedance is greater than 100kΩ and preamps whose output impedance is less than 10kΩ, appropriate for most tube systems.

As I mentioned earlier, the equalizer causes about 10dB of insertion loss. If you have insufficient gain to tolerate this, add a single op amp to recover this gain. This has the secondary benefit of allowing you to eliminate some loading requirements.

ASSEMBLY. Before attaching the base, mount T-nuts in the corners of the underside to accept carpet-piercing spikes (or carriage bolts for leveling on hardwood floors). I recommend the PinPoints spikes available from AudioPrism,[4] since they are sharp enough to pierce carpets. AudioPrism also sells threaded inserts that are more reliable than T-nuts.

As shown in the cutaway in *Fig. 4*, you should glue a piece of absorptive foam (using silicone sealer) inside the enclosure to partially separate the two bass/midrange drivers. Fill the lower compartment with sand to provide mass and stability and then seal it to prevent sand from reaching the drivers. The remainder of the enclosure is stuffed with polyester fiberfill to control standing waves. [*Be sure to thoroughly dry the sand before filling. Bake it in shallow pans in an oven, if necessary.—Ed.*]

An easy, effective means of sealing the drivers and crossover plate to the enclosure was by using ¼″ thick, self-stick, open-cell urethane weatherstrip tape. Simply apply the adhesive side around the edge of the hole and press the driver over it. This avoids the usual hassle when trying to remove or replace drivers sealed in place using silicone sealer.

Once you have mounted the drivers and crossover (I used ½″ and 1″ drywall screws), you can attach the grille. A simple, frameless piece of ⅜″ reticulated foam works well. Since the dimensions are small, the foam holds its shape nicely and has no effect on the sound. I attached it using several small pieces of Velcro for easy removal. (Radio Shack and many hardware stores sell Velcro with a sticky backing.) The grille was the one feature my wife criticized.

LMP MODELING. I used my Macintosh-based LMP Professional software (sold as "Souped-Up LMP" by Old Colony) to help develop the crossover for the DELAC S10. LMP Professional offers the choice of specifying the crossover model conventionally via circuit component values or directly specifying the crossover

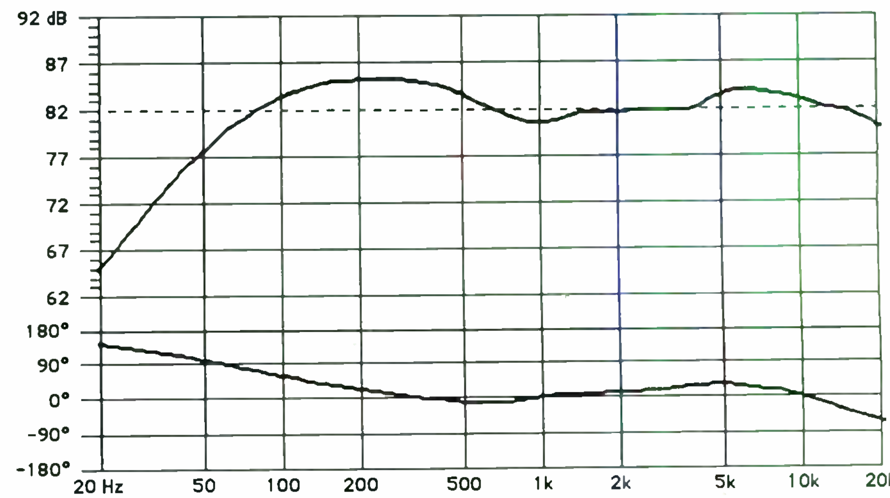


FIGURE 6: Predicted frequency response.

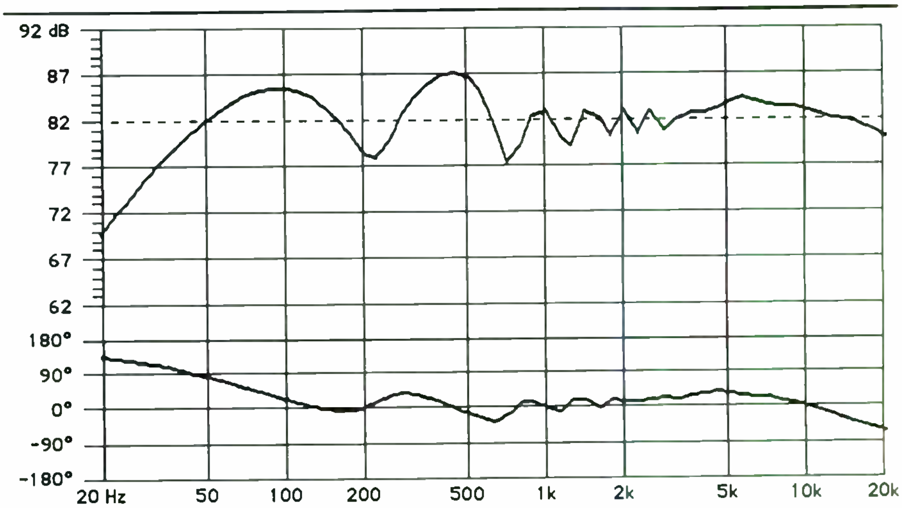


FIGURE 7: Predicted response including floor reflection.

transfer function. I used the latter approach to incorporate the bass equalization and minor driver aberrations into the woofer transfer function (*Fig. 5*).

The predicted system response (*Fig. 6*) suffers from partial cancellation at 1kHz due to the delayed rear driver output. I lessened the relative magnitude of this cancellation with the crossover by reducing the overall response in the region above 1kHz. (Subjectively, the additional midrange reverberant energy produced by the rear driver helps "fill in" this cancellation further.) What remains is a broad 3dB "hump" centered at 250Hz.

This is ameliorated if the floor reflection is included in the model, as shown in *Fig. 7*. If not for the rear-mounted driver, the jaggedness caused by the floor reflection would be worse. Also notice on this graph that the system's −6dB point relative to the midrange sensitivity is around 30Hz. (You can model the floor reflection using LMP by adding an imaginary driver with the same characteristics as your woofer, but attenuated and

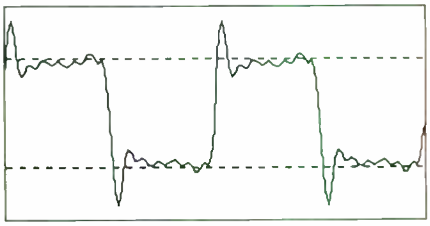


FIGURE 8: Predicted response to 1kHz square wave.

delayed according to the extra path length of the floor reflection. The carpet's effect may be modeled by adding a first-order, low-pass filter to the imaginary driver, at about 300Hz.)

LMP Professional's predicted square wave response is shown in *Fig. 8*. This unusually good result is a tribute to the S10's minimum phase crossover. A related result is the relative absence of phase shift from 200Hz to 10kHz shown in *Fig. 6*.

I apologize for not having up-to-date empirical measurements available. However, I have had good correlation be-

Continued on page 39

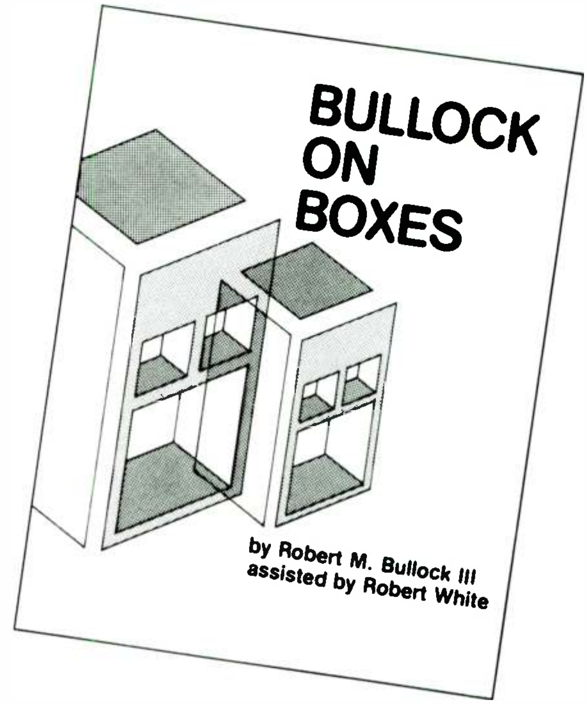
Now Available by Popular Demand!

DEALER INQUIRIES INVITED

BULLOCK ON BOXES

BY ROBERT M. BULLOCK III

with Robert White



Since 1980, readers of *Speaker Builder* magazine have delighted in the scope and clarity of Bob Bullock's explanations of vented-box loudspeaker systems. Now, for the first time, these articles have been collected into a single invaluable volume: *BULLOCK ON BOXES.*

This book is an easy-to-read guide to designing and building vented-box systems based on Thiele/Small models. And in addition to the background theory and descriptions of the models, *BULLOCK ON BOXES* carefully covers a host of related information drawn from numerous resources. Computer modeling, as demonstrated by the authors' BOXRESPONSE and BOXMODEL programs, is also explained in clear detail.

A professor of applied mathematics at Miami University (Ohio) since 1966 and long-time hands-on speaker buff, Dr. Bullock brings an unparalleled range of experience to the pages of *BULLOCK ON BOXES.* Fully indexed and referenced, with glossary, bibliography, and more than 100 illustrations, *BULLOCK ON BOXES* is a practical handbook and lasting reference indispensable to every serious speaker builder. Order yours today!

BKAA8 1991, 8½″ × 11″ **$10.95**

SPECIAL OFFER: For a limited time only, purchase *BULLOCK ON BOXES* and receive your choice of software package below at $5 off the regular retail price. Order **BKAA8/S.***

*Because of taxation requirements, software not available to residents of CT

CHAPTERS:
- Thiele, Small, and Vented Loudspeaker Design
- Determining Design Parameters for Your Loudspeaker
- Fine Points of Vented Speaker Design
- Alternative Alignments (incl. Fourth Order)
- T/S Calculator Programs (TI-58/59, HP-67/97)
- BOXRESPONSE: An Apple Program for the Thiele/Small Models
- Realizing BOXRESPONSE's Potential
- BOXMODEL: An Aid to Woofer System Design
- Correspondence & Updates

SOFTWARE PACKAGES AVAILABLE (Sale Prices Shown):

BOXRESPONSE $20
Apple (not Mac) #BOX1A5 Commodore #BOX1C5
IBM #BOX1B5 Comm. Cassette #BOX3CC

BOXRESPONSE WITH GRAPHICS $45
IBM only #BOX2B5G

BOXMODEL WITH GRAPHICS $45
IBM only #MOD3B5G

(NOTE: In general, BOXRESPONSE automatically generates alignments for a given driver. BOXMODEL does not do this, but instead compares up to four given alignments simultaneously. Both programs display various performance aspects of passive-radiator or vented-box systems.)

☐ CK ☐ MO ☐ MC/VISA

NAME ______________________

STREET & NO. ______________________

CITY ______________ STATE ______ ZIP ______

MC OR VISA NUMBER ______________ EXPIRE ______

☐ **YES!** Please send me ____ *BULLOCK ON BOXES* at $10.95 each! **BKAA8 $______**

☐ **SOFTWARE, TOO!** Please send me one *BULLOCK ON BOXES* at $10.95 **BKAA8/S ______**
PLUS my choice of software package #__________ at only $__________! **BKAA8/S ______**

POSTAGE AND HANDLING (see below) ______

TOTAL ORDER $______

OLD COLONY SOUND LAB
PO Box 243, Dept. B91
Peterborough, NH 03458-0243
(603) 924-6371 / 924-6526
FAX: (603) 924-9467
Answering machine for credit card orders only: before 9:00 a.m., after 4:00 p.m. and weekends. Have all information plus MC/VISA available.

POSTAGE AND HANDLING:
U.S.—Book only: $2.00 first book, 75¢ each additional.
Book plus software: $2.75.
Canada—Book only: $4.50 first book, 75¢ each additional.
Book plus software: $5.25.
Other—Add 20% to order.

CALL OR WRITE FOR YOUR FREE CATALOG!

Continued from page 37
tween LMP predictions and such measurement with earlier incarnations of the speaker. One purpose of simulation software like LMP is to reduce the need for repeated empirical measurements, which would otherwise mandate access to an anechoic chamber and/or computer data acquisition hardware.

USE. Connect the bass equalizer in a tape loop (make sure your Tape Mon button is pressed) or between the preamp and power amp.

Due to the 4Ω nominal impedance, be sure to use 16-gauge or heavier wire for the speaker connection and (for tube amps) the 4Ω speaker tap.

Position the speakers 12–18″ (or more) from the wall behind them, and slightly closer together than the distance from your listening position. For best results, place the speakers along the long wall of the room. Angle the speakers inward slightly so they are aiming at a point behind you rather than directly forward or directly at you. Your seated ear level should be at or below the tweeter level.

Now, fire them up!

REFERENCES

1. Gonzalez, Ralph, "An Introduction to Frequency Response and LMP," *SB* 1/87, p. 18; 2/87, p. 42; 3/87, p. 38.
2. Gonzalez, Ralph, "Minimum Phase Crossovers," *SB* 3/88, p. 34.
3. Braewood is a product of Brookside Veneers, Ltd., 215 Forest St., PO Box 4348, Metuchen, NJ 08840-0990, (201) 494-3730.
4. AudioPrism, PO Box 1124, Issaquah, WA 98027, (206) 392-0399.

If you should have a *TECHNICAL QUERY...*

about an article appearing in this magazine, write it clearly, leaving space for a reply and referencing the magazine, the article and the page about which you are inquiring. Enclose a self-addressed stamped envelope and send these to *Speaker Builder*, PO Box 494, Peterborough, NH 03458.

If it's warranted, we will forward your query to the author or a Contributing Editor for a prompt reply.

AN INFRARED REMOTE VOLUME CONTROL

BY JOSEPH O'CONNELL

This project is an inexpensive gadget that allows you to control your stereo system's volume and other functions from another room. Also, you can add a set of extension speakers without degrading the main system's sound. Over the four wires connecting the remote unit to the base unit, this project will vary the volume over a 113dB range, switch on a separate power amplifier for the extension speakers, and relay the commands from a hand-held infrared remote control to your main stereo.

The block diagram in *Fig. 1* shows how you can use this project and an extra power amplifier to add extension speakers to a main stereo system. Extension speakers are often connected to a main stereo system by switching them in parallel with the main amplifier's existing speaker outputs. Connecting two sets of speakers to one amplifier, however, has a number of disadvantages. The amplifier may have trouble driving the lower impedance presented by the two speakers.

Also, since the two sets of speakers will probably differ in sensitivity, impedance, and power handling, their volume levels will be difficult to match. Without an L-pad, the extension speakers' volume depends on the same control the main speakers use. With an L-pad, however, the usable power is limited and the amplifier still sees a lowered impedance. Either alternative results in a disagreeable compromise.

Driving the extra speakers with a separate amplifier helps you avoid these problems. In addition to the extra amplifier, a remote volume control is necessary; it controls the volume of the signal sent to a power amplifier, but it does so without sending the audio signal to the remote location and back.

The stereo signal is taken from the master system through its tape output jacks.

ABOUT THE AUTHOR

Joseph O'Connell received his B.A. in history and philosophy of science, graduating from the University of Chicago with highest honors, and plans to work as a professional historian or inventor. He has electronic experience as a hobbyist and is the author of *20 Innovative Electronic Projects for Your Home* (TAB Books).

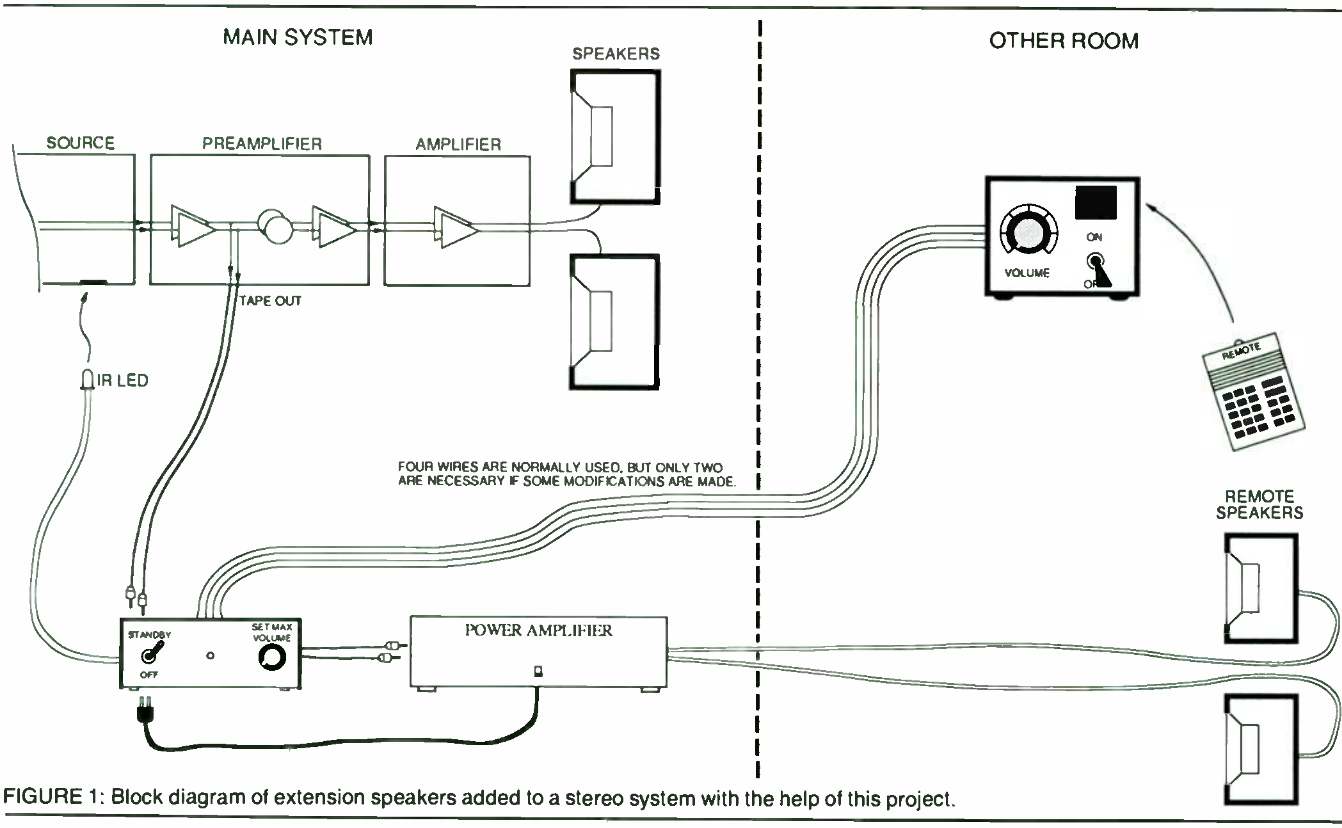


FIGURE 1: Block diagram of extension speakers added to a stereo system with the help of this project.

PHOTO 1: The base and remote units.

PHOTO 2: The back of the base unit has connections for the remote unit, the IR-emitting LED, and the power amplifier.

At this point, the signal level is independent of the main speakers' volume setting. The base unit takes that signal, electronically attenuates it in response to a remote potentiometer's setting, and sends it to the amplifier that drives the extension speakers. The amplifier is plugged into an outlet in the base unit where it receives power when the remote power switch (S2) is turned on. That way, it is used only when necessary. The remote and base units are shown in *Photo 1*.

THE BASE UNIT CIRCUIT. The circuit for the base unit is shown in *Fig. 2*. The Motorola MC3340P is an inexpensive IC designed for this type of application. It has a control range of −90 to +13dB and contributes a maximum total harmonic distortion of 0.6% or less. This is unacceptable in the main system, but is satisfactory for casual listening through extension speakers. The volume at the MC3340P's output (pin 7) depends on a variable resistor placed between its control (pin 2) and ground. A separate IC is used for each channel, but a single potentiometer controls both simultaneously.

POT3 is the remote volume control. It determines the ICs' output volume. Back at the base unit, POT1 is connected in series with POT3 to limit the maximum volume level allowed. This feature is optional but might be desirable if the extension speakers are particularly sensitive or if you wish to keep people from raising the volume above a certain level from the remote location.

The attenuation of the MC3340P ICs is not linear in response to changes in control resistance. Nor does it have a logarithmic response to compensate for the way the ear perceives sound levels. Rather, it is somewhere in between, making a compromise inevitable. *Figure 3* plots the measured attenuation (in decibels) of either channel of the circuit versus the control resistance placed across

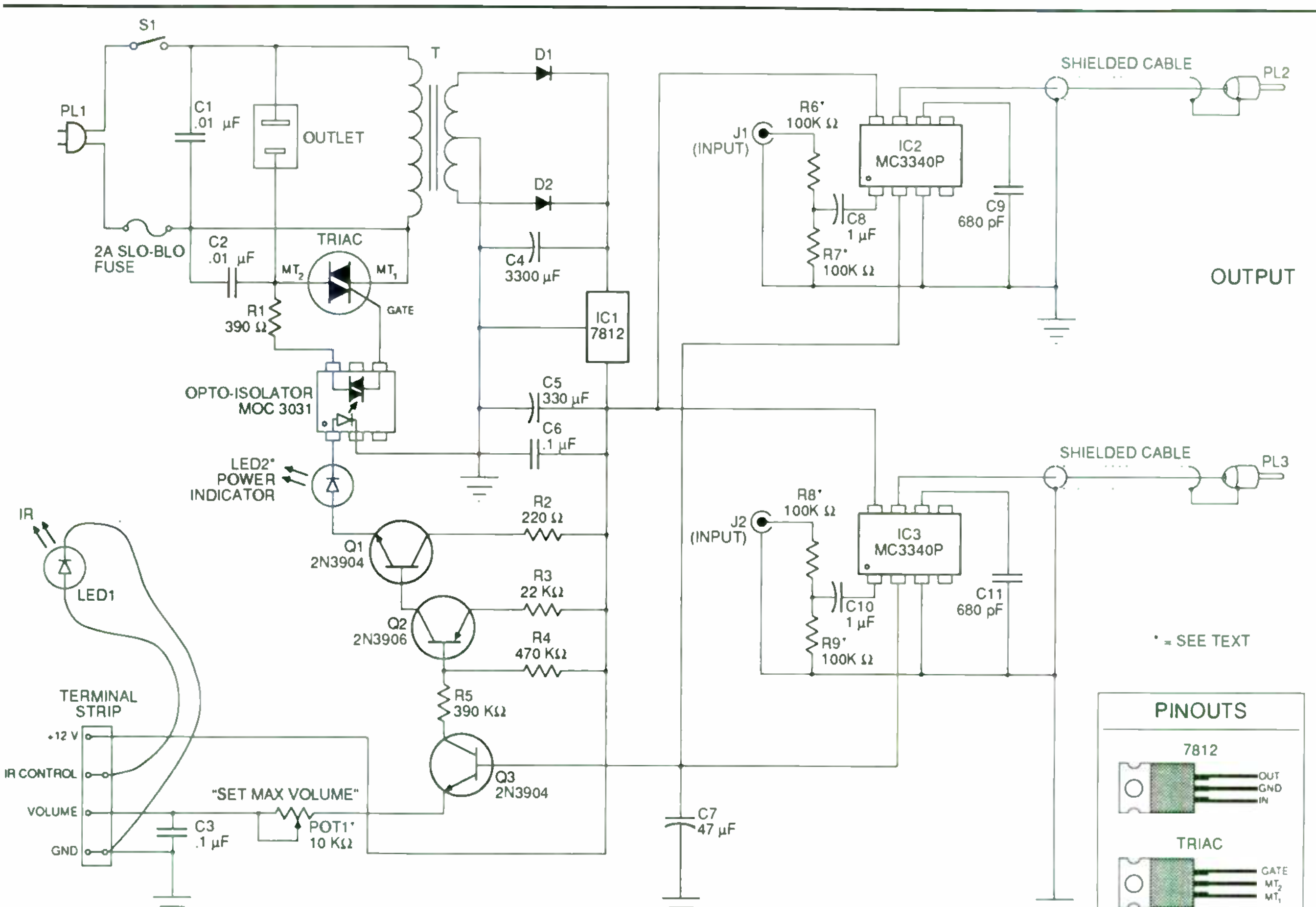


FIGURE 2: The base unit.

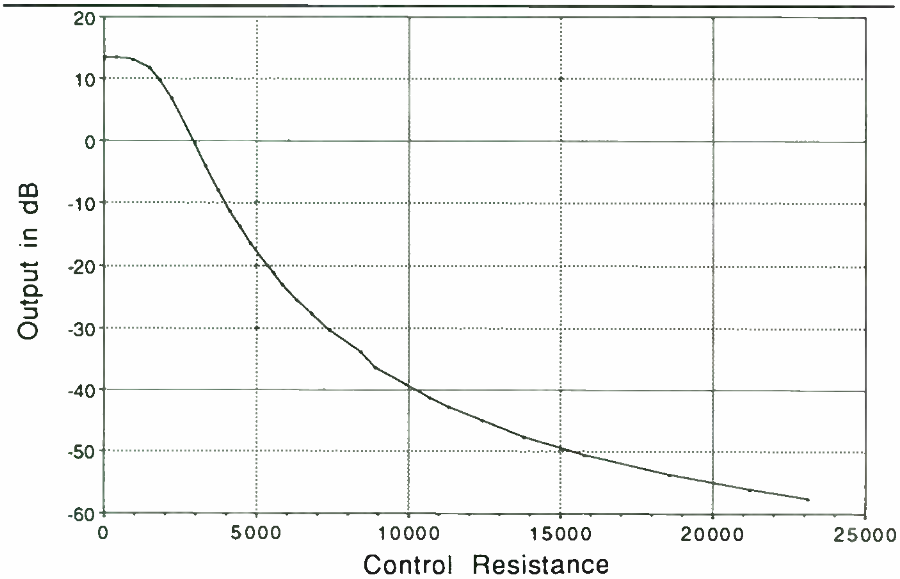


FIGURE 3: Plot showing the measured volume of the project as a function of control resistance. Ideally, it would be a straight diagonal line. The response of both channels was nearly identical, so only one is shown. Power supply voltage was 9V.

the two paralleled ICs. From this plot, I decided that a 7.5kΩ linear-taper potentiometer would give the right amount of range. Since that value was not readily available, I settled on a 10kΩ pot.

When S2 is on, current flows from pin 2 of each volume control IC through the remote circuitry and through Q3 to ground. The current through Q3 causes Q1 and Q2 to conduct, powering the opto-isolator, which causes the TRIAC to fire. The TRIAC allows power to flow to the extension amplifier, which is plugged into the outlet.

You can gang S2 with POT3 to control power and volume with a single knob. The schematic also shows an optional muting switch (S3). When activated, it switches from the control potentiometer to a fixed resistor, preselected to give a low volume level. If you wish to include a mute switch on your project, select the muting resistor (R17) according to your preferred sound level. Adding a switch or relay at the base station is a similar way you could activate muting. You can connect the relay to mute the volume level automatically in response to a phone call or ringing doorbell or when a tuner is scanning the dial.

THE REMOTE CIRCUIT. As *Fig. 4* indicates, a small piece of infrared filter plastic shields the photodiode (D3) in the remote unit from ambient light. This material looks dark red and is nearly opaque to visible light. You can obtain it at low cost from some electronics surplus dealers. If you can't find the correct filter material, use a thin piece of translucent red plastic instead.

The photodiode is biased by POT2, which must be tweaked to get the maximum range from the circuit. Three of the inverters in IC1 form an amplifier, while the remaining three are connected in parallel to drive the IR-emitting LED. The output from IC1 is sent back to the base unit over a wire where it drives LED1, carefully placed to shine on the IR windows of any equipment in the main system that you can control remotely.

CONSTRUCTION. You should build this project in a metal box to shield the audio circuit from hum pickup. As with

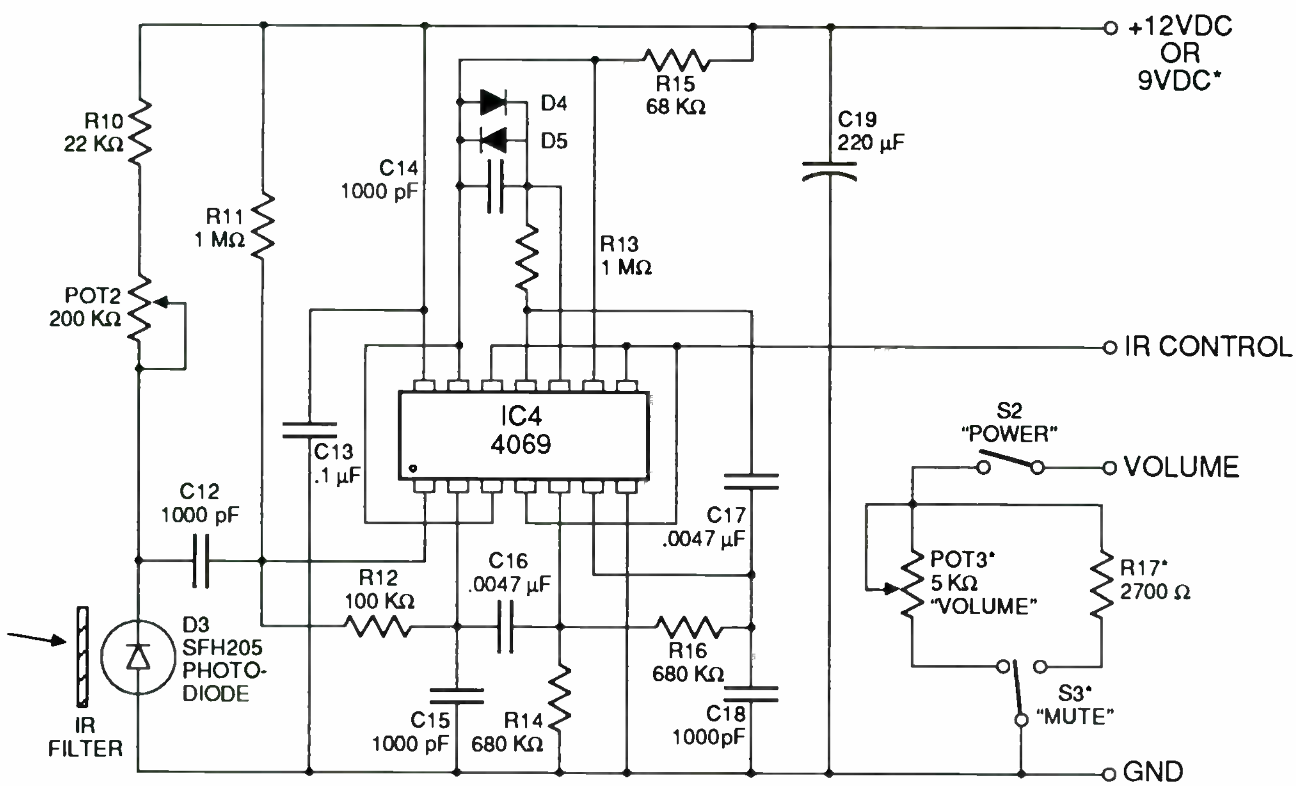


FIGURE 4: The remote unit.

TABLE 1

REMOTE VOLUME CONTROL AND INFRARED RELAYER PARTS LIST

Resistors					
R1	390Ω	R1	390Ω	R1	390Ω
R2	220Ω	C9, 11	680pF	J1, 2	panel-mount RCA jack
R3, 10	22kΩ	C12, 14, 15, 18	1,000pF	outlet	panel-mount power outlet
R4	470kΩ	C16, 17	0.0047μF	PL1	line cord and wall plug
R5	390kΩ	C19	220μF, 16V or better electrolytic filter	PL2, 3	RCA plug
R6–9	100kΩ, metal film. These form a pair of optional input attenuators if your stereo system's output would otherwise overload the 0.5V RMS maximum input voltage of the MC3340P.			POT1	10kΩ, panel mount pot (optional)
		Semiconductors		POT2	200kΩ, trimmer pot (to adjust photo-diode bias)
		D1, 2	50V (1N4001), 1A or better silicon rectifier diode		
				POT3	5kΩ or slightly higher (7.5kΩ) panel-mount pot (main volume control)
		D3	SFH205 photodiode (from Active Electronics: 800-343-0874)		
R11, 13	1MΩ			S1, 2	panel-mount power switch
R12	100kΩ	D4, 5	1N914	S3	panel-mount DPST toggle switch (necessary only for muting)
R14, 16	680kΩ	IC1	7812 or equivalent 12V three-terminal regulator		
R15	68kΩ				
R17	approximately 2.7kΩ. Find this muting resistor's value experimentally.	IC2, 3	MC3340P electronic attenuator (Motorola)	shielded cable	shielded cable, or cut an interconnect cable in half and use cable in lieu of this and PL1 and PL2
		IC4	4069 hex inverter		
		LED1	IR-emitting LED (LD271, from Active Electronics)		
Capacitors				T	power-supply transformer with 24V CT secondary
C1, 2	0.01μF, 200V or better bypass, value not critical	LED2	red LED		
		opto-isolator	MOC3031 or equivalent TRIAC opto-isolator	terminal strip	four-position, panel-mount terminal strip or other form of two-strip to four-terminal feed through
C3, 6, 13	0.1μF, 50V or better bypass, value not critical				
		Q1, 3	2N3904 or other general-purpose NPN transistor		
C4	3,300μF, 16V or better electrolytic filter				
		Q2	2N3906 or other general-purpose PNP transistor		
C5	330μF, 16V or better electrolytic filter				
		TRIAC	200V, 2A or better		
C7	47μF, 16V electrolytic			suitable enclosure, circuit boards, fuse holder, mounting for LED1, panel-mount holder for LED2, IR filter plastic, two to four conductor cable to connect remote to base, knobs, hookup wire, rub on letters, solder	
C8, 10	1μF, 16V Nonpolarized electrolytic is OK, but polystyrene is better.	**Miscellaneous**			
		fuse	2A slo-blow fuse, unless your power amplifier draws more current		
R1	390Ω				

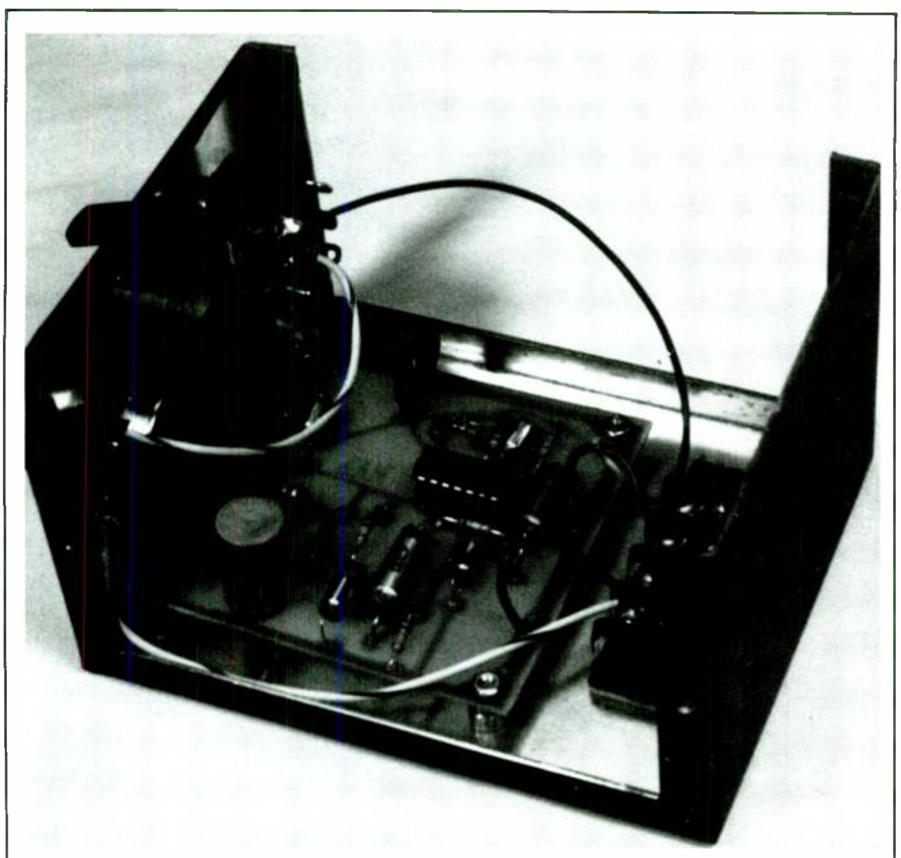


PHOTO 3: The remote circuit board mounts inside the remote unit with much room to spare.

all audio circuits, try to separate the audio portion of the controller physically from the AC wiring and power supply components. Other than these considerations, construction should present no special difficulties and depends mostly on your enclosure's dimensions.

The board etching and parts-placement diagrams in *Fig. 5* should simplify construction for those who etch their own boards. If you don't have access to PC etching equipment, the circuit is simple enough to build onto perfboard or onto a universal PC board like the Radio Shack #276-168.

The circuit's power supply is not very critical since the ICs can operate over a supply range of 9–18V, so you can use whatever power supply parts are available. For the prototype, I mounted the power supply and TRIAC components on a piece of PC board, cutting the foil into conductive islands with a Dremel Moto-Tool. Perfboard and hard-wiring will work just as well. Connections to the remote are made through a four-position terminal strip as shown in *Photo 2*.

You can tuck the base unit away somewhere and forget it, but it perhaps should remain visible so you can see LED2, which indicates that the remote system

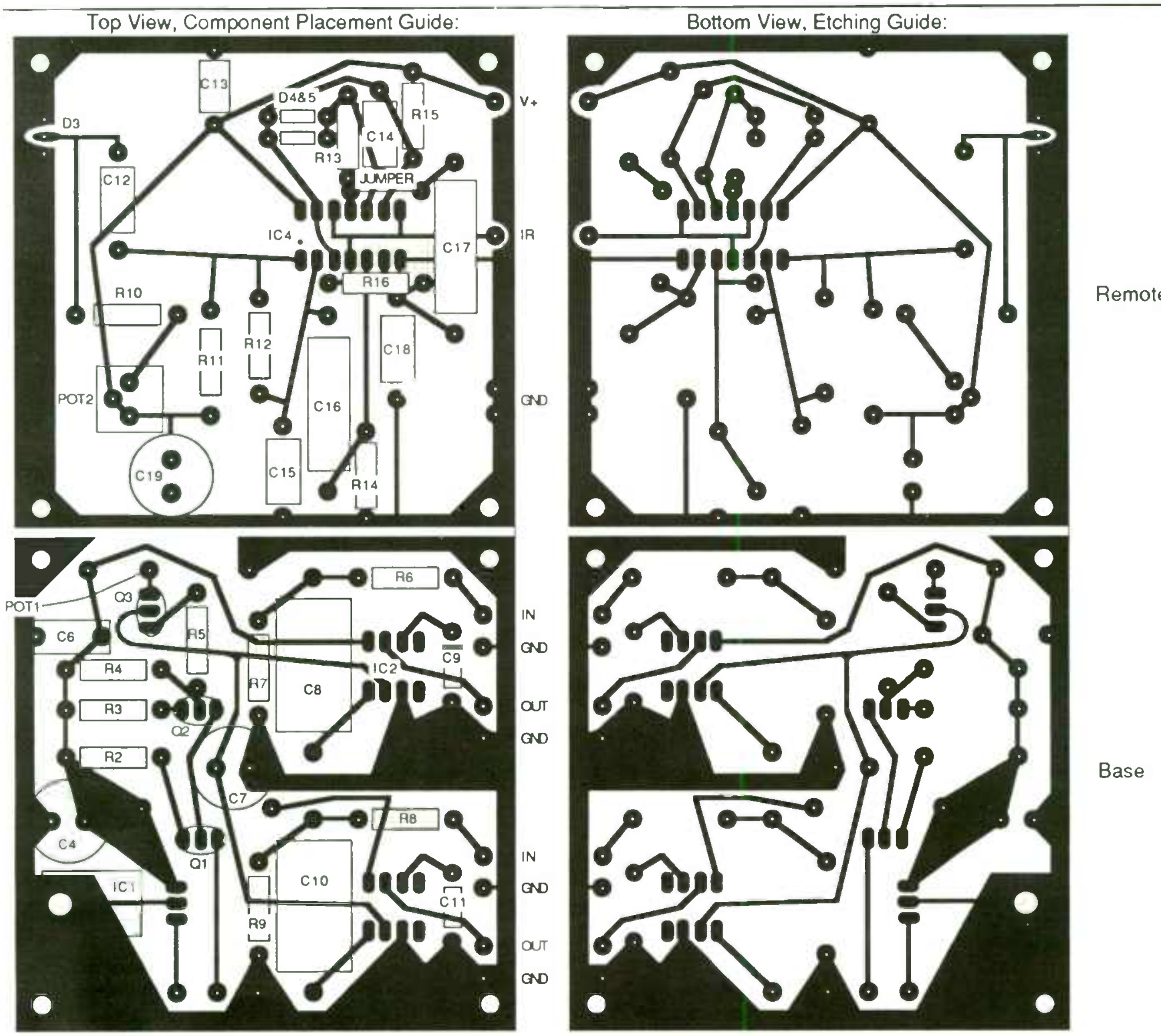


FIGURE 5: Parts-placement and etching guides for the base and remote units.

is in use. If the power amplifier for the extension speakers has its own pilot light, this can serve the same function as LED2.

How you build the remote station depends on where you will use it. For installation in a wall, you can mount POT3 and S2 in a faceplate since the station will not need a complete enclosure. Or you could use a small minibox if it will stand alone, as in *Photos 1* and *3*. Notice that the parts-placement guide indicates three holes are provided for D3, which actually has only two leads. This is so its orientation can be reversed to allow its sensing window to face the right direction.

As drawn in the schematics, this project requires four wires between the remote and base units. But if a common ground is available (as it should be in a house), you can reduce the wire count to three. To further reduce this to two wires, provide the remote unit with its own 9–18V power supply and eliminate the supply wire coming from the base unit.

You should mount the IR-emitting LED a short distance from the sensing windows on your main system, but keep it as unobtrusive as possible. It should not block the window completely since you will probably wish to use the remote control directly. If your system is inside a cabinet, the sides of the cabinet make a good mounting surface for the LED.

OPERATION. The remote volume control should be left switched on all the time and connected to your stereo as shown in *Fig 1*. Until you turn on the remote switch, it remains in a standby condition. Only when switch S2 is on is LED2 lit and the extension amp activated. The power the remote volume control draws during standby is insignificant and the part of the circuit connected to the stereo is isolated from the 120V AC power line for safety. If you do not wish to leave it on all the time, plug the remote volume controller into one of the switched convenience outlets your main system's components provides.

MORE CIRCUIT USES. With its wide control range, the MC3340P IC can serve as an inexpensive preamplifier for line-level signals. With it connected between a tuner or tapedeck and a power amp, it becomes easy to set up a simple stereo system without a preamplifier. Since the circuit is so compact and its power supply requirements are easily met, you could probably build it inside a tuner, tape deck, or amplifier chassis without much trouble. It could then obtain the power supply it requires from the equipment within which it is operating.

OLD COLONY'S NEW REFERENCE TOOLS!

COMPACT DISC PLAYER MAINTENANCE AND REPAIR — **BKT16** — **$14.95**
Gordon McComb, John Cook

Complete with a preventive maintenance schedule you can follow, this well-illustrated guide walks the reader through the steps involved in taking apart, cleaning, and lubricating a CD player. And if problems do occur, the troubleshooting techniques in this book provide easy relief. The book also includes background information on the theory and operation of CD players, what criteria you should use when shopping for one, a list of major manufacturers, a "similar model" cross reference table for troubleshooting, and a handy glossary. 1987, 245pp., 7 ½ × 9 ½, softbound.

DESIGN AND BUILD AUDIO AMPLIFIERS —INCLUDING DIGITAL CIRCUITS — **BKT17** — **$19.95**
Mannie Horowitz

This second edition is a complete course in designing and building audio circuits for all electronics applications, covering JFETs, equivalent noise generators, and much more. Whether you need a preamp, amp, power amp, mixer, tone modification circuit, power supply, or special accessory, here is all the data you need to create completely up-to-date circuits. 1980, 350pp., 5 × 8, softbound.

HOW TO MAKE PRINTED CIRCUIT BOARDS —WITH 17 PROJECTS — **BKT18** — **$15.95**
Calvin R. Graf

In addition to general workshop principles, in this book Graf discusses tools and safety habits and offers a complete refresher course on electronics theory, schematic diagrams, and soldering. He also explains design and layout as well as numerous types of boards and wiring. Key subjects covered include: how to get from an electronic schematic to a printed circuit board; etching a printed circuit board; cleaning, drilling, and mounting electronic parts onto the board; soldering and desoldering components; a listing of commercially available electronic project kits; and the various types of diagrams used in electronics, including block, pictorial, layout, and wiring. 1988, 207pp., 7 × 9, softbound.

TROUBLESHOOTING AND REPAIRING AUDIO EQUIPMENT — **BKT19** — **$18.95**
Homer L. Davidson

This volume supplies all the basic information you need, along with specific examples, to fix stereo components, compact disc players, telephone answering machines, and much more. The detailed discussion of servicing and repair procedures for individual items includes probable causes of malfunctions and tips on difficult-to-diagnose problems. Scores of photos, diagrams, and drawings are used, as well as actual manufacturers' schematics and service literature. In addition to the equipment above, the book also covers auto CD players, compact cassette tape decks and portable stereo players, boombox cassette players, deluxe amplifiers, auto stereo cassettes, and stereo turntables and speakers. 1987, 325pp., 7 ½ × 9 ½, softbound.

ACOUSTICS: WAVES AND OSCILLATIONS — **BKW2** — **$34.95**
S. N. Sen

The result of the need in an honors course at Jadavpur University for a better text, this rare yet refreshing book supplements the physical bases of acoustics with mathematical and experimental details wherever possible. Topics include simple harmonic motion; theory of forced vibration and resonance; theory of coupled oscillations; vibration in an extended medium; vibration of strings, bars, tuning forks, membranes, rings, and air columns; reflection, refraction, diffraction, reception, and transformation of sound; sound measurement and analysis; acoustics of building; recording and reproduction of sound; and ultrasonics. India, 1990, 234pp., 6 ½ × 10, hardbound.

THE MUSICIAN'S GUIDE TO ACOUSTICS — **BKS35** — **$27.95**
Murray Campbell, Clive Greated

The result of the authors' fifteen-plus years of lecturing on musical acoustics to students at Edinburgh (U.K.) University, this book is a must for everyone with some musical background who feels the need for a clearer understanding of the practical basis of their art. While reasonable familiarity with musical notation and terminology is required to make best use of the book, only elementary mathematics is needed. Chapters include The Creation and Transmission of Musical Sounds; Hearing Musical Sounds; Anatomy of a Musical Note; Playing in Tune; Sound Production in Musical Instruments; Bowed and Plucked Stringed Instruments; Stringed Keyboard, Woodwind, Brass, and Percussion Instruments (one chapter each); Organs; The Human Voice; Electronic Instruments; and The Musical Environment. 1987, 612pp., 6 ½ × 9 ½, hardbound.

SEMICONDUCTOR CROSS REFERENCE DATA BOOK — **BKS37** — **$19.95**
Engineers of Howard W. Sams & Company

Old Colony Sound Lab uses cross references all the time, every day. Of all the versions available for semiconductors, this is easily one of the very best. Sams' engineers have assembled replacement data for over 319,000 registered type and part numbers, including those for the United States, Europe, and the Far East, and present it here in a very easy-to-read and use format. Four suppliers are featured: NTE Electronics, Radio Shack, Philips ECG, and RCA. Semiconductors covered include bipolar transistors, FETs, diodes, rectifiers, ICs, SCRs, LEDs, modules, and thermal devices. 1991, 520pp., 8 ½ × 11, softbound.

THE RECORD SHELF GUIDE TO THE CLASSICAL REPERTOIRE — **BKSM1** — **$16.95**
Jim Svejda

From the host of American Public Radio's "The Record Shelf," this classy book is an irreverent, selective, and highly opinionated recordings guide to the best in CDs, LPs, and cassettes. Just as on his radio show, the author tries to include something pointed, interesting, amusing, or enlightening about each work, and then suggests which of the available recordings should be bought. 1988, 462pp., 6 × 9, softbound.

THE FOUNDATIONS OF ACOUSTICS: BASIC MATHEMATICS AND BASIC ACOUSTICS — **BKSV2** — **$189.95**
Eugen Skudrzyk

The goal of this book is to provide the acoustician with a mathematically-based insight into the physical phenomena of acoustics. Combining into one volume such subjects as mathematics, dynamics, hydrodynamics, physics, statistics, signal processing, and electrical theory, the author comes up with a powerful reference for the academic or professional acoustician. Well illustrated. Austria, 1971, 790pp., 7 ½ × 10, hardbound.

AUDIO AMPLIFIER CONSTRUCTION — **BKEV10** — **$7.95**
R. A. Penfold

The purpose of this book is to provide the reader with a wide range of preamplifier and power amplifier designs that will hopefully fill almost anyone's needs. The preamp circuits include low noise microphone and RIAA types, a tape head preamp, a guitar preamp, and various tone controls. The power amplifier designs range from low-power, battery-operated to 100W MOSFET types. Also included is a 12V bridge amp capable of giving up to 18W output. All of the circuits are relatively easy to construct using the PCB or stripboard designs given. Where necessary any setting-up procedures are described, but in most cases no setting-up or test gear is required in order to successfully complete the project. Most of the designs should be within the capabilities of constructors with limited experience as well as more advanced hobbyists. United Kingdom, 1983, 128pp., 4 ⅜ × 7, softbound.

AUDIO ENTHUSIAST'S HANDBOOK — **BKEV12**
B. B. Babani — **$4.95**

This book discusses a number of audio and hi-fi topics, including record/playback curves, stylus compliance, vinyl recordings then and now, evaluating loudness, acoustic feedback, hi-fi vs. stereo, stereo tape track standards, equipment performance figures, and tracking error's cause, effect, and cure. United Kingdom, 1975, 96pp., 4 ⅜ × 7, softbound.

BUILD YOUR OWN SOLID STATE HI-FI AND AUDIO ACCESSORIES — **BKEV13** — **$4.95**
M. H. Babani, editor

This volume presents the reader with the design and construction plans for a variety of useful projects, including a stereo decoder, three-channel stereo mixer, FET preamp for ceramic PUs, microphone preamp with adjustable bass response, stereo dynamic noise filter, speaker protector, and voice-operated relay. United Kingdom, 1976, 96pp., 4 ⅜ × 7, softbound.

DIGITAL AUDIO PROJECTS — **BKEV14**
R. A. Penfold — **$9.95**

The first section of this book takes a look at the basic principles involved in converting an audio signal into digital form and then converting it back again to an analog signal. It also deals with some practical aspects that have to be borne in mind when considering digital audio projects. The second section contains some useful and extremely interesting practical circuits for constructors to build and experiment with. The projects are not highly complex, but nonetheless they are probably beyond the range of complete beginners. United Kingdom, 1988, 96pp., 4 ⅜ × 7, softbound.

ELECTRONIC CIRCUITS FOR THE COMPUTER CONTROL OF MODEL RAILWAYS — **BKEV15** — **$9.95**
R. A. Penfold

This innovative book is an examination of the problem of interfacing the computer to the layout. It includes projects consisting of various types of controllers, including a high-quality pulse type, as well as circuits for train position sensing, signal and electric points control, sound effects, and more. Projects are equally adaptable to large or small layouts. United Kingdom, 1986, 96pp., 4 ⅜ × 7, softbound.

MODEL RAILWAY PROJECTS — **BKEV16**
R. A. Penfold — **$6.95**

This handbook provides a number of useful but reasonably simple projects for the model railway enthusiast to build, including such things as controllers and signals and sound effects units. Stripboard layouts are provided for each project. United Kingdom, 1981, 112pp., 4 ⅜ × 7, softbound.

MOOEL RAILROAD SPECIAL! — **BKEV/RR** — **$14.95**

Both *Electronic Circuits for the Computer Control of Model Railways* (BKEV15) and *Model Railway Projects* (BKEV16), at a savings of $1.95!

INTERNATIONAL TRANSISTOR EQUIVALENTS GUIDE — **BKEV17**
Adrian Michaels — **$11.95**

This book, one of two transistor guides recommended by Old Colony, is designed to help the user find possible substitutes for a popular, user-oriented selection of modern transistors. It includes devices produced by over 100 manufacturers, and wherever possible the equivalents are subdivided into European, American, and Japanese types. Also shown are the material type, polarity, and manufacturer, as well as an indication of device use or application. United Kingdom, 1981, 1990, 320pp., 4 ⅜ × 7, softbound.

THE WOOD EFFECT — **BKMA1**
R. C. Johnsen — **$12.95**

According to the author, masked by random combination with other distortions in the music reproduction chain an unsuspected major contributor has lain hidden: aural sensitivity to "phase inversion," the Wood Effect. In this book the results of an extensive study of the Wood Effect are reported and discussed, including an exhaustive survey of the relevant literature. Although no means are proposed to correct countless errors, customary practice in acoustics and audio is severely scrutinized. BKMA1's subtitle is "Unaccounted Contributor to Error and Confusion in Acoustics and Audio." 1988, 99pp., 6 × 9, softbound.

THE HOMEBUILT DYNAMO: DYNAMO DESIGN AND CONSTRUCTION WITH CERAMIC MAGNETS — **BKTF1** — **$64.95**
Alfred T. Forbes

We added this book to our offerings not just because it is a physically handsome volume, but also because it is unique and, as many reviewers have pointed out, an experimenter's "delight." In 1969, Al Forbes and his wife decided to live the simple life and retreated to a parcel of undeveloped land. Soon enough, they needed a generator. This book is the extremely detailed account of how Al built one from scratch. But it is more than that: it's a head-on encounter with just about every principle of electricity, magnetism, and hands-on construction known to man. Along the way, incidentally, the reader runs into how to build precision wire-winding jigs, a small lifting magnet, a foot-powered version of the dynamo, a diamond saw, a sheet metal cutter, and a 139-pound flywheel. With more than 300 illustrations, this volume is slowly becoming a worldwide cult favorite among do-it-yourselfers everywhere. New Zealand, 182pp., 8 ½ × 12, hardbound.

THE VTL VACUUM TUBE LOGIC BOOK — **BKVT1**
David Manley — **$12.95**

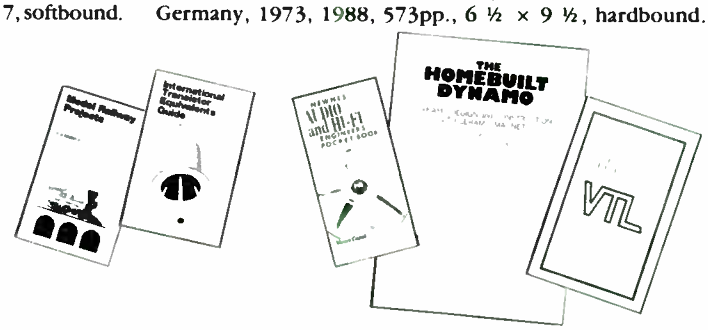
This third edition of every tube freak's must-have book is replete with tube matter of every description, including circuits, specifications, reviews, and history, presented with more than a liberal dash of humor and puckishness. 1991, 120pp., 5 ½ × 8 ½, softbound.

AUDIO AND HI-FI ENGINEER'S POCKET BOOK — **BKHN1** — **$27.95**
Vivian Capel

This book is a concise collection of practical and relevant data for anyone working on sound systems. The topics covered include microphones, record players, compact discs, tape recording, high-quality radio, amplifiers, loudspeakers, and public address systems. A lengthy section on acoustics is included for dealing with most aspects a technician is likely to encounter, from human hearing to sound insulation, and qualities such as heat and magnetism are covered as well. This is a pocket book, in a very handy size. United Kingdom, 1988, 190pp., 3 ¾ × 7 ¾, hardbound.

STRUCTURE-BORNE SOUND — **BKSV4**
L. Cremer, M. Heckl — **$84.95**

Translated from the German and revised by E. E. Ungar, this second edition is a thorough introduction to structural vibrations, with emphasis on those at audio frequencies, and the attendant radiation of sound. The book presents in-depth discussions of fundamental principles and basic problems, in order to enable the reader to understand and solve his own. Included are chapters on the measurement and generation of vibrations and sound; the various types of structural wave motions; structural damping and its effects; impedances and vibration responses of the important types of structures; the attenuation of vibrations, and sound radiation from structures. Germany, 1973, 1988, 573pp., 6 ½ × 9 ½, hardbound.



BUY FOUR, TAKE 15% OFF! SALE ENDS JULY 1!

OLD COLONY SOUND LAB

HOW TO USE OP AMPS — **BKEV3**
E. A. Parr — **$9.95**

Parr's book is a designer's guide covering many operational amplifiers which serves both as a source book for circuits and as a reference book for design calculations. The approach is as nonmathematical as possible and is thus equally understandable by the hobbyist or the engineer. United Kingdom, 1982, 160pp., 4 ⅜ × 7, softbound.

AUDIO PROJECTS — **BKEV4**
F. G. Rayer — **$6.95**

This book covers in detail the construction of a wide range of audio projects, including preamps, power amps, mixers, tone controls, and many others. All are fairly easy to build, and to that end the author has included a number of board layouts and wiring diagrams. United Kingdom, 1981, 96pp., 4 ⅜ × 7, softbound.

AN INTRODUCTION TO LOUDSPEAKERS AND ENCLOSURE DESIGN — **BKEV5**
V. Capel — **$9.95**

In this volume may be all you need to know about the theory and operation of speakers and the various boxes into which they may be fitted. The book also includes the complete design and construction details for the inexpensive but high quality enclosure called the "Kapellmeister." United Kingdom, 1988, 160pp., 4 ⅜ × 7, softbound.

MODERN OP-AMP PROJECTS — **BKEV6**
R. A. Penfold — **$6.95**

BKEV6 includes a wide range of construction projects which make use of the specialized operational amplifiers available today, including low noise, low distortion, ultra-high input impedance, low slew rate, and high output current types. Circuits using transconductance types are also included. United Kingdom, 1982, 112pp., 4 ⅜ × 7, softbound.

HOW TO GET YOUR ELECTRONICS PROJECTS WORKING — **BKEV7**
R. A. Penfold — **$6.95**

We have all built circuits from magazines and books only to find that they did not work correctly, or at all, when first switched on. The aim of this book is to help the reader overcome just these problems by indicating how and where to start looking for many of the common faults that can occur when building projects.

Chapter 1 deals with mechanical faults such as tracing dry joints, short circuits, broken PCB tracks, and so forth. The construction and use of a tristate continuity tester to help in the above is also covered. Chapter 2 deals with linear analog circuits and also covers the use and construction of a signal injector/tracer which can be used to locate and isolate faulty areas.

Chapter 3 considers ways of testing the more common components such as resistors, capacitors, op amps, diodes, transistors, SCRs, unijunctions, and so forth, with the aid of only a limited amount of test equipment. Chapter 4 deals with both TTL and CMOS logic circuits and includes the use and construction of a pulse generator to help in fault-finding. United Kingdom, 1982, 96pp., 4 ⅜ × 7, softbound.

CONCERT HALL ACOUSTICS — **BKSV3**
Yoichi Ando — **$69.95**

Number 17 in the Springer Series in Electrophysics, this book provides an interdisciplinary approach to solving acoustic design problems in concert halls. Considered one of the original sources in this field, the volume explains in detail the importance and interrelationship of total sound energy, delay of early reflections, reverberation, and the "spatial-binaural criterion," a measure of the spatial impression of the sound field. Germany, 1985, 151pp., 6 ¼ x 9 ⅜, hardbound.

ACOUSTICS SOURCE BOOK — **BKMH5**
Sybil P. Parker, editor-in-chief — **$39.95**

This widely-sought member of the McGraw-Hill Science Reference Series is an accumulation and explanation of all that has to do with acoustics, focusing on basic concepts, sound production, sound transmission, sound detection, and practical applications. One of the clearest and most basic references on acoustics available today. 1988, 333pp., 7 × 9 ½, hardbound.

LISTENING: AN INTRODUCTION TO THE PERCEPTION OF AUDITORY EVENTS — **BKMT1** — **$39.95**
Stephen Handel

This book combines broad coverage of acoustics, speech and music perception, psychophysics, and auditory physiology in a lively introduction to the perception of music and speech events. Coherence and clarity are the hallmarks of this book, and the author's strategy is to discuss specific points in detail rather than every possible thing superficially. All in all very fascinating, especially the treatment of the physiology and neurophysiology of the auditory system. 1989, 597pp., 6 ½ × 9 ½, hardbound.

AUDITORY PERCEPTION — **BKMX1**
F. Alton Everest — **$159.95**

This thorough, inventive, and understandable audio training course on psychoacoustics comes complete with eight lessons on four tapes and a manual with hundreds of technical diagrams illustrating the concepts. The aural examples and classic experiments enable the student to clearly and quickly comprehend the complexity of the hearing process. Topics include the perception of delayed sounds, auditory filters, masking of critical bands, the perception of pitch, timbre, and much more. 1986, 104pp., 5 ½ × 8 ½, softbound, 4 cassettes, vinyl case for all.

POWER ELECTRONICS HANDBOOK: COMPONENTS, CIRCUITS, AND APPLICATIONS — **BKB4** — **$79.95**
F. F. Mazda

The purpose of this book is to provide all the information required by power electronics engineers. It describes the design of power circuits used for a variety of applications, the characteristics of power semiconductor devices, and how they are used in power circuits. The author's approach is to give the maximum amount of information in a concise form, with the emphasis on the practical rather than the theoretical. United Kingdom, 1990, 417pp., 6 ½ × 9 ½, hardbound.

AUDIO ELECTRONICS REFERENCE BOOK — **BKBL1**
Alan R. Sinclair, editor — **$164.95**

Written by a team of expert, specialist contributors, this volume is of interest and profit to audio design and service engineers and technicians, as well as any amateur users of audio equipment who want to learn the technology of the art. BKBL1 is both a summary and a guide to the new state of audio, and includes in its coverage those areas where professional and amateur audio have begun to overlap, such as electronic music and public address systems. United Kingdom, 1989, 615pp., 6 ½ × 9 ½, hardbound.

TROUBLESHOOTING AND REPAIRING COMPACT DISC PLAYERS — **BKT20** — **$18.95**
Homer L. Davidson

This is an invaluable reference for both the electronics technician and the do-it-yourselfer alike. With it, you'll be able to troubleshoot and repair servo control loops, remote control systems, optical lenses and laser assemblies, and much more! Detailed examples of actual repair and adjustment procedures and a chapter on the care and handling of the discs themselves further enhance the usefulness of this book. 1989, 337pp., 7 ½ × 9 ½, softbound.

INCREDIBLE INTRODUCTORY BOOK SALE!

LOUDSPEAKERS, VOLUME 1 — **BKAS1/1**
Raymond E. Cooke, editor — **$29.95**

From The Audio Engineering Society's Anthology Series, 61 papers, covering the years 1953 to 1977, written by the world's greatest transducer experts and inventors on the design, construction, and operation of loudspeakers. 1980, 448pp., 8 ¼ × 11 ¼, softbound.

LOUDSPEAKERS, VOLUME 2 — **BKAS1/2**
Raymond E. Cooke, editor — **$29.95**

Also from The Audio Engineering Society's Anthology Series, 49 papers from 1978 to 1983 by experts in loudspeaker technology, extending the work initiated in Volume 1. 1984, 464pp., 8 ¼ × 11 ¼, softbound.

LOUDSPEAKERS SET! — **BKAS/S**
Raymond E. Cooke, editor — **$53.95**

From The Audio Engineering Society's Anthology Series, both BKAS1/1 (Volume 1) and BKAS1/2 (Volume 2) as above, at a savings of $5.95!

MICROPHONES — **BKAS2**
Louis A. Abbagnaro, editor — **$29.95**

Sixty-three papers covering calibration and testing, general purpose microphones, directional microphones, miniature types, and associated electronic circuits. From The Audio Engineering Society's Anthology Series. 1979, 392pp., 8 ¼ × 11 ¼, softbound.

SOUND REINFORCEMENT — **BKAS3**
David L. Klepper, editor — **$29.95**

Seventy-three papers dealing with the significant aspects of the development of sound-reinforcement technology and its practical application to sound system design and installation. From The Audio Engineering Society's Anthology Series. 1978, 339pp., 8 ¼ × 11 ¼, softbound.

STEREOPHONIC TECHNIQUES — **BKAS4**
John M. Eargle, editor — **$29.95**

From The Audio Engineering Society's Anthology Series, 67 articles and documents on the history, development, and applications of stereophonic techniques for studio technology, broadcasting, and consumer use. 1986, 390pp., 8 ¼ × 11 ¼, softbound.

SPICE: A GUIDE TO CIRCUIT SIMULATION & ANALYSIS USING PSPICE — **BKPH2** — **$19.95**
Paul W. Tuinenga

Designed as a reference on PSpice for the design and analysis of analog circuits, this book clearly explains how to use the features of PSpice to solve common electrical and electronic problems, as well as some in nonelectrical areas. Topics include DC operation, transfer functions, frequency response, and noise analysis. SPICE is an acronym for Simulation Program with Integrated Circuit Emphasis, and PSpice is a SPICE-derived simulator created by MicroSim Corporation. Companion software for the book is available (see below). 1988, 200pp., 7 × 9 ½, softbound.

SPICE BOOK/SOFTWARE SPECIAL! — **BKPH2/S** — **$27.95**

Student-version software to accompany BKPH2 is available for both the IBM PC and Macintosh II. The PSpice program can simulate circuits of up to five nodes and ten transistors, with the parameters inserted by the user. Special BKPH2/S includes both the book and one software package at a savings of $4.95!

Software packages available (be sure to specify!):
SOF-SPC1B5GD IBM PC
SOF-SPC2B5GD IBM PS 2
SOF-SPC1M3GD Macintosh II

TIME DELAY SPECTROMETRY — **BKAS5**
John R. Prohs, editor — **$29.95**

From The Audio Engineering Society's Anthology Series, 32 articles of the works of Richard C. Heyser on measurement, analysis, and perception, reprinted from the pages of the *Journal of the Audio Engineering Society* and other publications representative of the field, including *Audio* magazine and that of IREE Australia. The anthology serves as a memorial to the author's work and as fundamental material for future developments in audio, and will undoubtedly provide the stimulus for expanded discussion. 1988, 280pp., 8 ¼ × 11 ¼, softbound.

AUDIO IN DIGITAL TIMES: CONFERENCE PROCEEDINGS — **BKAS6** — **$34.95**

Forty-four papers presented by experts on digital audio at The Audio Engineering Society's Seventh International Conference held in Toronto on 14–17 May 1989. Digital audio, from the history, basics, hardware, and software, to the ins and outs, was the topic of the conference. Illustrated with many figures and tables. 1990, 384pp., 8 ¼ × 11 ¼, softbound.

THE SOUND OF AUDIO: CONFERENCE PROCEEDINGS — **BKAS7** — **$34.95**

Twenty-four papers presented by authors highly regarded in the engineering community at The Audio Engineering Society's Eighth International Conference held in Washington, D.C., on 3–6 May 1990. The topics were devoted to the progress of sound, including measurement, recording, and reproduction. Textbook style, fully illustrated. 1990, 384pp., 8 ¼ × 11 ¼, softbound.

DIGITAL AUDIO: COLLECTED PAPERS — **BKAS8**
Barry Blessner, et al., editors — **$34.95**

First publication of papers presented at The Audio Engineering Society's Premiere International Conference held in Rye, N.Y., on 3–6 June 1982, authored by the world's leading experts in the application of digital techniques in the field of audio engineering. Twenty-five of the 27 papers are transcribed, edited, and published for the first time. Subjects include basics, converters, measurements, rate conversion, recording formats, error correction, manufacturing, and applications. Includes a Soundsheet disk with audio demonstrations of digitally synthesized sounds. 1983, 268pp., 8 ¼ × 11 ¼, softbound.

MICROPHONES: TECHNOLOGY AND TECHNIQUE — **BKB3**
John Borwick — **$29.95**

Beginning with a brief history of the relevant technology, this book then goes on to explain the basic theory of acoustics, electricity, and magnetism. The working principles and design of all types of microphones are explained in considerable detail, with examples of popular current models and descriptions of microphone accessories. The second half of the book provides guidelines on the creative balance techniques to be used for musical instruments, voices, and ensembles of all kinds, in both classical and pop music. Production methods are outlined both for studios and for on location, with notes on public address operations for live shows. Borwick is considered THE authority. United Kingdom, 1990, 241pp., 7 ½ × 9 ½, softbound.

THE ART OF ELECTRONICS — **BKC1**
Paul Horowitz, Windfield Hill — **$54.95**

Completely up-to-date with the latest technology and standards, this second edition features completely rewritten chapters on microcomputers and microprocessors, digital electronics, and low-power and micro-power design (both analog and digital). Many new tables have been added, including ones for A/D and D/A converters, digital logic components, and low-power devices. The quintessential electronics text and reference. 1989, 1100pp., 7 ½ × 10 1¼, hardbound.

HARD-TO-FIND BARGAINS FROM AROUND THE WORLD!

ACOUSTICAL MEASUREMENTS — **BKAC1**
Leo L. Beranek — **$44.95**

Here is a completely revised edition of the famous text by the well-known authority, with more than half the pages and chapters revised or completely rewritten to cover new developments in acoustical instruments and measurement procedures, including sound and vibration meters, intensity meters, spectrum analyzers, recorders, computers, and noise measurement devices. 1948, 1989, 841pp., 5 ¾ × 8 ¾, hardbound.

ACOUSTICS—AN INTRODUCTION TO ITS PHYSICAL PRINCIPLES AND APPLICATIONS — **BKAC2**
Allan D. Pierce — **$37.95**

This volume introduces the physical principles and theoretical basis of acoustics, concentrating on those concepts and points of view that have proven useful in applications and phenomena such as noise control, underwater sound, architectural acoustics, audio engineering, nondestructive testing, remote sensing, and medical ultrasonics. The text is supplemented by problems and their answers. 1981, 1989, 678pp., 6 ¼ × 9 ½, hardbound.

EXPERIMENTS IN HEARING — **BKAC3**
Georg von Békésy — **$27.95**

Considered obligatory reading for all those who want to claim literacy in the auditory sciences, this classic on hearing contains many of the vital roots of contemporary auditory knowledge. 1960, 1989, 760pp., 6 × 9, softbound.

ACOUSTICAL DESIGNING IN ARCHITECTURE — **BKAC4**
Vern O. Knudsen, Cyril M. Harris — **$27.95**

This book is a comprehensive, nonmathematical treatment of architectural acoustics, completely revised with new illustrations and updated to eliminate obsolete materials. In handbook format, it covers all the general principles of acoustical designing and includes specific applications. 1950, 1980, 408pp., 5 3/8 × 8 ½, softbound.

ACOUSTICS — **BKAC5**
Leo L. Beranek — **$27.95**

This volume is an indispensable source of practical acoustical concept and theory for acoustical and electrical engineers, scientists, and consultants, with new information on microphones, loudspeakers and speaker enclosures, room acoustics, and acoustical applications of electromechanical circuit theory. 1954, 1986, 491pp., 6 × 9, softbound.

MARQUETRY AND INLAY: TWENTY DECORATIVE PROJECTS — **BKT21**
Alan and Gill Bridgewater — **$15.95**

Complete with 200 illustrations, a glossary, and an introduction to tools and techniques, BKT21 provides step-by-step directions for easy-to-do projects. This book gives woodworkers an opportunity to add a valuable new skill to their repertoire—inserting fine woods and decorative veneers into finished surfaces. 1989, 192pp., 7 × 10, softbound.

TONMEISTER TECHNOLOGY: RECORDING ENVIRONMENTS, SOUND SOURCES AND MICROPHONE TECHNIQUES — **BKTE1**
Michael Dickreiter — **$19.95**

Recently translated from the German by Stephen F. Temmer, this rare book for the first time provides an English-language reference integrating music and musical engineering. Used by the German Broadcasting System Technical Training Center as a text to train people who graduate from a music conservatory and opt to go into broadcast, TV, recording, or sound reinforcement engineering, this book will appeal to neophytes as well as professionals in the music and audio engineering fields. Profusely illustrated, with every other page illustrations. 1984, 1989, 142pp., 6 × 9, softbound.

HEARING: ITS PSYCHOLOGY AND PHYSIOLOGY — **BKAC6**
Stanley Smith Stevens, Hallowell Davis — **$27.95**

This multidisciplinary book leads readers from the fundamentals of the psychophysiology of hearing to a complete understanding of the anatomy and physiology of the ear, including the relationship between stimulus and sensation. 1938, 1983, 512pp., 5 ½ × 8 ½, softbound.

ELECTROACOUSTICS: THE ANALYSIS OF TRANSDUCTION AND ITS HISTORICAL BACKGROUND — **BKAC7**
Frederick V. Hunt — **$27.95**

This volume provides a comprehensive analysis of the conceptual development of electroacoustics, including the origins of echo ranging, the crystal oscillator, the evolution of the dynamic loudspeaker, and electromechanical coupling. 1954, 1982, 260pp., 5 ½ × 8 ½, softbound.

VIBRATION AND SOUND — **BKAC8**
Philip M. Morse — **$27.95**

One of the most widely used texts and references on the science of acoustics, this volume provides students and professionals alike with a broad spectrum of acoustics theory, including wave motion, radiation problems, the propagation of sound waves, and transient phenomena. 1936, 1981, 468pp., 6 × 9, softbound.

ACOUSTICS/HEARING SPECIAL #1 — **BKAC/X2**
$49.95

Your choice of TWO from BKAC3 through BKAC8, at a savings of $5.95! Be sure to specify books. (BKAC1 and BKAC2 not eligible.)

ACOUSTICS/HEARING SPECIAL #2 — **BKAC/X3**
$74.95

Your choice of THREE from BKAC3 through BKAC8, at a savings of $8.90! Be sure to specify books. (BKAC1 and BKAC2 not eligible.)

TOTAL HARMONIC DISTORTION —CARTOONS FROM STEREO REVIEW — **BKPB1**
Charles Rodrigues — **$7.95**

This hilarious collection of 118 of the author's classic cartoons on audio, video, and music is the perfect gift and/or a wonderful conversation piece. Also featured are a foreword by William Livingstone and an illustrated autobiographical sketch by Rodrigues himself. One of a kind. 1988, 132pp., 6 ½ × 7, softbound.

SHEET METAL HAND PROCESSES — **BKDE2**
Claude J. Zinngrabe, Fred W. Schumacher — **$18.95**

For help in building that special enclosure, this volume will prove indispensable, introducing the beginning sheet metal technician to basic instruction in the properties and selection of metal and the processes performed with measuring and marking, bench, cutting, piercing, joining, and soldering tools. BKDE2 includes an excellent selection of illustrations, and every process is described in an easy-to-understand, step-by-step fashion. 1974, 209pp., 7 ¾ × 10 ¼, softbound.

CABINETMAKING, PATTERNMAKING, AND MILLWORK — **BKDE4**
Gaspar J. Lewis — **$34.95**

If making your own speaker cabinets has led you into woodworking, this book will be a great addition to your library. Designed for advanced high school, vocational, technical, and apprenticeship programs, it provides the reader with the skills necessary for proficiency in each of the three areas of specialization. Also included are three sections common to all three fields: the nature of wood and its uses; how to use hand tools, portable power tools, and stationary woodworking machines; and methods of joinery. 1981, 438pp., 7 ½ × 9 ½, hardbound.

BUY FIVE OR MORE, TAKE 20% OFF! SALE ENDS JULY 1!

BEGINNER'S GUIDE TO READING SCHEMATICS — **BKT14** — **$11.95**
Robert J. Traister
This outstanding handbook takes you step-by-step through every phase of understanding and using schematics. Coverage includes how and why schematics are used; how each symbol is derived, used, and drawn; how individual symbols are combined to represent electronic circuits; proper numbering of components; and units of electronic measurement. 1983, 134pp., 5 × 8 ½, softbound.

GREAT SOUND STEREO SPEAKER MANUAL —WITH PROJECTS — **BKT15** — **$16.95**
David B. Weems
In this book, Weems strips the mystery from drivers, crossovers, and construction materials in order to help you build your own speakers and get the best possible performance from all of your sound equipment. Included are current trends in speaker design, getting great stereo sound from units you build yourself, calculating design parameters, transmission- line systems, crossover networks, and much more! 1990, 248pp., 7 ½ × 9, softbound.

PRACTICAL STEREO AND QUADRAPHONY HANDBOOK — **BKEV11** — **$4.95**
B. B. Babani
A reference book for all those interested in modern stereophonic and multi-sound-channel equipment. The subjects covered include explanations of many of the technical terms used in this audio field, stereo equipment and techniques, positioning of multiple loudspeakers for best results, stereo and four-channel pickups, and more. United Kingdom, 1975, 96pp., 4 ⅜ × 7, softbound.

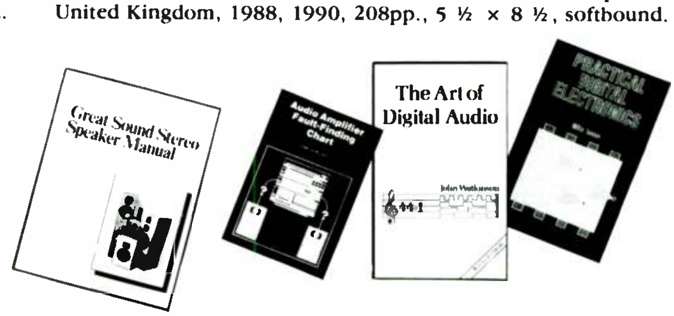
AUDIO SYSTEM DESIGN AND INSTALLATION — **BKS33** — **$59.95**
Philip Giddings
This book is a practical, hands-on tool designed to help the audio professional find information quickly. The many useful tables, checklists, photos, and diagrams included are all intended to improve and expedite system design. Giddings also provides many effective tips and strategies for efficient audio system installation. 1990, 350pp., 7 ⅜ × 9 ¼, hardbound.

AUDIO (BABANI ELEMENTS OF ELECTRONICS, BOOK 6) — **BKEV8** — **$11.95**
F. A. Wilson
In this book, analysis of the sound wave and an explanation of acoustical quantities prepare the way. These are followed by a study of the mechanism of hearing and examination of the various sounds we hear. A look at room acoustics with a subsequent chapter on microphones and loudspeakers then sets the scene for the main chapters on audio systems, amplifiers, oscillators, disc and magnetic recording, and electronic music. United Kingdom, 1985, 320pp., 4 ⅜ × 7, softbound.

AUDIO AMPLIFIER FAULT-FINDING CHART — **BKEV9** — **$4.95**
C. E. Miller
This tool is actually a foldable wall chart which will help the user trace most common faults which might occur in audio amplifiers. All the reader has to do is select one of the faults shown at the top of the chart and then follow the arrows, carrying out the suggested checks in sequence until the fault is cleared. United Kingdom, 1987, 17 ½ × 25.

THE ART OF DIGITAL AUDIO — **BKB2** — **$52.95**
John Watkinson
This book begins with clear definitions of technical terms specific to particular technologies employed in digital audio. The theory of each area is then discussed and their practical applications considered. All of the major digital recording formats are explained. Consumer machines, compact discs, DAT, and mastering recorders are covered, as well as multitracking. The professional's first digital audio reference, written in laymen's terms. United Kingdom, 1988, 500pp., 6 ½ × 9 ½, hardbound.

PRACTICAL DIGITAL ELECTRONICS HANDBOOK — **BKPC1** — **$14.95**
Mike Tooley
Aimed at amateurs and professionals alike, this book features nine digital test gear projects, pinouts for CMOS and TTL devices, and numerous tables of reference data. It introduces digital circuits, logic gates, bistables, timers, microprocessors, memory, and input/output devices, as well as looks at the RS-232C interface and the IEEE-488 and IEEE-1000 microprocessor buses. United Kingdom, 1988, 1990, 208pp., 5 ½ × 8 ½, softbound.



YES! Please send me

_____ BK_____ @ $_____ = $_____
_____ BK_____ @ _____ = _____
_____ BK_____ @ _____ = _____
_____ BK_____ @ _____ = _____
_____ BK_____ @ _____ = _____
_____ BK_____ @ _____ = _____

SUBTOTAL = _____
LESS SALE DISCOUNT = _____
TOTAL = _____
PLUS SHIPPING (see right) = _____
TOTAL ORDER = $_____

SALE ENDS JULY 1!

OLD COLONY SOUND LAB
PO Box 243, Dept. B91
Peterborough, NH 03458
24-HOUR LINES:
TEL. (603) 924-6371
FAX (603) 924-9467

NAME _____ MAGIC NO. _____
STREET & NO. _____
CITY _____ STATE _____ ZIP _____
MC/VISA _____ EXP. DATE _____

PAYMENT BY: ☐ MC/VISA ☐ CHECK ☐ MONEY ORDER

SHIPPING: In US, please add $2 for first book, 75¢ each additional. In Canada, please add $4.50 for first book, 75¢ for each additional. Others, please add 20% for surface delivery. Airmail: Please inquire.

FOREIGN CHECKS MUST BE IN US FUNDS DRAWN ON US BANK.

CALL OR WRITE FOR YOUR FREE CATALOG!

Kit Report

Synthesized Bandpass Subwoofers

By Gary Galo, Contributing Editor

Sub-1, Audio Concepts, Inc., 901 S. 4th St., La Crosse, WI 54601, (800) 346-9183 (orders only), FAX (608) 784-6367, (608) 784-4570 (customer service).

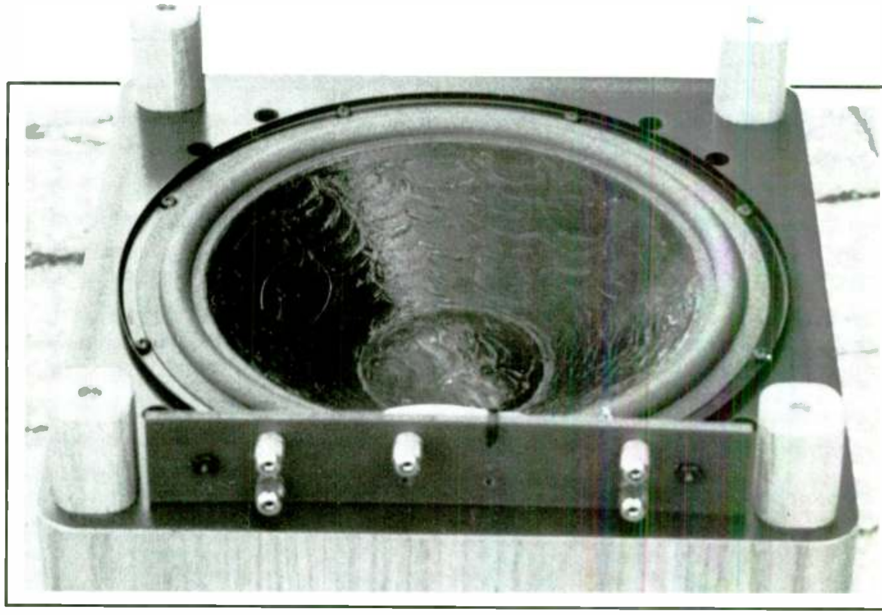


PHOTO 2: Bottom view of the Sub-1 showing the modified JC-12 woofer.

At the end of my review of Audio Concepts' marvelous Sapphire II loudspeaker (*SB* 6/90), I promised to ask AC president Mike Dzurko for a pair of Saturn subwoofers for review. Audio Concepts has discontinued the Saturns and replaced them with the Sub-1, a completely different design from the push-pull, compound Saturn.

Each Saturn enclosure contained a pair of Audio Concepts' AC-12 woofers in a large floor-standing enclosure. If you used the Saturns with Audio Concepts' outboard passive crossover, subwoofer positioning was critical. System integration was most successful if you placed the subwoofers to the sides of the listening position, rather than next to the satellite loudspeakers. This arrangement was not always convenient, however. Also, assembly required critical attention to the seal between the two woofers. If any leakage occurred, the compound system failed to operate as designed.

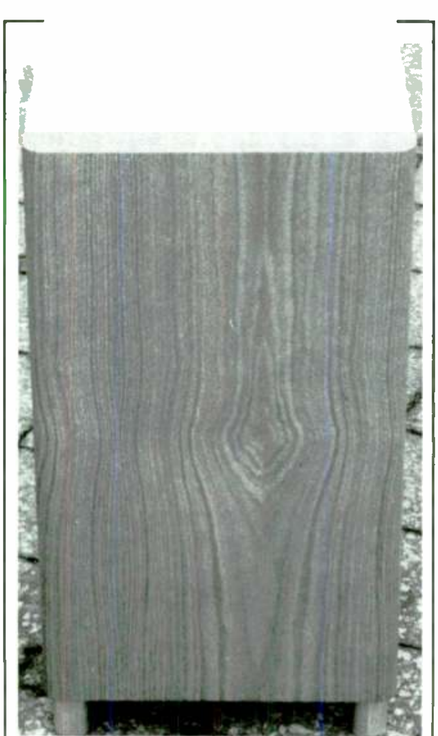


PHOTO 1: Top view of Audio Concepts' Sub-1 subwoofer.

The Sub-1 is the result of two years of design work by Mike Dzurko. It is an aperiodic design using a modified Audio Concepts JC-12 woofer. In addition to offering performance superior to the Saturn, the Sub-1 is smaller, easier to construct, and less critical of placement. Best of all, the Sub-1s are $300/pair less than the Saturns.

The JC-12 is a 12″ dual voice coil driver with a long fiber cone, foam surround, and a rear-vented pole piece. It is a superbly constructed woofer with voice coil lengths of 50mm and a peak linear excursion (X_{MAX}) of 10mm. This is the longest peak excursion I've found in any readily available 12″ driver (even the 12″ Dynaudios are limited to 8mm).

Like the other woofers in the Audio Concepts line, the JC-12 was designed specifically for sealed and aperiodic enclosures. (Like me, you may be frustrated because woofers with Thiele/Small parameters optimized specifically for non-vented enclosures are in the minority. I suggest looking at the Audio Concepts drivers next time you need such a woofer.)

The JC-12s have been modified by a coat of silicone seal applied to the cone and dust cap, forming a series of swirl patterns on the cone. The silicone application has several effects beneficial to the Sub-1 design; it further dampens the cone and seals the dust cap, both resulting in lower distortion. The damping also adds mass to the cone, lowering the free-air resonance (f_S). I measured the JC-12s supplied with the review kit and found f_S to be 15Hz.

In the Sub-1, the JC-12's dual voice coils are connected in parallel. The Sub-1s are sold in pairs and you must use them that way. You can't use a single Sub-1 as a mono subwoofer. From a perfectionist point of view, I have never believed in mono subwoofers, especially if they are crossed over as high as 80–100Hz.

Photos 1 and *2* show the Sub-1's top and bottom views; its woofers are downward-firing. Four posts support each Sub-1 enclosure, elevating the bottom by about 2″. The enclosures are heavily damped with AC Bonded Dacron, with two pieces measuring 6′ by 27″ folded into each. The enclosure's bottom contains six ½″ holes, covered on the inside with four pieces of ⅝″ foam, which provides the woofer's aperiodic damping.

The enclosures are made of ¾″ MDF and are extremely well-braced internally. Like the Sapphire IIs, the Sub-1s are beautifully crafted. The veneer work is first-class, and production pairs are supplied with matched grain.

The crossover components are mounted on separate Masonite boards, one for the low-pass section and one for the high-pass. These are on opposite sides of the enclosure to prevent magnetic crosstalk between the two sections. A special, 10AWG oxygen-free copper cable connects the

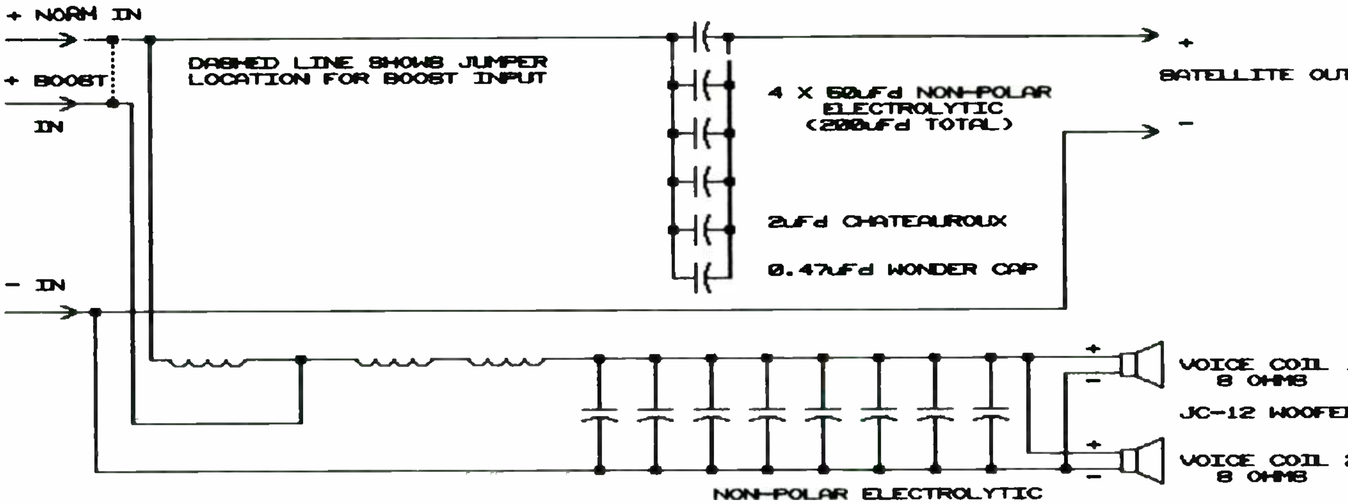


FIGURE 1: The Sub-1 crossover network. Do not use the boost input with Audio Concepts' satellites if you desire the most natural sound.

crossover, woofer, and terminals. The connectors are mounted on a ¼" piece of Masonite, mounted on the bottom rear of the enclosure. Audio Concepts uses the same high-quality, gold-plated posts supplied with the Sapphire II.

Crossover Design

Audio Concepts has a trademark on the term "synthesized bandpass." A bandpass filter contains low- and high-pass sections. Below system resonance, the response rolls off like a second-order high-pass filter. This is a normal characteristic of sealed-box and aperiodic designs. The system's natural high-frequency rolloff does not form the filter's low-pass portion. Instead, it is synthesized with a second-order passive crossover network. *Figure 1* shows the complete crossover.

We're dealing with a very low crossover frequency and a low impedance load (4Ω, since the JC-12's two voice coils are connected in parallel). This results in extremely large component values. The formulas for calculating series inductance are the same as for series resistance. However, magnetic coupling between inductors makes the formulas useless if the coils are too closely spaced. Audio Concepts has left sufficient spacing between the inductors to prevent this problem.

The low-pass section actually has two inputs. The boost input provides a 3dB increase in output in the 50–90Hz region. This may be helpful in matching satellites other than those made by Audio Concepts to the Sub-1s. The Sub-1 crossover has

Continued on page 54

Continued from page 52
been carefully designed to provide an ideal match with the Sapphire IIs and their smaller satellite, the Little Vs. When you use the Sub-1s with the Audio Concepts satellites, unless your room has a serious anomaly in the affected area, don't use the boost input because it raises the crossover frequency. You must install a jumper when you use this input, which bypasses the first inductor.

The high-pass section of the Sub-1 crossover is a simple first-order filter that begins electrically around 165Hz. When you take into account the Sapphire II satellite's response, the actual acoustic crossover point is around 100Hz. A capacitance of 200µF was required for such a low crossover frequency. Audio Concepts has used four 50µF nonpolar electrolytic capacitors to achieve this value.

To minimize the sonic problems caused by electrolytics in the midrange and high frequencies, the four capacitors are bypassed with a 2µF Chateauroux polypropylene cap and a 0.47µF WonderCap™. Purists may prefer only polypropylene capacitors here. Bear in mind that using polypropylenes instead of electrolytics adds nearly $100 to the cost and would probably not be audible unless the associated electronics were quite expensive. Mike says he considers replacing the electrolytics a legitimate tweak for those who believe the rest of their equipment will allow them to hear the difference. If you wish to do this, I suggest a 200µF Chateauroux polypropylene bypassed with the other two caps. I have not performed this modification to the review samples.

Assembly

The first pair of Sub-1s I received were factory assembled prototypes. After living with them for a few months, I ordered a complete Sapphire II/Sub-1 system for my production studio at The Crane School of Music. This studio is used primarily for preparation of tapes for our monthly radio series "Music from Crane," which I co-produce with our local public radio station. I needed an accurate monitor system for evaluating my live concert tapes of unamplified classical music (this is not a rock and roll studio). You're not supposed to know whether I like this system until you read the listening evaluations below. I guess I've just given you a hint.

Building the second system for my studio helped me prepare this review since I gained a few insights I can pass on to you. The Sub-1 kits are shipped in three well-packed cartons, one for each enclosure and the other containing the woofers, crossovers, stuffing, and remaining parts. The enclosures are double-boxed, which further protects them from shipping damage. The kit comes with a 10-page manual covering assembly, installation, and operating instructions. As a supplement to the manual, Audio Concepts has included a four-page description of the design philosophy behind the Sub-1. The Sub-1 is not available as a parts kit; you must purchase the full kit, including the enclosure.

The Sub-1 manual is fairly well-written, but I'd like to offer some helpful hints I discovered when building my pair. Audio Concepts' assembly instructions follow, in italics, along with my comments. Begin by turning the enclosures upside down,

Continued on page 57

Manufacturer's Specifications

Woofer: Modified Audio Concepts JC-12

Frequency response: 20–80Hz; low-frequency cutoff will vary from 18–22Hz, depending on room placement and dimensions

Crossover frequencies: 80Hz, second-order low-pass; 100Hz, first-order high-pass; f_c = 28Hz; Q = < 0.89

Dimensions: 23″ by 13 ¾″ by 14½″

Price: Full kit, $699 pair plus $50 shipping; special limited time introductory price, $599 plus shipping; assembled, add $100

The Audio Concepts, Inc. Synthesized Bandpass™ *Sub 1*

What if you could have a stereo pair of attractive, compact subwoofers that gave clean, precise response to below 20Hz with low distortion and outstanding impact?

Introducing the revolutionary *Sub 1* from Audio Concepts, Inc. At only 25" tall by 13" wide by 14" deep, the *Sub 1* is the first design using the "Synthesized Bandpass" (SB™), principle developed by Audio Concepts, Inc.

What is the "Synthesized Bandpass ?"
The SB™ utilizes a passive filter interacting with the driver and cabinet to approximate the characteristics of a vented bandpass design. However, the SB™ does not require multiple chambers and tuned ports, therefore the SB™ provides the dynamic range and compact size of a conventional bandpass while eliminating the tuning difficulties and resonances of ports and chambers.

What does the Sub 1 compare to?
The only competition for the Sub 1 comes from large transmission lines, giant vented enclosures and motional feedback systems. All competing systems are much larger and or much more costly.

How is the Sub 1 used in a system?
Each Sub 1 contains two completely separate crossover boards. One board feeds the subwoofer driver and one board connects to the outputs. All connections are heavy duty gold-plated five way posts and are mounted out of sight on the bottom of the enclosure. Connect your amplifier to the input posts and connect your main speakers to the output posts. The upper bass output level is adjustable via a jumper to compensate for your listening room and listening tastes.

What can the Sub 1 be used with?
The Sub 1 makes an excellent match with our own Sapphire II or Little V speakers. Package pricing is available. The Sub 1 can also be successfully matched with most speakers having a four to eight ohm load and a sensitivity of around 87 to 90 db. The Sub 1 presents an easy amplifier load and can be driven with as little as 40 watts per channel.

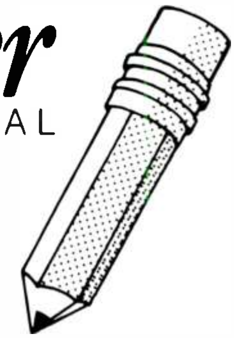
The Sub 1 is the finest subwoofer Audio Concepts, Inc. has produced. Low organ pedals, symphonic bass drums, and synthesizers are all reproduced with stunning realism by the downfiring 12" drivers. Equally important, the Sub 1 faithfully reproduces the subtle but musically critical qualities of difficult instruments such as piano and string bass.

All who have heard the Sub 1 agree, it is an exceptional subwoofer in a very attractive package. At $649 for a pair of full kits, ($749 assembled), including UPS Ground shipping, it is the absolute best value in high-performance subwoofers!

Call today to order or find out more!

Audio Concepts, Inc.
901 South 4th Street, La Crosse, WI 54601
Sales: (608) 784-4570 Fax: (608) 784-6367
Orders and Catalog requests: (800) 346-9183

SpeakerBuilder
THE LOUDSPEAKER JOURNAL



Subscriber Services

WOULD YOU LIKE TO . . .

Complete the applicable section(s) below and mail to:
Speaker Builder
PO Box 494
Peterborough, NH
03458-0494

☐☐☐☐☐☐
MAGIC NUMBER

☐ SUBSCRIBE?

YES! Just check the appropriate boxes in the "renew" section and fill in name and address.

☐ RENEW?

Fill in the appropriate sections below. For faster service use your mailing label and magic number.

☐ 1 year at $25
☐ 2 years at $45

☐ Check/MO enclosed

☐ Charge MC/VISA

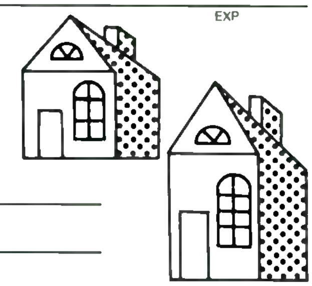
Canada add $4 per year postage
Remit in US $ only
Overseas rates on request

NAME ____________________
STREET & NO ____________________
CITY ____________ ST ____ ZIP ____
MC/VISA ____________________ EXP ____

☐ CHANGE YOUR ADDRESS?

Fill in your new address to the right and old address and magic number above.

MY NEW ADDRESS:



NAME ____________________
STREET & NO ____________________
CITY ____________ ST ____ ZIP ____

☐ GIVE A GIFT?

What a thoughtful idea! Fill out your name and address above, the recipient's name and address to the right. We'll even send a gift card announcing your gift.

NAME ____________________
STREET & NO ____________________
CITY ____________ ST ____ ZIP ____

☐ TELL YOUR SPEAKER BUILDING FRIENDS ABOUT US?

Print their names and addresses below. We'll take care of the rest.

NAME ____________________
STREET & NO ____________________
CITY ____________ ST ____ ZIP ____

NAME ____________________
STREET & NO ____________________
CITY ____________ ST ____ ZIP ____

YOU CAN CALL US
(603) 924-9464
Subscription Department
9-4 Business Days
Answering Machine
All Other Hours
FAX: (603) 924-9467
24 Hours

☐ COMPLAIN ABOUT YOUR SUBSCRIPTION?

Yes, it sometimes happens. We try to avoid it at all costs, but if you have a problem, we want to know. Call or write with all pertinent information.

Continued from page 54
preferably on a soft carpet so you won't scratch the veneer.

1. Carefully unpack your parts kit. Check to see that you have all the parts on the list. Remove the bag of hardware stapled to the inside of the enclosure. One of mine had fallen loose during shipment. Locate the staples used to fasten the bags to the inside of the enclosure and remove them.

2. Vacuum the inside of the cabinets to make sure they are totally free of any dust or chips.

3. Use the supplied nut and bolt pairs to attach the terminal strip to the two metal L-brackets. Attach these L-brackets to the bottom of the cabinet using supplied drywall screws (the holes are pre-tapped). The binding posts should be mounted to face outward.

The terminal strip mounts on the bottom of the enclosure, near the back edge. Only two aperiodic holes are near the back, and pre-drilled pilot holes are for the L-brackets. There's only one correct way to mount the terminal strip. Page 4 of the manual shows the five binding posts' correct orientation when the enclosure is turned right side up and you are facing the back. The edge with three gold posts will be closest to the floor, and the edge with two posts will be next to the enclosure's bottom.

4. Fold one of the pieces of supplied stuffing material and push it down into the lower half of the cabinet. Distribute evenly. Two internal braces around the enclosure's inside periphery separate the inside into three even spaces. Fold the first piece of stuffing material along its length, making a double-thick piece a little more than 12" wide. The piece should be 6' long. Cut off a 2' length. Fold the remaining 4' length and stuff it into the top third of the enclosure. Fold the 2' piece and stuff it into the middle third of the enclosure.

5. Use the supplied silicone sealant to attach the two crossover boards to the insides of your cabinets. When looking down at the open driver hole, position the cabinet so the terminal strip will be closest to you. The upper range crossover board should be mounted on the same side as the terminal strip, directly against the first brace. The lower range crossover board should be mounted on the opposite side in the opposite corner.

The wiring is extremely stiff. Handle the crossovers carefully to avoid breaking component leads and solder connections. The large inductors are heavy and may cause the crossover board to pull away from the enclosure before the silicone seal cures. Mount the crossover boards in two steps. Turn the enclosures on their sides, with the terminal strips toward the top. Apply a generous amount of silicone seal to the low-pass board's back side. Orient the board so the large 12mH coil is toward the woofer opening and press it into place onto the enclosure's side, against the internal brace. Allow the silicone to cure for at least eight hours before moving the enclosures.

Now turn the enclosures over so the terminal strip faces the floor. Mount the high-pass crossover the same way. Allow four hours for the silicone to cure (this board is much lighter and won't pull away as easily). I'm sorry to add an extra day to the assembly for the silicone to cure (I know you can't wait to listen to them), but you'll have a far more time-consuming problem on your hands if the boards come loose before the silicone cures.

6. Run the wires from the crossover boards up through the two holes in the bottom of the cabinet. Follow the crossover-terminal board diagram to make the correct connections to the terminal strip. Remove each loose locknut and washer from the backs of the binding posts. Install the correct flags on the posts and secure them with the washers and nuts. Seal the wires into the holes using the supplied silicone sealant.

Refer to page 4 of the manual and make sure the wires are connected to the correct binding posts. Apply your favorite contact cleaner to the gold flags before fastening them to the posts. Be sure the nuts are tight. Apply a generous amount of silicone seal to the holes where the wires come through. If you cut ½" off the applicator supplied with the silicone, you can force the silicone into the holes. AN AIRTIGHT SEAL IS ESSENTIAL.

7. Use silicone to install the foam pads over *the aperiodic pressure release holes. The foam is installed from the inside sealed up against the holes. Use only enough silicone to seal the foam over the holes (do NOT allow the silicone to block the holes).* The larger foam pads cover the two holes in each front corner. Apply a very thin bead of silicone to the foam pads' outer edge and press it into place, making sure no silicone covers the foam where it is exposed by the holes. THIS IS EXTREMELY IMPORTANT.

8. Fold the second supplied sheets of stuffing and insert them into the cabinet through the woofer cutout. Distribute them evenly in the upper half of the cabinet. Fold and cut the second piece of stuffing material as I described in step 4. Fold the 2′ piece and stuff it into the middle third of the enclosure, on top of the one you just put in. Fold the remaining 4′ piece and stuff it into the bottom third of the enclosure. If you follow these hints, you are assured of even stuffing distribution.

9. Apply the foam gasket tape around the woofer cutout. Discard the cardboard gasket that comes with the woofer; it is only for protection in shipping. Audio Concepts supplies an ample amount of foam tape. Cut a length about 2″ longer than you will need. Peel off the backing and apply the tape next to the cutout's edge. When you get within 2″ of your starting point, cut the tape 1/16″ longer than needed. The foam will compress to form a tight fit with the other end.

10. It is now time to hook up the woofer leads. Check the slip-on connectors for fit. They should fit snugly. If they don't, use pliers to gently squeeze them a little tighter. The use of contact-enhancers such as Tweak, Cramolin, or Aurium will ensure a perfect electrical contact, but are not essential.

Soldering is a good way to ensure perfect electrical contact, but unfortunately, several years ago about 80% of our warranty claims were due to careless soldering. Do it at your own risk; we will not replace drivers that have been soldered!

I recommend soldering the drivers. If you do so, don't apply any contact cleaners. First, check each voice coil's electrical continuity with a primitive analog VOM on the RX1 scale. Each voice coil should read around 3Ω. If you read between the two voice coils, on RX10,000 the resistance should be infinite. Now that you know your woofers aren't defective, you can proceed with the soldering. The cable Audio Concepts used is so stiff you may have difficulty removing the slip-on connectors and bending the wire around the woofer posts.

There's a simpler way. Slip the connectors over the woofer terminals as the manual indicates. Then carefully solder the connectors to the woofer terminals. While you're at it, you can solder the wire to the connectors (they're crimped rather than soldered). Do all of this at your own risk. As soon as you solder your woofers, your warranty is VOID. You may wish to complete the assembly without soldering. After you've listened to the Sub-1s for a couple of weeks, and you're convinced they're working properly, it will be easy to pull the woofers out to solder the connections.

11. You can now carefully lower the woofer, magnet first, into the enclosure. Align the woofer so the terminals are on the sides rather than on the top and bottom.

12. Install and tighten the screws evenly so the woofer fits flat into the enclosure. After you've finished tightening the woofer screws, check for air leaks. Take your primitive analog VOM and connect it to the input binding posts. When the meter is connected and disconnected, on RX1, the woofer cone will move slowly. The Sub-1 is a heavily damped system. If the cone movement is fast and "springy" you have an air leak. Check the holes where the wires come through to make sure they're sealed. If necessary, apply more silicone seal. The cone movement may also be springy if the aperiodic holes are covered with silicone seal. You must avoid this.

13. The jumpers installed on the terminal boards provide additional bass output. We recommend you remove the jumpers, store them in a safe place, and reinstall them only if you wish to have additional bass output. I recommend permanent removal of the

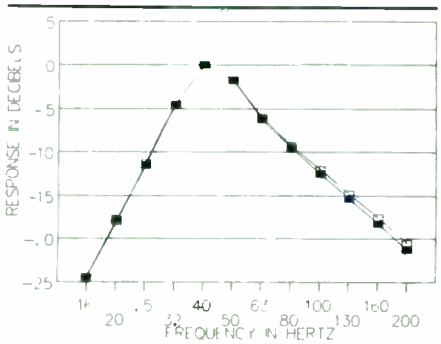


FIGURE 2: The Sub-1s' ⅓-octave, near-field response. The solid points show the first response, the open points the second.

jumpers. The system is simply too heavy in the upper bass region with the jumpers installed. Personal taste may play a role here, but the Sapphire II/Sub-1 system is more accurate with the jumpers removed.

At this point, the manual recommends breaking in the Sub-1s for 30–50 hours. I use a pink-noise generator for this purpose. Two days of continuous pink noise, at moderate level, will do the trick. If you can't stand the sound of pink noise for two days, lay the enclosures on their sides, with the woofers facing each other. Connect them to your amplifier out of phase. They will play much more quietly this way. If you own a CD test disc with a pink-noise band, you may be able to set your CD player to repeat the track continuously. Be careful with the level when running pink noise through the woofers. An eighth of an inch of visible movement is plenty to loosen the suspensions over a two-day period.

Measurements

The Sub-1 was interesting to measure. The design goal was to obtain flat low bass *in the listening room*, rather than in an anechoic or near-field situation. *Figure 2* shows the near-field response. I measured this using my Josephson Engineering Measurement Microphone and Old Colony's third-octave warble tone generator.

For this measurement, I turned the Sub-1s upside down and suspended the microphone to within ½″ of the woofer dust caps. The results aren't particularly useful. You certainly wouldn't buy a loudspeaker that reproduced only a few bass frequencies, rolling off everything on either side. But, this curve is instructive in terms of what the subwoofer is doing acoustically and electrically. The left-hand slope shows the system's natural low-frequency roll-off, and the right-hand slope shows the high-frequency roll-off synthesized by the crossover network. This measurement demonstrates that the high-frequency roll-off begins, electrically, around 50Hz. But, this is obviously not the way the Sub-1 was intended to be used.

For the Sub-1s to operate properly, the woofers must fire downward. This is an essential part of the Sub-1 design. *Figure 3* shows the combined response of the Sub-1s set up in my living room. I used the Audio Control Richter Scale Series III to make this measurement, since it is the best "in-room" low-frequency analysis system I own. The curve in *Fig. 3* was made with the satellites disconnected. It shows the Sub-1's response, with the microphone placed in my normal listening position. The performance is impressive to say the least. The Sub-1s are flat ±0.5dB from 22.5–90Hz, and the enclosure volume is less than 2 cubic feet.

Is this possible? Have the laws of physics been defied? Certainly not. Not even Mike Dzurko can change the laws of physics. Like most resourceful designers, he has *used* them to his advantage. In an enclosure of this size, a forward-firing 12″ woofer will never be flat to 22.5Hz. But, the downward firing woofer can take advantage of the rising low-end response of most listening rooms when woofers are placed extremely close to the floor.

Bear in mind that most of the woofer's radiating surface is 2–3″ off the floor. You can't achieve the same result by positioning a forward-firing woofer close to the floor, since only a small portion of the radiating surface is within a few inches of the floor. I can't accurately measure my room much below 22.5Hz, but the measurements I made indicated a −3dB point around 18Hz.

If you look at *Figs. 2* and *3*, you can see the bandpass characteristic isn't synthesized by the crossover alone. Firing the woofer downward flattens the response on the upper end as well. So, the final bandpass characteristic is synthesized by a combination of the crossover characteristic and the woofer/room interaction.

Figure 4 shows the Sub-1s' impedance curves. Over most of their operating range, the left and right samples were identical. F_C was actually a bit higher than Audio Concepts had specified. System resonance was 31Hz for both. This obviously did not prevent the Sub-1s from operating as specified, in terms of frequency response. The aperiodic loading and the heavy internal damping resulted in an extremely well-controlled impedance curve. This system's maximum impedance at resonance is only 11.3Ω.

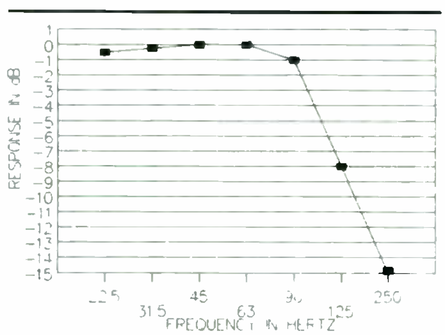


FIGURE 3: The Sub-1s' in-room response without the satellites. The performance is remarkable: 22.5–90Hz, ±0.5dB.

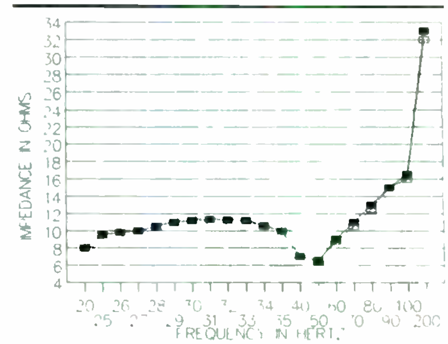


FIGURE 4: The Sub-1s' impedance curve. Aperiodic damping provides superb woofer control at system resonance. The solid points show the first response, the open points the second.

Connection Choices

My system currently consists of my "way beyond POOGE-4" Magnavox CDB-650, a highly modified Adcom GFP-565 preamp, and an Adcom GFA-585 power amplifier. I'm also using a pair of Adcom ACE-515 power-line filters, one on the power amp and the other on the low-level equipment. I was quite impressed by the GFA-585 prototype Victor Campos loaned me for the Sapphire II review.

I had to return the prototype to Adcom, but I purchased a pair of these amplifiers as soon as they were in production. Right now, I'm using only one of them, powering the Sapphire II/Sub-1 system with the complete, built-in passive crossover. I bought the second one knowing I would ultimately pursue system bi-amplification. I make my own interconnects, using Joseph Grado Signature Laboratory Standard Audio Cable, along with Neglex 7551 RCA connectors. My speaker cable is Livewire Indigo.

You must make a couple of choices when you connect the Sub-1s to your satellite loudspeakers. The simplest approach is to connect your power amplifier to the Sub-1 main inputs and connect the Sub-1 outputs to the satellite inputs. This arrangement doesn't take advantage of the Sapphire IIs' bi-wiring capability, but I ran the system this way for quite some time.

My listening partner, Lorelei Murdie, Concert Halls Manager at The Crane School of Music, offered a few second opinions in my review of the Sapphire II. She also logged a fair amount of time listening to the Sapphire II/Sub-1 combination. Although impressed with this system, she had one reservation: the Sapphire IIs sounded "stressed" in the high frequencies, as if the tweeter was "at its limit" at realistic volume levels. I didn't disagree with this, although I might express it a bit differently. To my ears, the Focal tweeter lacked some of the smooth-

Continued on page 61

Speaker Builder

THE LOUDSPEAKER JOURNAL

BACK ISSUES

1980 Speaker Cables: Measurable and Audible Differences • A Three-Enclosure System with Active Crossover and Delay (Parts 1-3) • Electrostatic Speaker Construction Project (Part 1-3) • Double Chamber Enclosure: Deep Bass from an 8" Woofer • Corner Mid-Range Horn • Thiele/Small and Vented Speaker Design (Part 1) •

1981 A Testing Unit for Speaker Parameters • Variable-Volume Enclosure • Thiele/Small Theory (in 3 parts) • Easy to Make Enclosure Using Concrete Blocks • The Tractrix Horn: Good Dispersion Bass From an Old Design • Diffuser Port for Small Boxes • Mini-Speaker Made From PVC Tubes • Closed vs. Vented Box Efficiency •Interview with P.G.A.H. Voigt • Dual 8" Symmetrical Air Friction Enclosure • Thiele/Small Calculator Computations • Thiele/Small Parameters for Passive Radiators •

1982 Transmission Line Theory • Thiele/Small Sixth-Order Alignments • The Quad 63 • Table Saw Basics • AR-1 Modifications • Active Crossovers and Phase • Three Transmission Line Speakers • A Beginner's First Speaker • How Passive Networks Interact with Drivers • Horn Loaded Heil • Phase Correcting Active Crossover • Wind Your Own Inductors • Series and Parallel Networks • High Performance Corner Speaker • Using Zobels to Compensate for Driver Characteristics •

1983 Building the Two-Way Dynaudio • A Crossover That Offsets Speaker Impedance • Using a Calculator for Box Design • Choosing a Calculator • A Simple Peak Power Indicator • A Small Horn Speaker • Audio Pulse Generator • How to Use Speaker Pads and Level Controls • An Easy-to-Build Voltmeter for Speaker Measuring • Nomograms for Easy Design Calculations • Interview with KEF's Raymond Cooke • Build a Simple Wattmeter • A New Type of Speaker Driver •

1984 Build an Aligned Satellite/Woofer System • BOXRESPONSE: A Program to Calculate Thiele/Small Parameters • Casting with Resins • A Phase Meter • An Interview with Ted Jordan • Building the Jordan-5 System • Self-Powered Peak Power Indicator • Closed Box Design Trade-offs • How to Build Ribbon Tweeters • Build a Dual Measurement Impedance Meter • A High-Power Satellite Speaker System • Build and Use a White/Pink Generator • Sound Pressure Level Nomographs •

1985 Compact Transmission Line Subwoofer • Bullock Crossovers, Passive and Active, three parts • Drive Attenuator, Computer-Designed • Curved Vertical Array • Sontek Powered Subwoofer • An Isobarik System •Modifying Strathearn's Ribbon • Ambience Systems • Experimental Transmission Line • Small Double-Chamber Reflex • Loose Walled Speaker • Modifying the Daline •

1986 The Edgar Midrange Horn • Sand-Filled Stands • Crossover Networks: Passive and Active • 5-Sided Boxes • A 2×4 Transmission Line • The Free-Volume Subwoofer • Notch Filters • By-Wiring the LS3/5A • A Push-Pull Constant Pressure System • Current and Power in Crossover Components • The Unbox (Egg) • Upgrade Speakerlab's S-6 Crossover • Measure Speakers with Step Response • A Gold Ribbon System • A Visit with Ken Kantor • A Tractrix Horn Design Program • Reviews: Audio Concepts "G"; Seven T.L. Midranges; Focal's Model 280; the Audio Source RTA-ONE.•

1987 A Compact TL Woofer • Frequency Response and Loudspeaker Modeling (three parts) • A Manual Coil Winder • The Model-One Speaker • Designing a Listening Room • A Six-Order Vented Woofer • Tapered Pipe Experiments • Visiting Boston Acoustics • A Vented Compound System • The Octaline • Spreadsheets for Speaker Design • In Memoriam: Richard Heyser (two parts) • Using Non-Optimum Vented Boxes • Building Speaker Stands • Evaluating Driver Impedance Compensation • Tuning Bass-Reflex • Six Woofers Compared • Bullock on Passive Crossovers: Alternate Bandpass Types • Fast, Easy Filter Calculations • A Mobile Speaker • Modifying the Polk 10 •

1988 Electronic Turns Counter • Two-Way Design • Minimus-7 Mod • Dome/Midrange/Tweeter Array• Plotting Complex Impedances • A Driver Design Primer • A Cabinet Primer • Tuning Up Old Systems • Low-Cost AR-3 Upgrade • Electronic Time Delay • Enclosure Shapes and Volumes • Minimum-Phase Crossovers • Spot Sound Absorbers • How to Add a Subwoofer • The Swan IV System • Sub-Bass Power Boosting • The Unline: A Short TL • Active Filter Computer Design Program • Low-Cost Two-Way Ribbon • Amp-Speaker Interface Tester and Construction Plans • The QB_3 Vented Box is Best • A Pentagonal Box System • Keith Johnson Profile • Sheathed Conductor ESL • A Symmetrically Loaded System: Part I • Ceramic Enclosure • Inductance Measuring Technique • Polk 10 Mods •

1989 A Middle/Side Speaker • A Satellite/Woofer System • A T/S Driver Program • All About Cordless Drills • Symmetrically Loaded System, Part 2 • Adjusting Auditorium Acoustics • The Audio Laboratory Speaker System • A Passively Assisted Woofer • Digital Filter Tutorial • The Listening Arc Alignment • Small IC Power Amp & Crossover • Easy Surround Sound • Building Speaker Spikes • An Isobarik in a Thunderbird • Sheetrock Cabinetry • A Picket Speaker • Servo-Controlling AR-1 • Silent, Safe Muting System • Equalizing the Klipsch Cornwall • A Test Switcher • Visiting the Klipsch Kingdom • Rehab for Kitchen Music • Spreadsheet Design • A Subwoofer/Satellite System • Impedance Measurement as a Tool • Practical Passive Radiators • A Symmetrical Dual Transmission Line Parts 1 & 2 • The Microline • A Voice Coil Wheatstone Bridge • Tweeter Q Problems • A Dipole Subwoofer for the Quad • Adjusting Woofer Properties • Modifying Paradigm's 7se •

1990 Acceleration Feedback System • Cylindrical Symmetric Guitar TLs • Compact Integrated Electrostatic TL (Parts 1-3) • Minimus-7 Super Mod • The Show (Bass Horn) • A Small Two-Way System • Helmholtz Spreadsheet • Heresy Upon a (Klipsch) Heresy • Beer Budget Window Rattler • Contact Basics • MDT Mini-Monitor Speaker System • Titanium + TPX + Polypropylene = Fidelity • Tom Holman, Skywalker, and THX (Parts 1, 2) • Bud Box Enclosure • Klipschorn Throat Riddle • Modular Three-Way Active Speaker • CD Speaker System • SPEAKER DESIGNER Software • Symmetrical Isobarik • Novice Crossvers • Triamplified Modular System • Magnetic Crosstalk in Passive Crossvers • Mitey Mike Loudspeaker Tester • Symmetrical Loading for Auto Subwoofers • Improved Vented Box with Low Q_{TS} Drivers • BOXMODEL Woofer System Design Software • Four Eight By Twos • Dynaco A-25 Mod • Klipschorn Throat Revisited •

Please send me the following back issues of *Speaker Builder*:

☐ **1990 $25**	☐ **1986 $18**	☐ **1982 $15**
☐ **1989 $23**	☐ **1985 $18**	☐ **1981 $15**
☐ **1988 $23**	☐ **1984 $15**	☐ **1980 $12**
☐ **1987 $18**	☐ **1983 $15**	

POSTAGE & HANDLING: $3 first two years, 50¢ each additional year
Canada add $4 per year
Overseas $4 per year surface mail, $8 per year air mail
Rates subject to change without notice
All remittances must be in US $ only

☐ MC ☐ VISA ☐ CK/MO

Total Magazines ________

Postage ________

Total enclosed $ ________

NAME ________ MAGIC NO ________

STREET & NO ________

CITY ________ STATE ________ ZIP ________

MC/VISA ________ EXP ________

Speaker Builder Magazine
PO Box 494, Dept. B91,
Peterborough, NH 03458-0494 (603) 924-9464 FAX: (603) 924-9467

Answering machine for credit card orders only–before 9:00 a.m., after 4:00 p.m. and weekends. Please have all information plus MC/VISA available

Continued from page 59

ness it had before the Sub-1 was added. It sounded a bit brighter and "rougher."

I decided on a different connection arrangement. The high portion of the Sub-1 crossover is not required to be in the tweeter signal path. It's needed only for the mid-bass driver. I reconnected the system as shown in *Fig. 5*. This arrangement has the power amplifier connected to the Sub-1 main inputs and the Sapphire II tweeter inputs. The satellite outputs on the Sub-1s feed only the mid-bass drivers on the Sapphire IIs. This arrangement takes advantage of the Sapphire IIs' bi-wiring capability.

Connecting the system this way improved performance significantly. First, Lorelei and I agreed the Focal tweeter's "stressed" nature disappeared. She now says the high-frequency balance is just right. The Sapphire II's high end sounds the way it did prior to connecting the subwoofers. The system also has greater inner detail and articulation from the upper midrange on up, the stereo image is more precise, and the soundstage is a bit larger. You may conclude that those electrolytic capacitors in the Sub-1 crossover were causing the problem. I believe this is part of the story; polypropylene bypassing will greatly minimize the effects of electrolytic capacitors, but it can't eliminate them completely.

There's more to it, however, than simply removing the electrolytics in the tweeter signal path. The Focal Kevlar tweeters are known to be difficult to control. To provide the best control, you must feed the tweeter and its crossover from a very low source impedance. Connecting them directly to my Adcom GFA-585's output does this nicely. The GFA-585 has a damping factor of over 600, from 20Hz to 20kHz, at 8Ω. This translates into a source impedance of less than 0.013Ω ($Z_{source} = Z_{load} / DF$). At 3kHz (the Sapphire II's crossover frequency), a 200μF capacitor's reactance is 0.265Ω ($X_C = 1 / 2\pi FC$). Adding the cap's reactance to the amplifier's source impedance reduces the damping factor to around 33 ($DF = Z_{load} / Z_{source}$; the Sapphire II impedance is 9.2Ω at 3kHz). This doesn't take into account a further reduction that the speaker cable causes.

Although the capacitor's reactance drops, the frequency increases, so does the Sapphire II impedance (*Fig. 1* in *SB* 6/90). At 10kHz, the Sapphire II impedance is 4.3Ω. The 200μF cap's reactance is 0.08Ω at this frequency and the damping factor is reduced to 86. Again, further degradation from the speaker cable's resistance hasn't been taken into account. From this, it should be obvious the connection in *Fig. 5* is best.

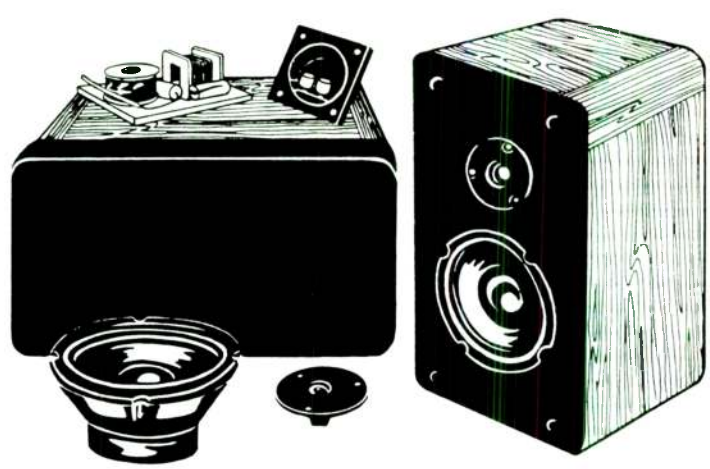
Listening

A subwoofer can't be evaluated by itself. Its performance can be judged only in the context of a complete loudspeaker system. The evaluation must also take place without any other loudspeakers in the room. So, with the help of Lorelei, her husband, and their pickup truck, we moved my 6' tall transmission line systems into storage. My wife was delighted. ("The Sub-1s look so much better than those 6' monsters. I hope you like the sound.")

I positioned the Sapphire IIs, on the AC Stands, in front of the subwoofers. The AC Stands aren't optional, in my opinion, if you are to achieve the real potential of this system. Placing the Sapphire IIs on top of the Sub-1s is the worst thing you could do. One advantage of having satellites and subwoofers in separate enclosures is the absence of low-frequency vibration coloring the midrange. If you set the satellites on top of the subwoofers, you'll defeat the purpose of having a system like this.

Adding the Sub-1s to the Sapphire IIs has produced a reference-quality dynamic loudspeaker. Mike's two years of work have really paid off. Integrating subwoofers with satellites can be difficult and is not always successful. Audio Concepts' system integration is absolutely superb. The Sub-1s blend seamlessly to the Sapphire IIs. No discontinuities exist in the tonal balance between the subwoofers and satellites and no hint exists that the bass

Continued on page 63



Continued from page 61

is coming from a different enclosure. The low end is clearly defined within the soundstage the Sapphire IIs reproduce.

Subjectively, the Sub-1s go lower than my large transmission lines. They rival my system in terms of bass detail and definition. The bass drum whacks on Robert Shaw's Telarc CD of Stravinsky's *Firebird* have tremendous weight, impact, and clarity. Even more impressive are the very soft bass drum beats, earlier in the piece. The soft, fluffy "woof" of the drum literally fills the room. The balance and extension of the low end is simply natural.

Compared to live, unamplified classical music (which Lorelei and I hear nearly every day), the low end is as tonally accurate as any I've heard. So many extremely large dynamic loudspeakers sound fat and bloated on the low end. I find the Duntech Sovereigns and many of the large Infinity systems are this way. In a large room at high volume levels, these large systems will certainly deliver more extreme low-frequency power than the Sub-1s. But, in a normal size listening room, the Sub-1s are more refined and more accurate.

The Sapphire II/Sub-1 system re-creates a large, three-dimensional, rectangular soundstage. One of my reference discs for soundstaging and imaging is Mercury's CD of music by the Second Vienna School, conducted by Antal Dorati. Dorati and the London Symphony gave the best performance ever captured in stereo of Schoenberg's *Five Pieces for Orchestra*. On the Audio Concepts system, this recording is reproduced with pin-point accuracy. Instruments literally dance around the soundstage, and the trombones are so far to the rear you feel transported to the hall where the recording was made.

The soundstage never collapses to mono at the rear. It maintains its rectangular character right to the back of the stage. RCA's CD of Fritz Reiner and the Chicago Symphony performing the Mussorgsky/Ravel *Pictures at an Exhibition* is a case in point. The *Gnomus* (track 2) contains colorful percussion writing, all placed clearly across the back row of the orchestra. You can hear the cracking of the whip deep in the far right corner of the stage. The bass drum literally bounces off the back wall.

From top to bottom, the Sapphire II/Sub-1 system is incredibly clean. In my Sapphire II review, I mentioned the midrange lost some detail in heavily scored passages. Adding the Sub-1 has removed the stress on the midbass driver, resulting in superb midrange clarity even in the most densely scored music. The performance of the Focal midbass driver used in the Sapphire II completely surprised me. I never expected a 7″ driver to be clean and detailed in the upper midrange. But Focal's Kevlar driver is a very unusual driver. With clean electronics and a clean power line, the midrange is free of hardness or breakup. (If you are unable to get consistently clean midrange from your system, I suggest power-line filtering. It made an amazing difference in my system.)

The system reveals an uncanny amount of harmonic detail from instruments. The string passages in the *Gnomus* are particularly impressive. Massed violins are smooth and detailed. Hall ambience is reproduced in abundance. Subtle ambience cues let you precisely determine the recording location size and the engineers' recording techniques.

Vocal music is extremely impressive on this system. Herbert von Karajan's London CD of Verdi's *Aida*, with Tebaldi, Bergonzi, and Simionato, is one of my favorite tests. The soloists sound warm and lifelike, free of any edge or harshness. The massed choral passages, such as *Sul del nilo* on track 5 of disc 1, are detailed and clean. Choral music is a real test of the midrange. Many otherwise fine loudspeakers add a hardness to massed choral music. This system doesn't. This recording is also an excellent demonstration of soundstaging and you can hear every soloist's exact location.

Charles Dutoit's London CD of Respighi's *Pines of Rome* is a difficult disc. The orchestration contains a tremendous amount of high-frequency energy, which the Decca/London engineers realistically capture. On many systems, this disc sounds edgy and harsh. On the Sapphire II/Sub-1 system, this recording sparkles, with superb inner detail and articulation. This is one recording that really shows the importance of connecting the system as shown in *Fig. 5*. The treble doesn't sound strained, overly bright, or edgy.

Overall, the Sapphire II/Sub-1 system reproduces music with a sense of effortlessness. Dynamics leap out at you without any sense of strain or compression. This is a loudspeaker system you can easily describe as "musical" and "accurate."

Comparisons

In *SB* 6/90, I compared the Sapphire IIs to Lorelei's Vandersteen 2Cis. I believed that, with a good subwoofer, this system would give them stiff competition. As it turns out, the Sapphire II/Sub-1 system isn't really competition for the Vandersteen's at all; it is vastly superior in virtually every area of performance. The system has a far better low end, much cleaner midrange and treble (without the Vandersteen harshness), and better soundstaging. What's amazing is you can purchase both units as kits for just under $1,500. This is only $200 more than the Vandersteen 2Cis, if you include the cost of their stands.

The Sapphire II/Sub-1 combination is a formidable competitor to far more expensive loudspeaker systems. Using my CDs, I have listened critically to the highly regarded B&W 801 Matrix Series II. The 801 Matrix has earned the reputation of being one of the best dynamic loudspeaker systems. It should be, considering its $5,900 price tag.

The first time I heard them, they were connected to Mark Levinson electronics and a Proceed CD player. This is, indeed, a superb loudspeaker. It is virtually state of the art in soundstage presentation. The B&Ws are very analytical. At the time I heard them, I thought they were a bit hard and bright in the treble in comparison to the Sapphire II/Sub-1 system, which to my ears is more musically natural. The sonic differences may be a matter of personal taste. If you consider the $4,400 price difference, you can appreciate the system's impressive achievement.

I wished to get Lorelei's opinion on the 801 Matrix, so I arranged a visit to another dealer and brought along a stack of CDs, including the ones mentioned in this review. This time, the B&Ws were driven by a pair of Linn LK-280 stereo power amps with the optional Sparks power supplies. The 801 Matrix allows bi-wiring or passive bi-amping with four channels of amplification. The total cost of the amplification was $5,980. The preamp was a Linn LK-1 MK III with the Dirak supply, costing $2,140. A Marantz CD-94 player and the CDA-94 outboard digital processor, totaling $3,600, completed the system.

We were both quite disappointed. Imaging was not as precise as the Audio Concepts system and the soundstage was lacking in depth. The B&Ws' bass wasn't as well defined, and the midrange and high end were not as clean as on the Audio Concepts. The B&W system failed to reveal differences in my recordings, as if they were done in the same hall by the same engineer. Dynamics were compressed. In particular, the quiet passages were too loud. The system didn't have power-line filtering, which could be part of the problem. To be fair, I haven't evaluated the B&W loudspeakers in my living room with my electronics. My first experience with the B&Ws was far more favorable than the last one. It is fair to say the Sapphire II/Sub-1 system can at least hold its own with this highly regarded system costing four times as much.

I haven't used the Vandersteen and B&W loudspeakers for comparison because of any prior dissatisfaction with them. On the contrary, these two loudspeakers have become standards by which other dynamic loudspeakers are judged, in their respective price categories. When a manufacturer sets a high standard, they invite comparisons. Such comparisons should be viewed as a compliment, even if something better finally comes along.

Conclusions

The Sub-1 subwoofer is an outstanding performer that integrates superbly with the Sapphire II satellites. Together, they form a highly refined full-range loudspeaker system rivaling the best dynamic

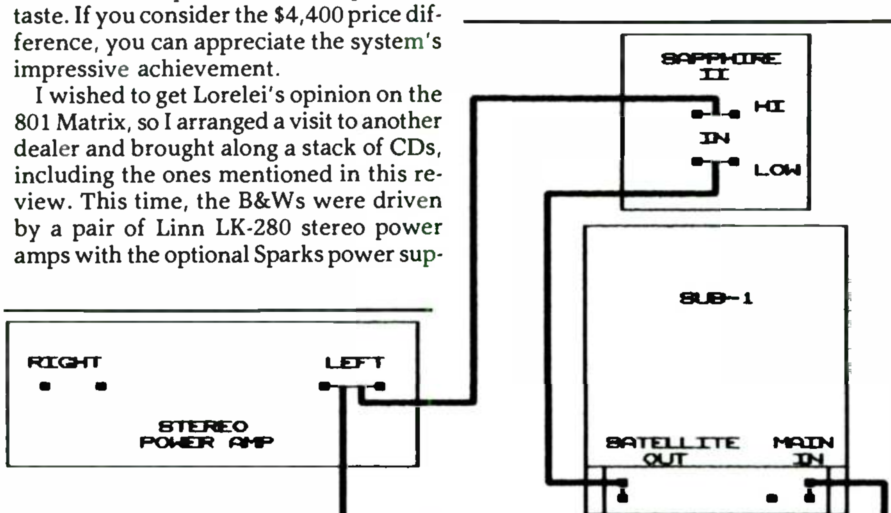


FIGURE 5: Recommended method for connecting the Sub-1s to the Sapphire II satellites. The left channel is shown. The right channel is identical.

systems. In fact, the Audio Concepts system has a transparency and effortlessness that will compete favorably with many dipole systems. At the price Audio Concepts is asking, these loudspeakers are a steal. Buying the complete system factory assembled adds only $160 to the total price, but you won't learn as much or have as much fun.

If you are skeptical and think these loudspeakers are "too cheap" to be as good as I've said, bear in mind something about Audio Concepts' pricing. They sell their systems factory direct to you. If they were to offer their systems to dealers, the dealers would pay roughly the same prices you and I pay. Then, the dealers would add the usual 80% to 100% markup, and that would be the list price of the loudspeakers. So, just tell your friends the "list price" is twice what you actually paid, and everyone will feel better.

If you intend to use the Sub-1s with satellites other than those made by Audio Concepts, the high section of the crossover may not be optimum. With some experimentation, you should be able to design a high-pass filter right for your satellites. You could mount your own filter on the back of your satellites or place it inside. If you do this, the Sub-1 outputs won't be used. Bi-amplification should be possible, but you should not defeat the low-pass filter in the Sub-1s. The low output on your electronic crossover should be a simple, flat unity-gain buffer. The high output should be a first-order filter with a −3dB point of 165Hz.

Audio Concepts is beginning to gain the recognition they deserve in the high-end audio community. The company is providing a great service by offering outstanding loudspeakers amateurs can build themselves. If you're not into building your own, they offer assembled systems at bargain prices, which compete favorably with some of the best high-end loudspeakers available.

I'm afraid my towering transmission lines won't see my living room again. The Sapphire II/Sub-1 system has become my new reference loudspeaker, and I plan to purchase the review samples. They have given me months of musical satisfaction, and I look forward to many more.

Mike Dzurko comments:

Gary's conclusions closely parallel those of other reviewers and owners of the systems. Having originally conceived of the concept more than 12 years ago with much of the last two years spent perfecting the design, it is indeed gratifying to receive such high praise.

Gary's review is so thorough and accurate that it should serve as an excellent guideline for potential Sapphire II/Sub 1 owners. One small correction: the introductory system price of a pair of Sapphire IIs, Sub 1s, and AC Stands is less than $1,400 and carries a money-back guarantee.

Technology Watch

90th AES CONVENTION

By Peter Muxlow

Several times each year, the Audio Engineering Society holds conventions in which exhibitors display their wares and audio engineers, acousticians, experts in psychoacoustics, mathematicians, and others meet, exchange ideas, and generally catch up on the latest research and technology. Also, papers are presented at the official forum. These are published directly from the author's manuscript without editing, corrections, or any formal consideration by the AES review board. This year, I attended the 90th convention in Paris, where several fascinating papers were presented.

Monitoring music during the recording process has always been a problem. Quality evaluation of the recording depends on the monitor speakers' accuracy and the control room's acoustic properties. To date, no standard control room is internationally recognized. Dr. A. Persterer of AKG proposed a solution to this problem: binaural simulation of an "ideal control room" for headphone reproduction.[1] Monitoring is done on headphones to eliminate the room acoustics and to simulate a constant ideal control environment.

Headphone listening, however, has two problems: in-head localization (which distorts the stereo perspective) and the outer ear transfer functions (which varies widely from person to person). The binaural signal overcame these problems by using complex digital audio processing.

Four presets related to four classes of head shapes are available. Also, you can measure your own outer ear transfer function and store the information on a ROM card the size of a credit card. AKG was displaying a market model of the processor with headphones, so it's not simply an interesting idea, it's a product.

Also related to headphone technology, V. Barteis of Sennheiser[2] described a headset with active noise reduction. In a noisy environment, you need to block out background noise in communications headsets. Conventional closed-type headphones and hearing protectors give a good range of attenuation of sounds outside the ear piece in the high-frequency range, but are not very effective below 500Hz.

Active or electronic cancellation of this low-frequency sound is the goal. The principle is simple, the implementation more difficult. A microphone inside the ear cup samples sound within the closed ear cup. This is fed back 180° out of phase into the transducer's earphone, creating a sound wave that cancels the offending noise. I believe active cancellation techniques will be commonplace soon, as much is being done in this area with at least one headset on the market.

The thorny subject of subjective evaluation of loudspeakers[3] was addressed by R. Jason, who gave a current overview and the experiences at National Public Radio USA. Subjective evaluations of preferred loudspeaker directivity[4] was tackled by researchers from Cannon and the Institute of Sound and Vibration Research. They concluded listeners had no clear consensus of preferences, although naive listeners showed a tendency for a more omni-directional response. Researchers from the Audio Research Group, University of Waterloo, investigated the effect of sound radiation by speaker cabinet walls.[5]

Of the two papers on Class D amplifiers,[6,7] I thought one had extremely promising prospects.[6] The aim is to convert oversampled digital signals directly to analog via pulse width modulation—Class D. This work has been continuing for decades and has been the subject of several previous preprints. As I listened to the paper being read, and the subsequent questions and answers afterward, I sensed the authors' excitement at the way it was progressing. They have a large team working on the project, including some of the circuitry fabricated as an application-specific integrated-circuit (ASIC) in a package requiring 6800 transistors. If it can be made to work well, and economically, the Class D amplifier may at last become a commercial reality.

A loudspeaker using the corona effect[8] by A. Deraedt of France looked interesting. The prototype demonstration was disappointing. It seemed to work only at a very low level with some hiss. Also, the problem of ozone production acknowledged in the paper had not, I think, been solved. Ozone generation is also a problem of the Magnat, the only commercial corona speaker in production of which I am aware. Incidentally, the Magnat was also invented by a French designer, S. Klein.

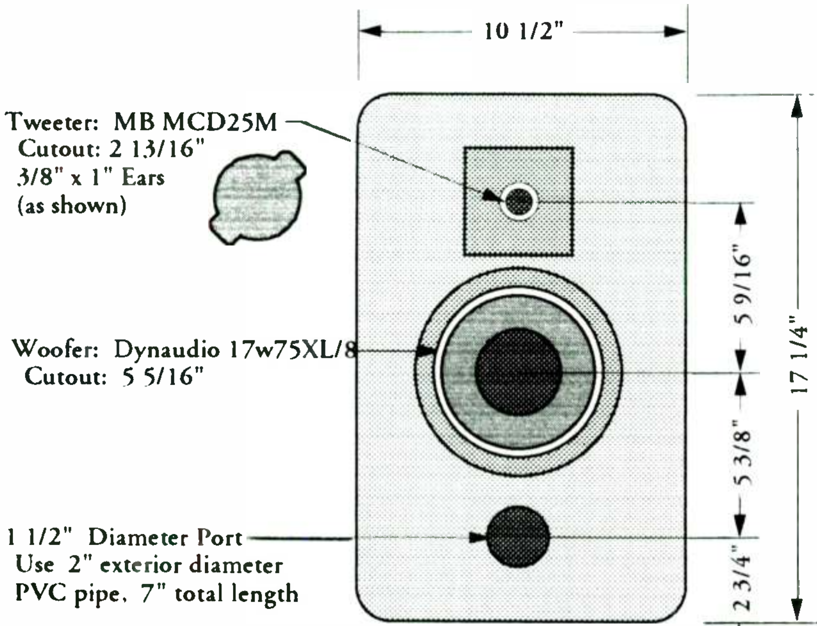
Computers are playing an increasingly important part in acoustics. EASE is a program generating a lot of interest particularly with acoustic consultants. You input the dimensions of your auditorium, hall, or other location and EASE simulates the ETC, the frequency response, and the reverberation time. Dr. W. Ahnert, the program originator, presented simulated responses of two theaters and one planetarium and compared them to the measured results.[9]

The range of topics of other papers included "Overheat Protection Circuits for Moving Coil Loudspeakers,"[10] "Electromagnetic Compatibility—The Art of Grounding,"[11] and "Optimization of the Nearfield Monitoring Environment."[12] As you can see from the few papers I have selected, the subjects were diverse, all with a common goal—better audio. It was an exhilarating experience.

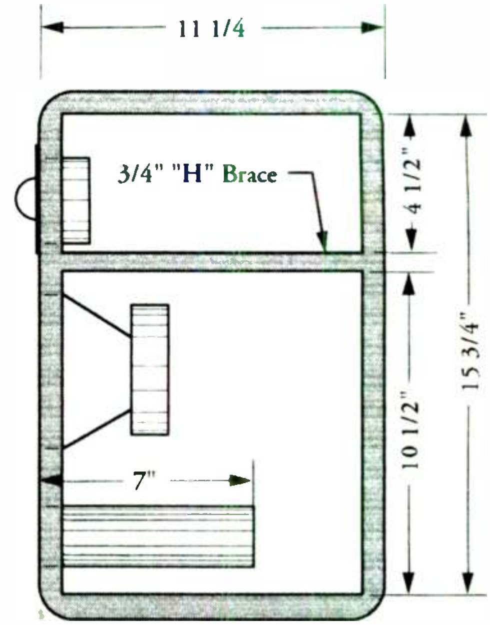
This is a limited coverage of some of the papers. A complete index of the preprints is published in the *JAES*, and you can obtain copies of the preprints for $4 each by writing to the Audio Engineering Society, 60 E. 42nd St., New York, NY 10165.

REFERENCES

1. "Binaural Simulation of an 'Ideal Control Room' for Headphone Reproduction." Preprint 3062 k-4.
2. "Headset with Active Noise Reduction for Mobile Applications." Preprint 3061 k-3.
3. "A Real-World Implementation of Current Theory in Loudspeaker Subjective Evaluation." Preprint 3048 h-5.
4. "Subjective Evaluations of Preferred Speaker Directivity." Preprint 3076 h-2.
5. "An Investigation of Sound Radiation by Loudspeaker Cabinets." Preprint 3074 f-4.
6. "Developments in Realizing an All Digital Power Amplifier." Preprint 3034 e-7.
7. "The Nonlinear Distortion in Class D Amplifiers." Preprint 3035 e-8.
8. "Electroacoustic Transducer Using Corona Effect." Preprint 3037 f-2.
9. "Comparison Between Simulated and Measured Energy Time Curves." Preprint 3028 d-7.
10. "Overheat Protection Circuits for Moving Coil Loudspeakers." Preprint 3036 f-1.
11. "Electromagnetic Compatibility 'The Art of Grounding.' " Preprint 3041n g-1.
12. "Optimization of the Nearfield Monitoring Environment." Preprint 3077 h-4.

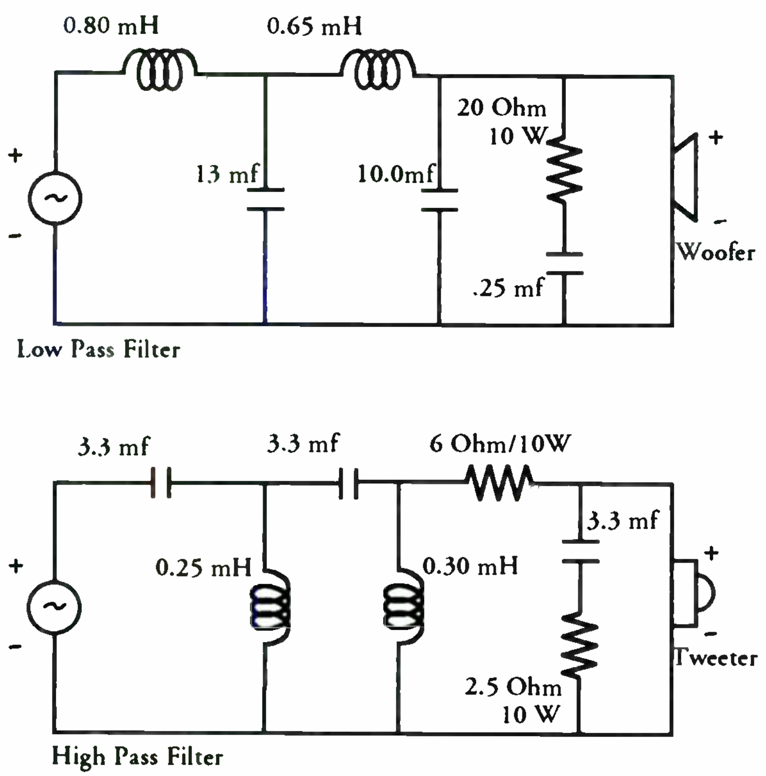


FRONT BAFFLE



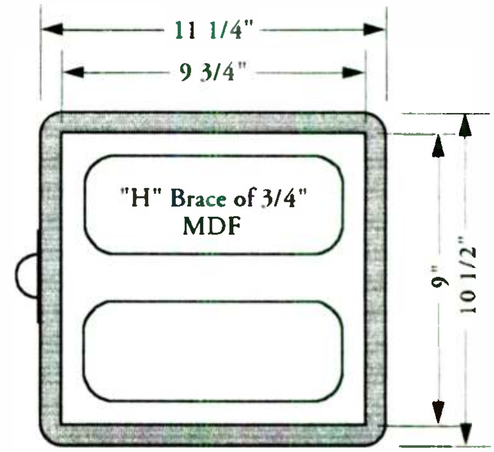
SIDE SECTION

Typical Cabinet Construction:
3/4" Medium Density Fibreboard (MDF)



Note: All Coils are 19 Gauge

CROSSOVER NETWORK



TOP SECTION

The Black Dahlia

Designed by Dick Olsher
Prices: Complete Kit, $500.00
Without Cabinet, $350.00

A & S SPEAKERS

3170 23rd Street; San Francisco, CA 94110
Telephone: 415-641-4573 Fax: 415-648-5306

Fast Reply #GF572

Tools, Tips & Techniques

A SIMPLE CROSSOVER NETWORK

Electronic crossover design is out of the question for most novice speaker builders. Eighteen decibel per octave zobelled networks probably come in a close second. But a simple passive network suggested by Ashley and Kaminsky in an AES article[1] offers simplicity, minimal phase shift, and surprisingly good sound at low cost.

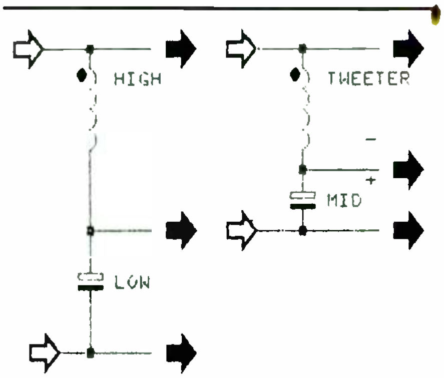


FIGURE 1: A two-piece two-way network (on the left) daisy-chained to form a three-way network (on the right).

The two-piece two-way network (*Fig. 1*) offers a 12dB/octave slope in the crossover region and can be daisy-chained to form three- and four-way dividing networks. It behaves symmetrically and has so few parts that experimental swapping will not break the bank.

I first tried the network as an upgrade to Radio Shack's Minimus two-way that always struck me as sounding dull in the midrange. The cost was incidental, the improvement dramatic.

TABLE 1

VALUES FOR NOMINAL 8Ω DRIVERS

CROSSOVER FREQUENCY	ZETA = 0.707 L (mH)	ZETA = 0.707 C (µF)	ZETA = 0.5 L (mH)	ZETA = 0.5 C (µF)
158.49	5.68	178	4.02	250
251.19	3.58	112	2.53	158
501.19	1.80	56	1.27	80
794.33	1.13	35	0.80	50
1,000.00	0.90	28	0.64	40
1,584.89	0.57	18	0.40	25
1,995.26	0.45	14	0.32	20
2,511.89	0.36	12	0.25	16
5,011.87	0.18	6	0.13	8
10,000.00	0.09	3	0.06	4

Table 1 gives you a starting point for experiments. Selecting the zeta = ½ option gives you the smoothest transition. This option allows you to put a bump in the midrange for a more forward sound. This bump can be quite pleasing with smooth drivers, but grating if matched with the wrong combination. The values do not seem critical and a little cut-and-paste will take off the rough edges.

A simple crossover like this would make an inexpensive choice for a satellite/subwoofer combination. It worked wonderfully on my main speakers that consist of a pair of large ported solid oak towers with Peerless 1599 woofers and 655 tweeters. Perhaps a pair of Peerless 1591 woofers and a 1730 tweeter in a D'Appolito configuration would make a $75 wonder.

Jerry Stump
Portales, NM 88130

REFERENCE

1. Ashley, J. Robert and Allan L. Kaminsky, "Active and Passive Filters as Loudspeaker Crossover Networks," 39th Convention of the Audio Engineering Society, New York, October 12, 1970.

CLOSET BOLTS

Over the years, I have seen advertisements for speaker spikes. Due to the expense of some spike sets, I have looked for alternatives.

Recently, at a local hardware store, I found a viable solution—"closet bolts," which have standard machine threads cut into one end and standard wood threads cut into the other (*Fig. 1*). How I used them as speaker spikes is quite simple and provides an effective speaker-to-floor coupling. I screwed the half with the wood threads into the bottom of the speaker and formed the machine thread half into the spike.

Select an appropriate size nut and washer to fit the machine thread half of the bolt. Then screw on the nut and make the point of the spike by carefully grinding the bolt on a grinding wheel or placing it in a vise and filing it to a point. If you do the latter, wrap the wood threads in a heavy cloth or place them between two pieces of soft wood to avoid damaging them. [*Two nuts on the machine-thread end will make vise clamping easier, safer, and more stable.—Ed.*]

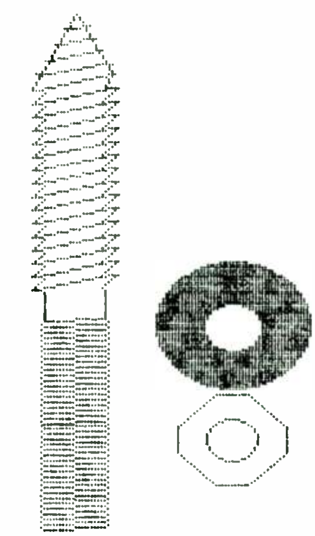


FIGURE 1: Closet bolt.

If the length you need is quite a bit smaller than the size of the bolt, first cut it to the desired length with a bolt cutter or hack saw. Next, paint the bolt if you wish, place the washer on the wood thread end of the bolt, and screw the newly machined spike with a wrench into the appropriate size hole drilled into the bottom of the speaker (*Fig. 2*). When the nut has reached the end of the thread on the bolt, the spike is screwed tight

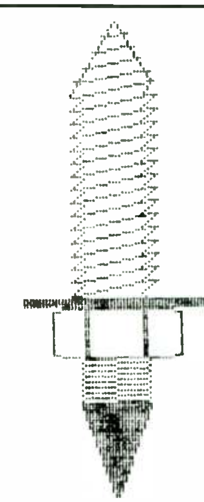


FIGURE 2: Machined spike.

enough into the speaker. Thus, you are forcing the nut (and ensuring it is tight and will not back out) onto the non-machined portion of the bolt.

These bolts come in several lengths and diameters suitable for different thicknesses of cabinet walls. As a final note, not every hardware store carries these bolts or often they are not called closet bolts.

Todd M. Kreuser
Green Bay, WI 54304

L-PAD IN A SPREADSHEET

To use a spreadsheet for Glenn Phillips' (*SB* 2/83, p. 20) and Kurt Klockeman's (*SB* 6/89, p. 56) L-Pad calculations, place the required load impedance in a cell you will use as a reference. I use row 1 for titles and notes, row 2 for impedance reference and column headings, and the remaining rows for results. In my spreadsheet (*Table 1*), cell A2 is the load impedance, column B is the value of the parallel resistor (R_P), and column C is the value of the series resistor (R_S). Column A is the amount of required attenuation for the circuit, in 0.2dB increments.

TABLE 1

SPREADSHEET CELLS AND FORMULAS

CELL	FORMULA/NOTES
A2	Load impedance (for example, 8)
A3	−0.1 (minimum attenuation)
A4	−0.2 Copy A4 down column A in −0.2 steps to the maximum required attenuation.
B2	R_P column heading
B3	$=((10^{A3/20})*\$A\$2)/(1-(10^{A3/20}))$
B4	Copy B3 down column B to match column A
C2	R_S column heading
C3	$=\$A\$2-(((1/B3)+(1/\$A\$2))^{-1})$
C4	Copy C3 down column C to match column A
To use Klockeman's formulas	
B3	=($A$2/(($A$2/D3)*($A$2 − D3)))*$A$2
C3	=$A$2 − D3
D3	$=(1/(10^{ABS(A3)/20}))*\$A\$2$

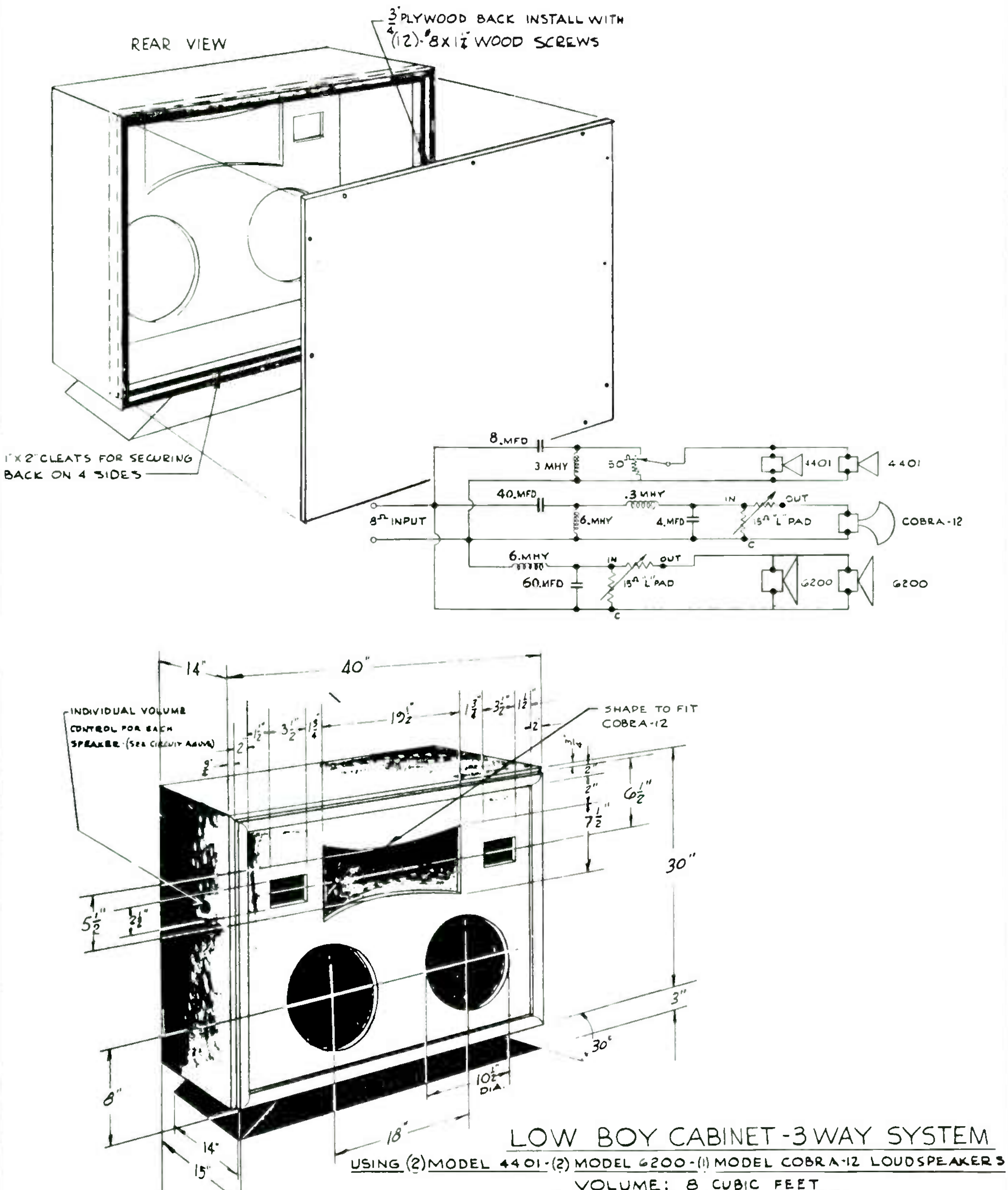
To change the load impedance value, altering the value in cell A2 will update the spreadsheet automatically. The dollar signs in front of the references to cell A2 are absolute reference symbols for my spreadsheet (Microsoft Works); you may need to use a different symbol.

Gregg S. Irwin
Seal Beach, CA 90740

Vintage Designs

The E/V Low Boy Cabinet

Courtesy of Bruce Edgar, Torrance, CA 90503



All a Speaker Builder Needs to Know About Dual Voice Coil Woofers

Model	81524DVC	10204DVC	10207DVC	1052DVC	12204DVC	1252DVC	15254DVC	15258DVC
Size Inches	8	10	10	10	12	12	15	15
Fs Hz	30	22.5	21.5	21	22	19	19	20
Mmd g	38	53	53	46	106	98	160	170
Cms	$.7\text{x}10^{-6}$	$.9\text{x}10^{-6}$	$.9\text{x}10^{-6}$	$1.3\text{x}10^{-6}$	$.48\text{x}10^{-6}$	$.75\text{x}10^{-6}$	$.48\text{x}10^{-6}$	$.41\text{x}10^{-6}$
Vas Liters	46	165	139	212	206	318	360	360
Rscc Ω per coil	3.6	3.5	5.8	5.7	3.5	5.7	3.5	5.7
Zmin Ω	4.3	3.5	7.1	6.5	3.36	7.0	3.8	6.3
Zmax Ω	100	46	92	86	24.9	69	43	58
VcL Mh @1Khz	1.0	1.2	.63	.7	.75	1.7	.72	1.2
Qms	9.2	3.35	3.7	4.11	2.12	4.4	4.79	4.69
Qes	.32	.27	.24	.29	.36	.39	.41	.53
Qts	.31	.25	.23	.27	.316	.36	.38	.47
Xmax mm peak	2.7	6	6	6	6	6	5.5	5.5
Efficiency db 1W/1M	90	90.4	90.4	91	90	90	90.4	90.4
Response	30-2.3K	30-1.5K	30-1.5K	25-2.5K	25-1.5K	20-1.8K	30-1.5K	30-1.5K
Power Watts	80 (40/40)	200 (100/100)	200 (100/100)	100 (50/50)	200 (100/100)	100 (50/50)	200 (100/100)	200 (100/100)
Magnet Wt. oz.	20	40	40	30	40	30	60	60
All units have Black Polypropylene Cones, foam surrounds, and voice coils made of high temperature aluminum.								
Voice Coil	1.5"	2"	2"	1.5"	2"	1.5"	2"	2"
Cutout & Depth	7.12" 3.37"	9.12" 4.45"	9.12" 4.45"	9.12" 4.45"	11.11" 5"	11.11" 5"	13.87" 6"	13.87" 6"
Price Each	$32	$47	$49	$40	$52	$40	$68	$68

Alignments	81524DVC			10204DVC			10207DVC			1052DVC		
Vb Liters	14	19	28.3	21	28	36	21	35	56	42	56	70
F3 Hz	51	43	39.5	54	48	44	49	40	34	42	38	35
Fb Hz	35	39	37.2	40	39	36	37	39	36	36	32	35
Peak At Res	+.1	0	-.5	+.8	+.4	0	+.2	.1	0	+.8	+.4	0
Port Dia. In.	2	2	2	2.5	2.5	2.5	3	3	3	3	3	3
Port Length In.	12.5	6.8	4.75	5.6	4	3.7	16.6	7.95	6.15	8.3	7.25	4.1

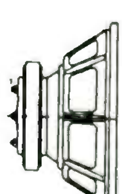
Alignments	12204DVC					1252DVC					15254DVC				15258DVC			
Vb Liters	28	42	42	70	92	30	56	70	120	170	56	70	100	100	56	70	70	100
F3 Hz	55	53	43	36	31	48.7	43.8	40.8	28.7	25.5	43	42	40	39	41	39	40	36
Fb Hz	No Fill		32	31	28	Fill	Fill	No Fill	26	25.5	Fill	Fill	Fill	No Fill	Fill	No Fill	Fill	Fill
Peak At Res	+.3	-.5	+1.0	+.2	0	+.8	0	-.3	+1.2	+.3	+.2	0	-.4	0	+1	+1.5	+.5	0
Port Dia. In.	Qtc. .83	Qtc. .71	3	3	3	Qtc .92	Qtc .73	Qtc .78	3	3	Qtc .8	Qtc .74	Qtc .66	Qtc .75	Qtc .96	Qtc 1.04	Qtc .89	Qtc .79
Port Length In.	Sealed		12.4	4.5	3.8	Sealed			4.5	3.8	Sealed				Sealed			

We thought that Speaker Builders would appreciate a chart of Madisound Dual Voice Coil Woofers with specifications and applications. There are several reasons why you may wish to choose a woofer with two voice coils:

1) You want to use a common subwoofer and drive it with a stereo amplifier.
2) You wish to parallel the voice coils at low frequencies to give added bass boost to your system.
3) You may use an impedance not commonly available. Two 4 ohm coils can be paralleled for a 2 ohm net impedance. Two 8 ohm coils can be used in series for a 16 ohm load.
4) Many Madisound dual voice coil woofers can be used in very small boxes. This is ideal for quality autosound bass and unobtrusive video systems.
5) Adding a subwoofer is not expensive; low frequency crossover filters are available starting at $50 a pair, (even less if you buy parts for home assembly). Cabinets are limited only to your imagination.
6) A subwoofer is the most dramatic improvement you can make to the average existing stereo system.

So as Arsenio would say, *"Let's get Busy!"*

Ordering Information All speaker orders will be shipped promptly, if possible by UPS. COD requires a 25% prepayment, and personal checks must clear before shipment. Adding 10% for shipping charges (Residents of Alaska, Canada and Hawaii, and those who require Blue Label air service, please add 25%). There is no fee for packaging or handling, and we will refund to the exact shipping charge. We accept Mastercharge or Visa on mail and phone orders.



Madisound Speaker Components
8608 University Green
Box 4283 Madison WI 53711
Phone: 608-831-3433
Fax: 608-831-3771

SB *Mailbox*

MITEY MIKE FIXES

Two equations in *Table 1* of my "Mitey Mike" article (*SB* 6/90, p. 10) are incorrect. "Max Undistorted SPL" should be >120dBA and "Wideband Noise Level" should be <42dBA.

Joe D'Appolito
Andover, MA 01810

VENTED-BOX ERROR

Figures 3 and *4* of my "New Guidelines for Vented-Box Construction" article (*SB* 2/91, p. 14) were interchanged.

G.L. Augspurger
Los Angeles, CA 90039

24W100 PROBLEMS

I am writing to comment on Stephen Wakeman's letter in *SB* 6/89 (p. 62) and Robert Bullock's response concerning the performance of the Dynaudio 24W100 woofer. I have had a negative experience evaluating two of these units.

I built a three-way system in a sealed box with a volume of approximately 50 liters. I am currently using the Dynaudio 21W54 woofer, D76 midrange dome, and 21AF tweeter. Obviously, I am not prejudiced against Dynaudio. At one point, I had narrowed the choice of woofer to the 24W100 or the 21W54 and had already tested the 21W54. The first 24W100 had what I consider major problems, so I sent the unit back to the dealer with a rather detailed explanation.

The dealer apparently did not test the driver, but instead sent me another of the same type. This unit had exactly the same characteristics, so it did not seem to be an isolated problem. I sent the second one back with a plea to replace it with a 21W54, which the dealer did. I never heard further from the dealer or anyone else on this matter.

The problems I encountered were threefold. First, at rather modest power levels (about 1-2W) at frequencies of 50Hz and below, the driver, when mounted in my sealed box, created an objectionable "wheezing." I think this is because the driver magnet is not vented to the rear, as it is in the 21W54. Rather, several small holes are found in the front of the cone approximately where the voice coil is attached.

I suspect the sound I heard was generated by the air being forced through these holes when the cone underwent the large motions necessary for good base response. The wheezing is probably due to turbulence this motion generates. The sound was inaudible when the speaker was driven at frequencies above 100Hz or at much lower power levels.

Second, I carefully measured the impedance curve of the drivers to establish the resonant frequency and Q in my box. The method I used is rather sophisticated, I think, because I measure the real (resistive) and imaginary (reactive) parts of the impedance as a function of frequency. From this data, you can easily obtain the resonant frequency and Q_M, where Q_M refers to the mechanical Q neglecting the effect of electromagnetic damping.

The Q_M was roughly consistent with that quoted by Dynaudio after including the Thiele/Small factor of square root of 1 + alpha. However, the calculated total system Q_{TS} was considerably larger. Specifically, from Dynaudio's claim of Q_{TS} = 0.3 in an infinite baffle, you can calculate a Q_{TS} of about 0.46 for a 50 liter box. My results indicated a Q_{TS} of about 0.8, which is much larger. Only with the 24W100 driver have I had this sort of discrepancy; I have measured three other woofers and always come fairly close to the expected value.

Furthermore, I believe Dynaudio's literature belies their claimed value of Q_{TS}. The point is that by examining the impedance curve, you can infer the value of Q_{TS} from the Q_M. The ratio of these two Q values is equal to the ratio of the net resistances for open and closed circuit connection. For example, the minimum resistance is about 6.5Ω, while the maximum according to Dynaudio's data sheet is about 13Ω. This implies that the electrical equivalent of the mechanical resistance is about 6.5Ω (13 − 6.5). Hence, if the Q_M = 1.6 as Dynaudio claims, the Q_{TS} should be about half that, or 0.8. It is impossible for the Q_{TS} to be as low as 0.31, as Dynaudio claims, without the impedance curve rising much higher (to about 40Ω). Something is inconsistent with their measurement techniques.

Third, the sound quality with modest power drive was rather bad at 50Hz. The harmonic distortion was very objectionable, aside from the question of the wheezing noise.

I would gladly relay this criticism to Dynaudio, except their address in Denmark seems to be a well-kept secret.

I close with two rather irrelevant remarks. First, I am a Ph.D physicist and am deeply involved in research into another kind of resonance—nuclear magnetic resonance. The point is I am very familiar with the physical and mathematical principles of resonance. Second, the systems I did build sound beautiful, and I am grateful to Dynaudio for the high quality of the driver I selected.

I would be interested to hear of the experiences of other speaker builders on this matter, and especially to learn the opinion of Mr. Bullock whose expertise is well-known.

Peter M. Joseph
Upper Darby, PA 19082

POWER AMP QUESTIONS

I am breadboarding a mono power amp onto the back of each bass box. This eliminates building an amplifier chassis. Each side will have its own amp and power supply. The power supply transformer and caps will reside inside the box. I increased the box depth by 1½" to allow for a recess in back and for the power supply's extra volume. *Figure 1* shows an idea of the physical layout. I have several questions, however.

1. I am concerned about 60Hz noise, since the input cables and power cord will most likely be near each other. Assuming I buffer the output of the pedal coupler, is regular coax cable OK? Would it be advantageous to gain up the signal as well to run a high-level signal through the input cables? Any other suggestions

Continued on page 74

NOW IN NORTH AMERICA
EUROPE'S LEADING MANUFACTURER
Low distortion and resistance Sinterbar Coils.
This style former can be used for Air Core Coils as well as with a Ferrite bar.
Bell Core Coils reduce strayfield and saturation.
Corobar Coils for High End Crossovers with High Power Capability and Low Distortion.
Ferrobar Coils for super low distortion.
Transformer Core Coils have super low resistance and with our unique core configuration, less distortion.
Metal-Polyester Foil Capacitors.
Metal-Polypropelene Foil Capacitors.
High Quality bipolar Electrolytic Capacitors.
Air Core Coils and Ferrite Core Coils.
OEM cross-overs to meet precise specifications with narrow tolerances.
Inter technik
P.O. Box 186
Kent, WA. 98035
Phone (206) 852-4049
Fax (206) 852-1203

Continued on page 72

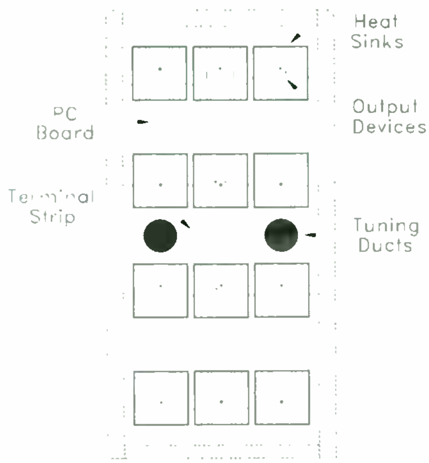


FIGURE 1: Rear view of the Swan IV bass speaker, showing the power amp.

or thoughts on this approach would be welcomed.

2. I am having trouble locating 2½″ ID conduit for the tuning ducts. Can you recommend either 2″ or 3″ diameter, and at what length?

3. Does anyone know how to produce a "black walnut" or "black ash" wood-grain finish? I am laminating with ash and wish to match the gloss black satellites.

Hubert Biagi
Tuscon, AZ 85719

SERVO PROS AND CONS

Speaker Builder has featured several theory and construction articles on servo-controlled speaker systems (Brown, 3/89 and 1/91; Mortensen, 1/90). While I am always willing to be shown a new or better way to do things, in my experience, servo-controlled woofers simply do not perform adequately. Perhaps these *SB* articles embody provisions to eliminate the serious deficiencies inherent in these systems; if so, I have missed them.

I am concerned that readers who lack experience in this area will spend considerable time, effort, and money constructing such systems, only to discover their deficiencies are significant. Any real-world design necessarily embodies compromises and tradeoffs, but servo-controlled bass speakers are inherently flawed and problematic—at least in terms of today's technology. My purpose in writing is to identify and define these problems and to question the wisdom of using servo control for most applications.

Let me describe these alleged deficiencies and why I believe servo control actually causes more problems than it solves. First, I think we can agree that an ideal woofer in an ideal enclosure does not need servo correction. Woofers and enclosures are not ideal, however, so all speakers embody compromises.

The servo system is usually employed in two circumstances: when the woofer system needs added low-frequency extension and when a reduction of harmonic distortion is desired. More often, it is the first case servo designs address.

In both cases, you can achieve the same benefit by using either more or better drivers or larger cabinets—in simplistic terms, "more speakers." Solving the problem this way is more than equivalent to using a servo, as linearity is naturally improved. But, this often leads to what some describe as an unacceptably large system.

In brief, the servo design is intended to provide the benefits of the "large" system in a small system size. I include those commercial 18″ servo-controlled subwoofers in this "small system" category (since their cabinets are undersized). This is the crux of the matter. Trying to get large system performance from a small system through the use of electronic correction by feedback is what is being attempted. Can you do this successfully?

Immediately, the problems leap out. The main issue is one of linearity, the driver's ability to move exactly according to the electrical signal input to it. Most drivers are nonlinear to some extent. They are typically fairly linear for most of their range, with deviations from linear behavior increasing dramatically at the extremes of motion. In practice, the more you try to make the driver move, the louder it is played, the higher the distortion, and the less the linearity.

This leads to the simple understanding that more drivers + lower level per driver = less distortion. Also, the relationship to mass for an infinitely stiff material says that less moving mass = lower distortion. Electrostatic speakers take advantage of both ideas. Given enough surface area, ESLs will produce as much bass as any other speaker design. Horn designs, like the Klipshorn, also take advantage of these principles to produce lower harmonic distortion.

In practice, the servo system is trying to trick the speaker into displaying the above two characteristics. Here is the catch. The speaker is being given electrical signals, presumably instantaneously and concurrently with the input, to "correct" its motion. This creates unavoidable problems.

Continued on page 76

VACUUM TUBE DEALER PRICE LIST (MAY 1991)

SOVTEK, USSR

EL84/6BQ5	$2.60 each	12 at $2.25 each	25 at $1.75 each	
6V6GT	3.90	3.35	2.95	
6L6GC	4.50	3.95	3.65	50 CALL FOR SPOT QUOTE
5881/6L6WGC	6.75	6.20	5.75	100 CALL FOR SPOT QUOTE

The 5881 is Russia's selected military 6L6WGC and regarded as the very best 6L6 type in the world. Eric Clapton recently replaced the Philips/Sylvania 6L6 STR's in his custom Soldano amp which he uses live, with SOVTEK 5881's because of their milky smooth sound. Discriminating musicians thruout the world are retubing with the SOVTEK 5881/6L6WGC.

SINO, CHINA

5U4G	$4.60 each	10 at $4.15 each	25 at $3.75 each	100 CALL FOR SPOT QUOTE
EL34	5.70	4.80	4.25	" " " " "
6L6GC	4.70	4.20	3.95	" " " " "
12AX7a	3.85	3.35	3.15	" " " " "
12AT7	3.90	3.40	3.30	" " " " "
6146B	12.50	11.50	10.50	" " " " "
6550	9.40	8.95	8.50	" " " " "
KT88	13.50	12.25	10.95	" " " " "

EI, YUGOSLAVIA

EL34/6CA7	$5.10 each	10 at $4.65 each	25 at $4.10 each	100 CALL FOR SPOT QUOTE
6DJ8	3.25	2.95	2.65	" " " " "
12AU7/ECC82	3.15	2.80	2.35	" " " " "
12AX7/ECC83	3.15	2.75	2.35	" " " " "

SIEMENS, GERMANY

EL34	$7.25 each	10 at $6.75 each	25 at $6.20 each	100 CALL FOR SPOT QUOTE

GE

5U4GB	$11.85 each	10 at $11.25 each	6550a	$17.95 each	10 at $16.50 each
6CA7	14.95	13.50	7025	14.75	13.25
6L6GC	14.95	13.50	7027a	16.95	16.25
6LF6	19.50	17.95	7591a	14.75	13.50
12BH7	9.95	9.50	7868	16.95	16.25
12AT7	6.85	6.40	8417	15.75	14.25
12AX7a	4.95	4.40			

SOLID STATE RECTIFIER

Built into tube socket. Direct plug-in replacement for all 5Y3, 5U4 and 5AR4 types. $6.25 each 10 at $5.90

ODD BALL TUBES (MOSTLY USA STOCK)

5AR4	$7.95 each	6BA6	$2.25 each	6J5	$4.50 each	6X5	$3.90 each	6189W (12AU7W industrial)	3.80 each
5BC3/9	1.95	6BK11	7.90	6J7	5.50	6X8	2.25		
5V4GT	5.50	6C4	2.50	6K11	6.90	12AY7	5.95	6973	11.90
5Y3GT	4.95	6C56T	2.75	6SC7	6.90	12AZ7	3.90	7189	7.90
6AL5	2.75	6C10	4.50	6SG7	3.50	12DT8	4.95	7199	17.80
6AN8	4.95	6CA4	3.50	6SJ7	4.45	12DW7	12.95	7247	11.95
6AQ5a	4.25	6CG7	6.90	6SK7	3.15	12FQ8	4.75	OA2	2.75
6AT6	3.25	6EU7	11.80	6SL7	4.35	57	1.50	OA3	1.95
6AU6	2.50	6FH8	10.90	6SN7	5.45	5749 (6BA6W industrial)	4.75	OB2	1.50
6AV6	2.50	6GT5	14.75	6U10	4.90			OC3	4.75
6AX5	4.90	6GW8	6.90	6X4	3.75	5879	$6.50	4CX350A	139.00

GIVE US A CALL ON ANY TOUGH-TO-GET TUBE. WE'LL FIND IT FOR YOU!

ALL TUBES GUARANTEED! QUANTITY DISCOUNTS AVAILABLE!

MINIMUM ORDER $50.00

ADD $5.00 SHIPPING ($10.00 OUTSIDE UNITED STATES)

SEND CHECK OR MONEY ORDER TO:

NEW SENSOR CORPORATION

245 EAST 63RD STREET, SUITE 526, NEW YORK, NY 10021

CONTACT: MIKE MATTHEWS TELEPHONE: (212) 980-6748 TOLL FREE: 1-800-633-5477 FAX: (212) 753-7340

Fast Reply #GF491

Continued from page 74

If this motion is large, two things happen. First, large levels of "correction" signal are applied to the amplifier and to the speaker's motor assembly. Correction applied to the latter is often in opposition to the input signal and to cone motion (out of phase). To "correct" this (the sum of the input + correction = motion − error), the correction signal must differ from the input signal.

Even at modest levels, the amp can be overdriven, creating additional distortion. Even more importantly, the speaker motor can become saturated and overheat. To avoid this, the designer must use a larger power amp and a driver with a larger, heavier motor assembly. In the 3/89 article, a *200–500W* amp is called for to drive a single 8′ speaker. Or, you can choose to live with reduced output and dynamic range.

Second, as the output level from the system goes up, so does the percentage of servo feedback. In other words, the louder the system is played, the more the servo is required to work. This is fine, until the mechanical limits of driver suspension are approached. In practice, you would be shocked at how often a typical system is driven into this limit. Usually, our ears forgive much of it (no, not all, which is one reason you pay more for better speakers), since it is heard as a "soft" limiting, and is not audible as a discrete sonic event.

This second problem is often the more serious. The servo loop is given the "electronic program" to keep the system linear at all costs. As the system is driven harder, the loop tries more and more to keep it linear. It does this by creating a signal that is the inverse of the nonlinear motion. In effect, this electrical signal literally tries to stop whatever the cone is doing at that time. No amount of servo correction can fix input signals instructing the speaker to exceed its nonservo output range. Practically speaking, servo correction under these conditions amounts to amplitude compression. The Philips Little David system is an excellent example of the compression problem. I have found this compression effect to be audible in all servo-controlled systems I have heard.

The idea of a real system trying to get extended frequency response through the application of a servo loop is terrifying to me. Why? Well, if you look at the system to which a servo is to be applied, before the servo is added, and plot the low-frequency excursion limitations of the driver in the cabinet, you will find that usually there isn't much excess diaphragm option available, especially below $f_s/f_B/f_3$. If there is sufficient excursion, then either f_B is very high (no bass extension), the driver is from an alien civilization, or the extension is not needed. Practically speaking, this excursion limitation is clearly understood by the servo sensor. It sees the limitation, with the attendant onset of nonlinearity, and tries mightily to correct it—once again, with diminishing returns. The driver, alas, cannot exceed its own excursion limits.

Why not simply add an equalizer to obtain the extension or shaping of response you desire? Obviously, this will not work in many cases, since the system alignment is so far from optimal that the distortion produced by such an attempt is unacceptable. Certainly, without some control, the cone could be blasted out from the front of the cabinet.

Advocates of the servo system might suggest it is best not to operate a system near its limits; that is, if you operate your system within a "window" of loudness, most objections are eliminated. This is a good idea, to some extent. However, if you are going to build a servo system that will operate with sufficient output, keeping the speaker cone operating below a certain electrical/mechanical threshold, then haven't you actually built a larger, more linear, standard speaker system? If this is the case, why bother with the added expense and difficulty of a servo?

In practice, more often than not servo-based systems are not designed to improve

Continued on page 78

All Capacitors Are NOT Created Equal

In 1980, Richard Marsh co-authored the groundbreaking article on capacitors for Audio Magazine. His point: Caps, like all audio components, have different sonic characteristics—even film and foil caps differ. Their quality depends on materials, construction techniques, and design concepts.

In the wake of this article, a number of new brands entered the market and claimed sonic improvements—though most simply used different lead materials (which get cut off and thrown out anyway). Not one offered anything new in basic design. Not one solved the sonic problems Marsh described.

Richard Marsh wasn't satisfied. So he put his years of research experience at a National Laboratory into his own creation—the MIT MultiCap™:

* A superior multi-sectioned capacitor that **is new in concept.** So new, Marsh has a patent on its design.
* A superior capacitor that **does improve sound**. Audibly. Listeners report that the MultiCap provides extraordinary transparency and neutrality, from the most delicate musical note to the most vigorous and demanding full ensemble tuttis. The closest to no capacitor at all.

"We search the world over for the very finest components for our designs. The MultiCap is—unconditionally—the best capacitor going—the next-best doesn't even come close. . .

—David Manley, Designer, VTL

"We spent countless hours auditioning the MultiCaps against all the alternatives, regardless of price, in our analog and digital chains . . . the improvements leap right out at you —undeniably."

—Ed Wong, Chief Engineer
Jackson Browne Studios
Santa Monica, California

"In my system, the MultiCap brought a greater improvement than upgrading my cables. Once you try them in your equipment, you just won't want to take them out."

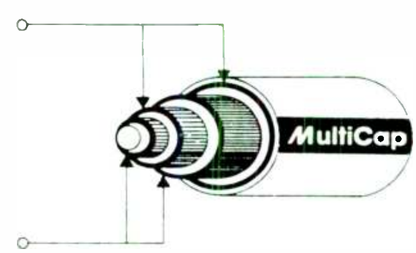
—Leland Pratt,
Ancaster Audio, Canada

Measurements also reveal the MultiCap's superiority, showing:

* the lowest ESR
* the lowest phase deviations
* the most stable phase characteristics
* the greatest transient performance

What makes the MultiCap so much better than other capacitors?

* Not only the selection of the best polystyrene and polypropylene films.
* And not only the selection of the best foils available.
* But in each MultiCap, you get ten precision, paralleled capacitors wound into one compact, optimized unit—that is, the MultiCaps are internally bypassed.



This cross-sectional drawing shows the coaxial construction of a Multicap. For simplicity's sake, only two of the ten sections are included.

And thanks to proprietary top-quality construction techniques, the MultiCap will not degrade over time, as so many others have done.

MIT now offers the **new metallized MultiCap**, using the same patented design, for low-current usage. Smaller in size, lower in cost—same great clarity of sound.

Call or fax for values, prices, and dimensions. And for white papers by Richard Marsh on capacitor design.

Music Interface Technologies, 3037 Grass Valley Highway, Suite 8212, Auburn, CA 95603.
Phone: 916.823.1186. Fax: 916.823.0180

MIT MultiCap™: "The most advanced capacitor in the world."

Fast Reply #GF703

Continued from page 76

already high-output/wide-bandwidth speakers. Rather, most servos are still set up to try and get something for nothing—more output, more linearity, and deeper bass than a given driver/cabinet can actually produce. You cannot expect modern digital dynamic range sources to stay within relatively narrow (dynamic level) windows. It seems to me a better idea to spend your time and money on better drivers and more linear cabinet loading systems, with a "rumble" filter to eliminate excess subsonic excursions and limit the driver's thermal load, if required.

In the end, the age-old tradeoffs of bandwidth for output and loudness for distortion remain immutable. Also, you must weigh the added costs of servo electronics and mechanics against the costs of other possible solutions.

Randall Bradley
Hannacroix, NY 12087

Arthur Brown replies:

I will attempt to give the basis for my enthusiasm for servo-operated woofers and relate my comments to your letter where I can.

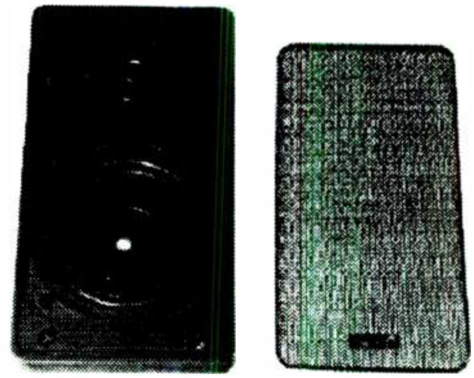
I agree that no woofer/enclosure is ideal. Any real-world design, whether an open loop or servo speaker, embodies compromises and tradeoffs. Harmonic distortion reduction is the servo speaker's primary benefit, with extended frequency range a desirable extra.

One point you discussed was trying to get something with "more speaker and more cabinet." If this *more speaker* gives you a wider operating limit, then it is a good solution. The servo speaker does not extend the fundamental speaker limits. A servo design must also take advantage of improvements in speaker operating limits. I am referring to the maximum cone travel, maximum input power, and cone area (sound pressure). It is necessary, therefore, to identify the benefits of servo speakers over open-loop ones.

If you must operate with a given set of speaker limits, improved or not, *more* box may help the open-loop speaker. Generally, the speaker operating limits dictate the amount of improvement possible. In this case, a servo speaker can also extend the system's performance in much the same way, but without *more* box volume. In fact, it is often desirable to have a small(er) enclosure in a servo speaker than in an open-loop one to help establish the servo open-loop gain/phase characteristics. I

Continued on page 80

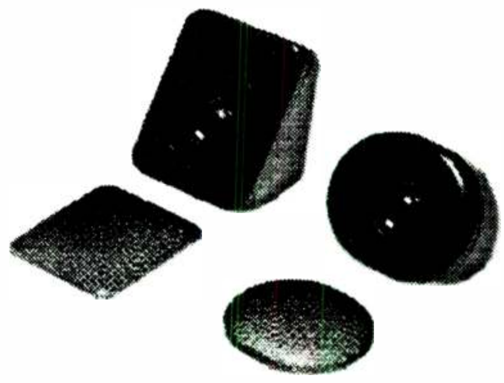
Sledgehammer Excellence in Auto Sound



The **SLEDGEHAMMER PLATE** system answers many of the needs of high quality autosound installation. It can be surface mounted with a below surface opening of only 3 inches in diameter and 1 inch depth. It gives you the quality of a one inch titanium dome tweeter and a polyester film filter capacitor. It can be used as low as 100 hz. or as a full range system with the provision to add a subwoofer at a later date. The construction is an all metal cast structure found in only the most expensive systems. This system can be installed on the rear deck or door panel in a fraction of the time it takes to mount flush mount systems. Dimensions:8.25" x 4.5" x 1.375"

The system is sold complete, and when you add up the savings in time and extras, you will see that the **SLEDGEHAMMER PLATE** system is an extremely good value.

Introductory Special Price:$79 per pair.



The **SLEDGEHAMMER TWEETER MODULES** allow you to put quality high frequency reproduction exactly where it is needed. You can now surface mount tweeters on the dash or doors, headliner, rear deck, or literally any upholstery panel. The units are self contained and include grills and a quality film filter capacitor. Response is from 4.5 to 22K Hertz. The dome is a high fidelity polyamid construction for reliability and smooth response. There are two models available: one with a rounded shape, and the other is a wedge, so you can match the units to the cosmetics of the auto interior. Installation is easy, and it does not require cutting the interior panels. You can take these tweeters for use in your next vehicle.

Because you can buy these units without all of the customary steps in distribution, you will find the cost a very pleasant surprise.

Introductory Special Price: **Model 23 Square $25 per pair.**
Model 25 Round $25 per pair

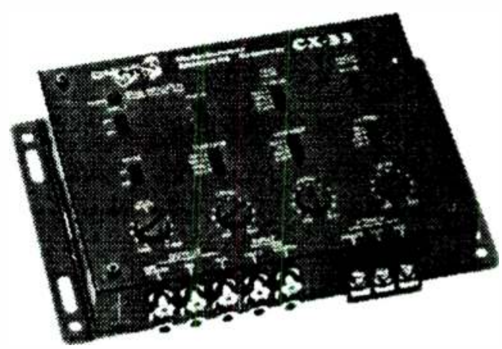
In the past Madisound has offered an autosound electronic filter called the MGT 33, but the company that assembled them for us went out of business. Since we were, in effect, back to square one, we decided that we would completely redesign the filter and develop a product that would be the best in the field.

As a result we are proud to introduce the **SLEDGEHAMMER CX-33** Electronic crossover with the following features:

- ★ Two or Three way operation.
- ★ Bass boost at 40 Hz variable +/- 12 Db.
- ★ Separate phase switch for both low and high bands.
- ★ Gold plated RCA inputs and outputs.
- ★ Level adjustments for each amplifier
- ★ Switched bipolar power supply
- ★ Crossover slopes 12 db Butterworth

Specifications include:

- ★ Frequency response 10 Hz-50K Hz +/- 3 Db
- ★ Total Harmonic Distortion below .01% @ 1 K Hz.
- ★ Crossover frequency: Woofer 50/80/120/180 Hz
- ★ Midbass Low: Flat/80/120/180 Hz
- ★ Midbass High: 2.4K/3.5K/5K/Flat Hz
- ★ Tweeter: 2.4K/3.5K/5K/7K Hz



The SLEDGEHAMMER CX-33 comes with individual packaging and a comprehensive owner's manual.

For a limited time, Madisound is offering the SLEDGEHAMMER CX-33 at a special introductory price of $74.

Madisound Speaker Components
8608 University Green
Box 4283 Madison WI 53711
Phone: 608-831-3433
Fax: 608-831-3771

Ordering Information: All speaker orders will be shipped promptly, if possible by UPS. COD requires a 25% prepayment, and personal checks must clear before shipment. Adding 10% for shipping charges facilitates shipping procedure (Residents of Alaska, Canada and Hawaii, and those who require Blue Label air service, please add 25%). There is no fee for packaging or handling, and we will refund to the exact shipping charge. We accept Mastercharge or Visa on mail and phone orders.

Fast Reply #GF1059

Continued from page 78

agree we are trying to get large speaker performance in a smaller package with the servo speaker. In addition to the smaller enclosure for the given speaker driver, we also reduce the system's distortion with the servo speaker.

I will describe a significant deficiency I see in the open-loop speakers to which I have listened. The characteristic can be described as the boombox response, the exaggerated low-frequency response of speakers designed for popular music. Too many open-loop hi-fi speakers exhibit this low-frequency behavior, generally to a much lesser extent than the boom box.

Another way of looking at this characteristic is based on the fact that the speaker has mass. Once excited, the cone continues to move until an opposing force is applied. Most open-loop speakers are designed to have less than critical damping (Qs of more than 1) to obtain the maximum flat or slight emphasis in the low-frequency range.

The speaker, therefore, is under-damped and behaves according to the laws of an under-damped oscillating spring-mass system. Uncontrolled motion is the result. This behavior results in a speaker being heard after the input signal is no longer present and in some cases when no intentional signal has been supplied.

This rise in response *can* be included in a servo speaker also. A flat response, though, can easily be provided in a servo speaker while truly extending its response range. Have you seen a speaker system flat within ± 1dB from 20–110Hz? Low-mass speakers can reduce this tendency to oscillate to a large extent. As you pointed out, relative to distortion correction, ESL speakers and multiple speaker arrays are good candidates. A practical compromise in these designs is the limitation of low-frequency response. It is still difficult to get enough air movement with them.

Horn speakers are a different breed of animal. They can move the air, but need very large cabinets to accomplish the low-frequency response. When these approaches are used, the system cost (space, as well as dollars) is excessive. These speaker systems are usually very large. Separate high-power driver amplifiers are almost always recommended with these high-end speaker systems. The servo needs a separate amplifier also, but the result is a smaller cabinet and low distortion.

The servo speakers with which I have experience are absolutely quiet when no sound input is present. People who have listened to my servo speakers initially think the speaker has no bass response until they hear a low-frequency passage on the recording. They are amazed and convinced the speaker is reproducing the lowest frequencies. I have some records I use to demonstrate the speaker performance; Telarc's thunderstorms and soft jazz records are typical. I would appreciate hearing more of your servo speaker experience.

I agree that you cannot operate any speaker driver outside of its operating region without generating distortion. It was my primary consideration to know my drivers' operating limits. I demonstrated this in my concern with knowing the driver linearity as shown in *Fig. 6* of the 1/91 article. I have run the same data on the AR-1 system to establish its limits.

I also got into the problem of low-frequency travel limits. The sidebar "Travel Limits of Drivers" (*SB* 1/91, p. 27) investigated the problem and showed the speaker's *limits* in terms of drive power. As you know, the travel limit requires reduced power as the frequency decreases. This is a requirement of any speaker, whether servo or open loop. I attempted to build a circuit that automatically limits the drive power to the curve of *Fig. B*. I have not yet found a satisfactory answer, free from hum and noise. I therefore use an LED VU meter to monitor the instantaneous power going to the speaker. This allows me to avoid excessive sound levels.

I calibrated the VU meter so the maximum power segment just lights at 50W RMS. I have yet to see a speaker input power that exceeds −9dB of the maximum segment, or 6.25W in my listening situation. I can and have turned the volume up and approached 50W on occasion. This is not my listening practice since that level is too loud for my 16′ by 19′ listening room. Those who will exceed the level I find satisfactory in my listening room should take into account their listening habits when specifying a low-frequency speaker system. See my discussion below.

I also monitor the frequency of the signal going to the overall speaker system using a segment level device associated with various equalizers. Mine is a BSR. It shows the decibel level between 32Hz and 1.6kHz for each octave. It also shows the total level.

Although this device is not complex, it does indicate the frequency range of the sound going to the speakers and gives me further confidence I am not exceeding the speaker limits. I recall listening to one Christmas record. Everyone in the room wondered aloud what they *felt*. On investigating, we decided it was probably the signal of a 16′ organ pipe. It was felt rather than heard due to the pressure the subwoofer generated. The power meter indicated a level of less than 6.25W at the time.

You speak of large levels of *correction* when using a servo speaker. These potentially large levels occur only when the speaker motor is not generating a feedback signal consistent with the input signal. A music transient into the system normally creates this type of condition. The transient's high frequencies, however, do not go to the low-frequency speaker due to the crossover. The low-frequency speakers, servo or otherwise, do not see severe transient signals.

Observe on an oscilloscope a square wave signal at the amplifier's speaker connections and at the low-frequency speaker driver after the crossover to understand what I am saying. The square wave signal will become almost a sine wave. The subwoofer hardly sees a pulse. The servo is easily fast enough to keep the speaker in phase with this input signal and the speaker drive signal (correction or error) will be moderate.

This error signal is controlled even when the speaker cone tries to oscillate as a spring-mass system. The feedback signal causes a correction that opposes that part of the movement not in the input signal. This correction is always within a defined phase/gain relationship (*SB* 1/91, steps 5 and 6, p. 20 and *Fig. 16*, p. 28) in a correctly designed servo. This servo action provides dynamic damping of the oscillation to make the speaker follow the input signal closely. The damping is automatically adjusted to match the situation—large sometimes, small others.

An open-loop speaker, however, is designed with a given Q for the system. The speaker/enclosure alignment and the amplifier damping determine the speaker damping (1/Q). Most open-loop speakers, as mentioned earlier, are designed to have less than critical damping (Qs of slightly more than 1) to obtain the maximum flat response. The speaker damping is therefore fixed and must handle all conditions as they come. The servo speaker damping adjusts to the conditions at the time.

I believe that any proposed system specification must include the low-frequency power (displacement) versus frequency I described in *Fig. B*. To establish the power level a system requires at low frequencies, it is necessary to have a clear idea of the type of music to be played, its frequency range, and the sound energy at these various low frequencies. If this information is unknown, you are choosing *any* speaker (not just servo speakers) with too little information. I estimated power level requirements briefly in Step 1 of designing a servo speaker (1/91, p. 20).

Let's explore these power levels further. I found the statement that peak-to-average signal power in recorded music is 10dB in some publication, which one escapes me now. The other variation in power requirement comes from the speakers' inherent requirements. They fall off in their output at low frequencies. If the response is to be made flat to some low *frequency*, it is necessary to add drive power, either by shaping the amplifier response curve or by servo techniques. Refer to my open-loop data on the AR-1 (*Fig. 16*). The variation in the gain between the maximum value and the 20Hz gain is about 11dB. This falloff is typical of open-loop speakers at low frequencies also. Therefore, you must add 11dB or 3.6V gain to the speaker drive. By design, the servo does this.

Let's add up these numbers. You might start by specifying an average sound pressure level in the room of 110dB. If you use an average speaker sensitivity of 86dB at 1M for 1W input, you need a 24dB increase in sound pressure at 20Hz (24 − 11 or 13dB at 100Hz). Each increase in sound pressure of 3dB requires doubling the input power. Thus, 86 + 3, or 89dB, requires 2.8284W × 2 = 5.6658W. Applying this relationship, you get a speaker input power of 724W for 110dB at 20Hz. This is the average or RMS power requirement for 110dB sound pressure at 1 meter at 20Hz. Generating 110dB at higher frequencies requires less input power. That sound pressure level is excessive and causes deafness. Never use it in any reasonable music listening room without ear protection.

Applying the 10dB peak-to-average rule above, you have the 110dB only on peak levels of sound and an average of 100dB. This requires about 90W average input power at 20Hz. The 724W still must handle the peak signals at 20Hz if you can provide the drivers with sufficient cone travel. I doubt, however, that peak sound pressures would occur at 20Hz in most music. Not many instruments can generate that kind of power at 20Hz. I also doubt that many people would listen to this 100/110dB level for more than a test minute.

Thus, we have the problem of determining the system specs. You might say you need these numbers to be sure the system distortion is acceptable at the listening level. In which case, you should define acceptable distortion levels in the specs. Do not expect to actually attain these power levels if distortion is your rationale; you have unrealistic specifications. You must evaluate the signal input behavior at all frequencies before you can specify a speaker system fully. My experience with my LED VU meter and frequency indicator has convinced me I can live with my system as discussed above.

As listening room requirements call for more sound power, several options are available. You can use two servo subwoofers rather than one. You can use larger diameter cone speakers or longer cone travel (larger V_{AS}). You can also probably use multiple speakers in a servo system, although I have not devised such a system. Multiple speakers in an open-loop configuration will increase the effective radiating area. The distortion of multiple speakers can be low if each speaker is limited to a low enough level input.

I continue to believe that a servo speaker is not an excessive complication considering the results. I do not recommend just anyone trying to design and build one since special knowledge and instrumentation are necessary. I am continuing to look for new drivers with extended limits. I would like a larger speaker and longer travel. I wish to build a future system design that can stroke linearly (not exceed its travel limits) with 100W input at 20Hz. I still need a relatively small box to provide the servo loop characteristics necessary for the servo. I have sent inquiries to several manufacturers for information on their products.

I hope I have responded to your concerns and I thank you for your interest in servo speakers.

Hans Mortensen replies:

Thank you for your stimulating letter about servo-controlled bass speaker systems. It's always nice to get thought-provoking comments. I hope my remarks will clarify my standpoint.

The Swedish audiophile Josef Svalander (designer of the all-tube preamplifier "Tuba," among other things) is a rare individual who has had the opportunity to create a truly ideal woofer system. He used a basement room as a speaker enclosure. So, from that point of view, you are right: ideal is best.

In general, it is also true that "more speakers" is better than corrections. It is interesting, however, that even many speakers in Infinity's large systems use servo controlling. Moreover, I am unsure it is a matter of "naturally" improving speaker performances. I cannot see much that is "natural" in highly sophisticated technological devices like speakers and enclosures. It is true, nevertheless, that from a purist's point of view, all the electronics servo systems require tend to lengthen the signal path, usually an undesirable effect.

Your objection that servo correction amounts to "amplitude compression" is very relevant and serious. I'll deal with that objection below, but one comment on your premises is needed: "In effect, this electrical signal (the inverse of the nonlinear signal fed to the speaker) literally tries to stop whatever the cone is doing at that time." This is not true, for several reasons. First, speakers shouldn't do whatever they choose, but should follow the electrical representation of the music as closely as possible. This is done in all sorts of ways in conventional speakers: by securing a near perfect power amplifier to speaker interface, by keeping the internal impedance of the power amp down, and so on.

Second, in a servo system, any cone movement is immediately converted to an electrical representation. In principle, this conversion is done at a speed of the order of the speed of light. In practice, quite a bit of delay is introduced. Compared to the speed of sound and the speed of cone movements, however, the electrical monitoring of the cone movements is much faster. It is so fast, it is fair to say the servo more resembles the eye-brain-sensation combination when you move your extremities toward a goal: a constant monitoring of whether or not the goal is reached. If reached, the movement is stopped; if not, it is continued. Consequently, a servo system basically provides an ultra precise control over the cone.

But you're right about the amplitude compression. The explanation is quite simple: Mr. Brown and I are trying to apply acceleration feedback to our systems. Klaassen and de Koning[1] show that pure acceleration feedback is the electrical equivalent of adding mass to the cone.

This has at least three consequences: lowering the resonance-frequency, increasing the quality factor (Q), and lowering the sensitivity. The relationship is this: increasing the cone mass by a factor of 2.5 lowers the resonance frequency f_0 by a factor of $\sqrt{2.5}$ and increases the Q by the same factor. The increased Q is taken care of by adding velocity feedback, which Mr. Brown's careful phase and amplitude checks have provided to some extent. As for my own design, refer to "An Acceleration Feedback System" (*SB* 1/90, p. 10). The decreased sensitivity is compensated for by an extra power amplifier.

But that is only a tiny part of the wretched business. Usually, a servo-controlled speaker's driver enclosure unit is of the acoustic suspension, or "infinite baffle" (IB), type. IB units quite often suffer from a sense of amplitude compression without corrections (even true, to my ears at least, of the famous LS 3/5).

Undersizing doesn't help. In principle, the servo ought to take care of this problem, but it doesn't (always) for many reasons. The first one comes to think of it as a power amp with too few muscles, as you indicate. That can be remedied without much effort or cost—at least if you're an amateur. But, honestly, I do not believe this is the worst problem.

I've tried my systems with heavy power amps with little difference compared with the 70W devices I'm currently using, but they provide a lot of current anyway. What I consider to be the worst problem is this: the accelerometer is mounted at the speaker's most rigid point and it monitors the movements at that point. What the rest of the cone is doing while playing heavy transients is not the accelerometer's business.

I don't believe ordinary cones in undersized acoustic suspension enclosures move as the voice coil tells them because the spring constant of the compressed air behind the cone influences the movement of different parts of the cone somewhat. How much depends on the cone in question. In some cases, undersizing, perfect electronic Q, and amplitude compensation result in lower distortion. In all, extremely rigid cones in reasonably sized enclosures ought to be a good bid. Modern Kevlar fibers and the like may be part of the answer.

Finally, your comments on the drivers' excursion limits are very relevant and true, even though I think you overrate the practical consequences. It is my experience that surprisingly little cone movement is required—even during loud music—in ordinary living rooms.

All this said and done, I wish to make it clear I don't know whether I "go in for" servo-controlled speakers as such. But I do find such systems interesting and experimenting with them is worthwhile. If I have given the impression that servo controlling provides something for nothing, my command of written English is worse than I thought. There's always a penalty to be paid. It is the designer's choice with which sort of currency he wishes to pay: space, drivers, enclosures, electronics, or any combination of these.

With my article, I only wished to share my enthusiasm with others. That is also why I tried to write the article the way I did: as bench notes. Nobody could be more sorry than I if I have inflicted wasted time, effort, and money on others.

Thank you for the opportunity to rethink problems. The questions, choices, and parameters are so many, the answers too often ambiguous.

[I invite Mr. Bradley's disclosure of which servo systems he has experienced, and especially any he has built himself.—E.T.D.]

REFERENCE

1. Klaassen, J.A. and S.H. de Koning, "Motional Feedback with Loudspeakers," *Philips Technical Review*, Volume 29, pp. 148–157.

A 5/2 SYSTEM

I have been interested in loudspeaker design and construction since I was a teenager and yearned for systems that were far beyond my budget. My main considerations have been clean, detailed sound and the ability to reproduce high SPLs. Although many of my systems have fulfilled the first requirement, I was inevitably disappointed by shrillness and ringing in the high end and bass bottoming during high-volume listening sessions.

When the Swan IV system was published (*SB* 4/88, p. 9), my brother and I were immediately struck by this system's dynamic capabilities. We had considered using dual 12" drivers per side and a D'Appolito-type configuration for the top end in our design, but since the Swan IV was fully designed, we decided to construct that system.

I cannot say enough about the Swan IV's quality and Mr. D'Appolito's and Mr. Bock's friendly advice and help. Their system has supplied many hours of high-quality listening. During high-volume sessions, the Swan IV outperformed any other loudspeaker I have heard. Although we have been quite pleased with this system, the need to experiment has urged us to try to improve the dynamic capabilities of even this formidable audio force.

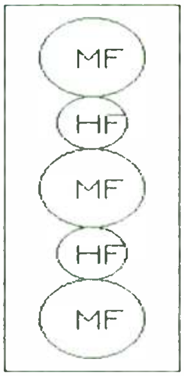


FIGURE 1: A 5/2 system.

This brings me to my specific questions. Has Mr. D'Appolito considered expanding his 3/2 geometry to a 5/2 system (*Fig. 1*) to reduce the strain on the mid- and high-frequency components? Would the separation between high-frequency components cause problems with the "D'Appolito effect"? The three midbass units would present either a parallel wiring problem (very low impedance at 2.6Ω nominal) or a series one (at least 12Ω). Any suggestions?

Also, when the Swan IV was in its initial design stages, did Mr. D'Appolito consider using a dual Isobarik or compound bass system? Since they had originally investigated closed-box designs for their purportedly superior transient characteristics, but abandoned them due to box size limitations, it seems a compound system would have fit the bill.

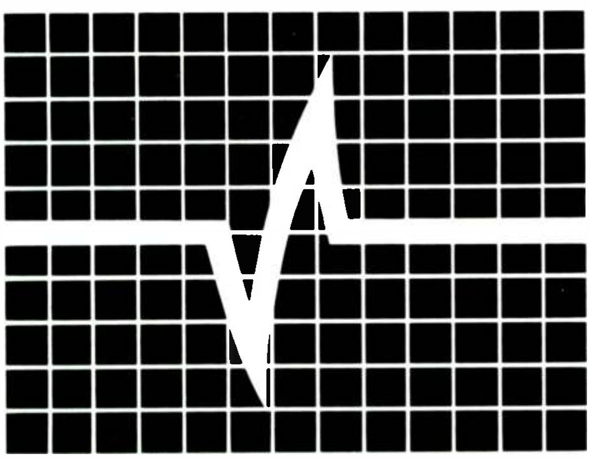
Lincoln Fowler
Milwaukee, WI 53202

Contributing Editor Joe D'Appolito replies:

The 5/2 configuration you suggest would not work well. There would be serious combing error in the vertical axis throughout the tweeter's working range. That is, there would be multiple nulls in the vertical polar response. Dynamic compression of the Dynaudio tweeter in the Swan IV is not a problem.

At the low end of the satellite's range, you could conceivably need some help. Here you could use four midbass drivers, two above and two below the tweeter. The outermost midbass units would be active in the 100–800Hz or so range. Their function would be twofold. First, they would double maximum SPL capability in that range. Second, by appropriate choice of crossover, they would compensate for diffraction loss. These would be single voice coil units like the Focal 5K013L.

A closed-box bass module was rejected for the Swan IV on the basis of maximum SPL in the lowest octave and not physical size. We considered a compound system, but dropped it because the bass module goes well up into the midrange and radiation off the back of the cone at higher frequencies is seriously compromised by diffraction and reflection off the driver frame. An Isobarik configuration was deemed too expensive and too difficult to build for the gain in performance. Again, there is a problem with interdriver cancellation over the active range of the bass module.



SWAN IV QUESTIONS

I have a couple of questions regarding Joe D'Appolito and James W. Bock's article in *SB* 4/88 (p. 9) about the Swan IV speaker system. Is there an advantage in placing the drivers in the symmetrical bass speaker in a push-pull configuration to cause odd-order nonlinearities to cancel and improve distortion? Also, can I get by with smaller drivers in a sealed enclosure with a Q_{TC} of 0.5? The distance between bass drivers and the opposite wall is less than 15 feet.

Steve Andersen
N. Highlands, CA 95660

Contributing Editor Joseph D'Appolito replies:

The Swan IV bass module uses a relatively slow slope crossover and produces significant output to several hundred hertz. These higher frequencies are not radiated efficiently off the back of a woofer. Serious response anomalies caused by diffraction and reflection off the woofer frame would be more objectionable than low-frequency distortion.

Your second question falls into the category of a system redesign, which we clearly stated in our article as being the type of question we would not entertain. Without trying the modification ourselves, we cannot guarantee the quality of the result.

I will, however, offer the following comments. Small rooms can provide substantial gain below their fundamental resonance modes. You may well realize satisfactory bass performance with smaller drivers in a sealed box, but it is something with which you must experiment. You might consider using two of the Eclipse 10W38Rs in a closed box with a resonant frequency in the range of 40-45Hz. The Pedal Coupler Bass boost circuit must be revised slightly to produce around 6dB of boost in the range of 27-30Hz. Alternatively, you can eliminate the boost circuit, use a Q in the range of 0.5-0.6, and hope the room boost is sufficient for satisfying bass.

THE ARIA 7

I need the assistance of Mr. D'Appolito, the designer of Aria 7, which I plan to build. I am working on the plans for the cabinets and have noticed a few discrepancies between the plans Focal has provided and your comments and the overall appearance of the cabinets versus their dimensions.

The volume is written in the accompanying article as 34 liters; however, using the plan's dimensions (less bracing and venting), the volume comes to 38.56 liters. What should it be? Is there some tolerance in which modifications can be made without changing the sound? Or has the design changed?

The width of the front is noted at 8½″ and the Focal speaker flange is 8″. This leaves only ¼″ on each side, making rounding the corners to ¾″ rather difficult. The picture of the completed speaker (*SB* 6/90 cover) looks more like ¾″ to 1″ per side. If this is the case, the total width would be closer to 9½″ to 10″, making it necessary for one or both other dimensions to change (to maintain the volume as a constant at whatever it should be).

Can the front be widened without changing the sound? If yes, by how much? What should the other dimensions be?

If any changes or improvements to the "7" have been made since the original plans/crossovers/speakers as provided by Focal America, I would appreciate your advice.

R.T. Brownlee
Clinton, NJ 08809

Contributing Editor Joe D'Appolito replies:

Unfortunately, I do not have a copy of the current Aria 7 literature. The dimensions of my cabinets, however, are 44½″ by 8½″ by 9¾″. These yield a gross internal volume of 40.7 liters. However, the port and vent tunnel occupy 5 liters and the drivers and crossover roughly another 2 liters for a net of approximately 33.7 liters.

The 7K011DBL frame is 7″ wide, not 8″. Thus, there is just enough room for the driver flange. Admittedly, this may present problems in construction. If you prefer, you can widen the front baffle to 9½″. You should then attain the correct volume by reducing the enclosure's height and not its depth. You must not change the port area; instead reduce its height to compensate for the increase in width from 7″ to 8″.

Also, do not change the vent length. With the wider front baffle, I strongly recommend that you mount the T120K tweeter slightly off-center to reduce the diffraction problems. With a 9½″ baffle, place the tweeter center 4″ from one edge. Make the speakers in left and right versions and place them in the listening area with the 4″ distance toward the center.

ROOM ACOUSTICS

I wish to express my thanks to Donald F. Scott (*SB* Mailbox 3/89, p. 61 and 6/89, p. 70) and add my comments concerning the subject of room treatment. Like Mr. Scott, I'm a "mature" audiophile, having put together my first hi-fi set in 1954. From time to time, I've attempted to improve my system by upgrading its components. For a long time, I thought that upgrading was the only way available and that I could overcome "deficiencies" if only I could afford this or that exotic component. I was wrong.

Mr. Scott's clear, unhyped, and persuasive discussion got me moving. I have made nine portable panels and arranged them around my listening room. Also, I placed four approximately 5′ by 6′ sheets of 2″ polyurethane foam on top of some panels. It's not a pretty sight and my wife gets canonized next year. But what it lacks in beauty, it more than makes up for in creating a "music friendly" environment. I know now what Mr. Scott means when he says something "sounds the way you always hoped it would sound." I won't bore you with claims of frequency response, SPL, and the rest. I now simply hear the orchestra in my room, not the recording. Instruments reveal their timbre, Kunzel brings his gang over any night I wish.

Not being a woodworker of even the most meager skill, I built my panels from "linen shelving," obtainable at home fix-up stores. It can be cut to any length and it's inexpensive, rigid, and mostly open air. I attached a layer of 2″ polyurethane foam to each side of this frame and for good measure put in 3″ of fiberglass insulation in the center (as per Mr. Scott). This has held firmly for almost a year using nylon wire ties obtainable at radio parts stores. I'm still experimenting with placement, but anywhere seems OK as long as most of the panels are between me and the walls.

I have a satellite/subwoofer system and I believe it lends itself particularly well to this treatment. My subs are in the corners of the back wall and the Spica Congelus pair are 6′ from the back wall and 4′ from the sides. The system is bi-amped and the subs are 15″ home-built, compound units in sonotubes.

I strongly believe any system will benefit from room treatment and the serious music listener should consider its possibilities and benefits. I'm glad I did.

Oz Miller
Lake Mary, FL 32746

D.F. Scott replies:

Thanks for your kind comments on my suggestions. I'm quite taken with the way you built your panels. Using nylon ties is especially noteworthy. This shows there are always better ways to accomplish something.

Your comment on bringing Kunzel and the gang in for an evening of entertainment closely parallels my listening experiences and the mention of Kunzel brings to mind an article by Jack L. Renner, Chairman of Telarc International in their "Quarter Notes" Vol. 4 No. 3. In his opening statement, he opines, "The use of acoustical materials in helping smooth out frequency response and control reflections in on-location control rooms (as well as permanent studios, concert halls, and home listening rooms) is absolutely essential." This seems to indicate we are not too far off the beam in recognizing the importance of that last link in the chain, the "speaker to ear" interface. That old "weak link in the chain thing" applies to the tail end, as well as to the middle.

I'd be interested in learning how you optimized your listening geometry. To establish best vertical stereo window, I assumed a normal listening position in the apex of a triangle and with a wide-range recording playing; then I crouched down near the floor and slowly rose to standing position. I went to an approximately 3′ tall window where detail really came alive. I figured this was the area where mid, woofer, and tweeter were in best alignment by time coincident with smoothest response through crossover. I then lifted the speakers to where this centered on a normal listening height.

For horizontal alignment, I started with toe-in axes intersecting about 1′ in front of the listening position. Then I fine-tuned toe-in, toe-out to where a 3-D illusion appeared behind the speakers. Too much toe-in caused the image to go toward monaural; too much toe-out brought on a flat ping-pong effect with no depth. When properly aligned, the speakers are sonically non-existent and you're in Symphony Hall with Kunzel breathing down the back of your neck—you're on the podium.

I have a burlap curtain behind my satellites hiding the subwoofers, plus a sheet of acoustifoam (5′ by 9′). All listeners, who are not familiar with my idiosyncrasies, swear I have a monstrous kilobuck system back there.

Tell me more about the acoustifoam over the top of your panels. Is this to tame ceiling reflections? An acoustic tile ceiling seems adequate in my setup. During early experimentation, I also found the most important thing was to erect an absorbent barrier between me and the walls or other reflective objects. The barrier seemed to work better by heading

We Enjoy Hearing From You.

Speaker Builder is a place for readers to meet and discuss ideas, projects, and challenges. When you write, please include a self-addressed envelope, with stamp attached with a clip, if you expect a reply. If the author lives outside of the US, please include two International Postal Reply coupons (available at your post office) instead of stamps. Because of space limitations, we reserve the right to edit all letters. Although we read and note all letters, only those of general interest are selected for publication. When responding to an article, please cite the name of the magazine, article, letter, department, and the page number. Address all correspondence to the **Letters Department**, *Speaker Builder*, PO Box 576, Peterborough, NH 03458-0576.

off stray sound as close to the speakers and listening spot as possible, without interfering with direct sound.

Reflected sound, which leaks through if unbalanced, will affect image and tonality. Hold a book or notepad (about 5″ by 7″) behind your right ear with the long dimension extending outward. Move the outer edge toward the front until it reflects more left speaker information than normal into your right ear. The image shifts to the right, the left side loses body, and the right side gains body and can't be completely corrected because tonality, as well as amplitude, is altered—reflected sound differs from direct sound.

I'll bet you've noticed that smoothing out response isn't the best part; it's preservation of transients that really brings things alive.

Keep me posted on any new methods or discoveries; I like to talk shop.

COMPUTING BOX RESPONSES

SB 6/90 (p. 24) published formulas for computing response of a vented-box system. Can you please also supply the formulas for a sealed-box system?

Philip Casper
Justice, IL 60458

Contributing Editor G.R. Koonce replies:

The response equations for the closed-box (CB), vented-box (VB), and passive-radiator (PR) systems are shown in *Figs. 1-3*. Each equation computes the relative response in decibels at the frequency of interest (f).

When a driver is mounted in a tuned enclosure, its resonant frequency slightly shifts (from f_s to f_{sB} in a VB). Since I believe people wish to investigate a system's performance before they construct it, I have used f_s in all the equations. This, in my experience, produces very little error and is no less realistic than assuming the total box Q is known before you construct the system.

The following terms pertain to each system:

f	Frequency of interest in hertz
f_s	Driver free-air resonance in hertz
f_B	Box tuned frequency in hertz
f_c	System resonance of CB system in hertz
f_p	Final passive radiator resonance when tuned in hertz

$$R_{dB} = 20\ \mathrm{LOG}_{10} \frac{f_n^2}{\sqrt{(1 - f_n^2)^2 + (f_n/Q_{TC})^2}} \qquad f_n = f/f_c$$

Starting with the T/S parameters and the net effective box volume, you compute f_c and Q_{TC} as follows:

$$Q_{TC} = Q_{TS}\sqrt{1 + V_{AS}/V_B} = Q_{TS}\sqrt{1 + \alpha}$$

$$f_c = f_s\sqrt{1 + V_{AS}/V_B} = f_s\sqrt{1 + \alpha}$$

FIGURE 1: Closed-box system response equation.

$$R_{dB} = 20\ \mathrm{LOG}_{10} \frac{f_n^4}{\sqrt{(f_n^4 - C f_n^2 + A)^2 + f_n^2 (D f_n^2 - B)^2}}$$

$$f_n = f/f_s$$

$$A = f_B/f_s)^2 = h^2$$

$$B = A/Q_{TS} + f_B/(Q_B f_s) = A/Q_{TS} + h/Q_B$$

$$C = 1 + A + V_{AS}/V_B + f_B/(f_s Q_B Q_{TS}) = 1 + A + \alpha + h/(Q_{TS} Q_B)$$

$$D = 1/Q_{TS} + f_B/(f_s Q_B) = 1/Q_{TS} + h/Q_B$$

FIGURE 2: Vented-box system response equation.

$$R_{dB} = 20\ \mathrm{LOG}_{10} \frac{f_n^5 - f_n^3 (f_p/f_s)^2}{\sqrt{f_n^2 (f_n^4 - B f_n^2 + D)^2 + (A f_n^4 - C f_n^2 + E)^2}}$$

$f_n = f/f_s$ Numerator can be written as: $f_n^5 - f_n^3 y^2$

$$A = 1/Q_{TS} + f_B/(f_s Q_B) = 1/Q_{TS} + h/Q_B$$

$$B = 1 + V_{AS}/V_B + f_B/(f_s Q_{TS} Q_B) + (f_p/f_s)^2 (1 + V_{AP}/V_B)$$

$$= 1 + \alpha + h/(Q_{TS} Q_B) + y^2 (1 + \delta)$$

$$C = (A + V_{AP}/(V_B Q_{TS}))(f_p/f_s)^2 + f_B/(f_s Q_B) = y^2 (A + \delta/Q_{TS}) + h/Q_B$$

$$E = (f_p/f_s)^2 f_B/(f_s Q_B) = y^2 h/Q_B$$

$$D = (f_p/f_s)^2 (1 + V_{AS}/V_B + V_{AP}/V_B) + E/Q_{TS} = y^2 (1 + \alpha + \delta) + E/Q_{TS}$$

FIGURE 3: Passive radiator system response equation.

Q_{TS} Driver total Q at f_s
Q_{TC} Closed-box system total Q at f_c
Q_B Total box Q for VB or PR system (sometimes called Q_L)
V_B Net effective box volume in same units as V_{AS} and V_{AP}
V_{AS} Driver compliance volume
V_{AP} Passive radiator compliance volume

The following standard relationships are defined:

f_n Normalized frequency—all response equations are in terms of f_n; f is the frequency of interest
f_n f/f_s for VB and PR systems
f_n f/f_c for CB system
h f_B/f_s for VB and PR systems
y f_p/f_s for PR system
α V_{AS}/V_B for all systems
δ V_{AP}/V_B for PR system

For information on obtaining the parameters for the passive radiator unit, see the references below. Remember, the value of f_p used in the equations is the final value after the box has been tuned.

REFERENCES

1. Dickason, Vance, *The Loudspeaker Design Cookbook*, Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458, p. 30.
2. Koonce, G.R., "Finding f_p for Passive Radiator Speakers," *SB* 4/81, p. 25.

KLH 9 PARTS

Having been in the consumer electronics business for many years, I purchased four used KLH model 9 electrostatics. I have owned them in the past, and so I was disappointed to find all conventional sources for parts long gone.

Mike Thompson at KLH, a subscriber, advised me to see your recent issues for schematics for the model 9. Also, I need *any* source you can suggest for ES panels and other items for rebuilding these great speakers.

Gary Beckstrom
Moorpark, CA 93020-8099

Roger Sanders replies:

The KLH 9 used a single Janszen tweeter and four bass panels made for or by KLH. (I'm not sure who actually manufactured them.) I know of no contemporary source for these panels. I haven't seen your speakers, but will assume they have not been physically damaged. If so, they can be fixed. The problem surely is either in the electronics, the connections, the diaphragms, or a combination of all three.

Like most electrical problems, check out the obvious first. Make sure you actually have drive and polarizing voltages at the ESL panels. My experience with the Janszen tweeters suggests you should look closely at the contacts for the polarizing power supply. These tend to oxidize, which prevents transfer of the charge to the diaphragm.

Keep in mind that it is unlikely that *all* circuits have failed. If you get zero output from all panels in all speakers, look for something like blown fuses, power supply failures, or other factors that could affect the entire speaker.

If you get voltages at all the panels and still no sound, suspect the panels themselves. The weak point will be the diaphragms. The diaphragm contacts could have failed, but foreign material between the diaphragm and a stator can short it out. Again it is unlikely all diaphragms have failed. If all the panels in all the speakers produce no sound, you probably should look elsewhere for the trouble.

Replacing the diaphragms requires no magic. You can do so by running a *sharp* putty knife around the perimeter of the cell to separate the stators. Using the putty knife, clean off the glue and old diaphragm material from the insulators.

Installing new diaphragms is beyond the scope of this letter, but is not difficult. I suggest you follow the instructions in my ESL article in *SB* 2/90. Somewhat more detailed instructions are in my upcoming book, *The Electrostatic Loudspeaker Design Cookbook*, which I expect to be available in late 1991.

Having told you how to fix your KLH 9s, I must now say you shouldn't bother—they are obsolete. Their performance falls far short of modern ESLs in three critical areas: output, frequency response, and imaging.

Their output is poor because they use dipole woofers. Their frequency response is poor because they deal with phase cancellation in the bass and midrange by increasing the diaphragm area at lower frequencies. This concept is sound, but its execution in the KLH 9 is woefully inadequate. The imaging is poor because you must listen to them off-axis.

Because you have owned KLH 9s in the past and liked them, you understandably are probably ques-

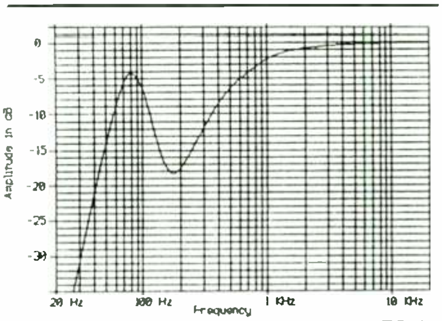


FIGURE 1: General trend of a full-range ESL's frequency response.

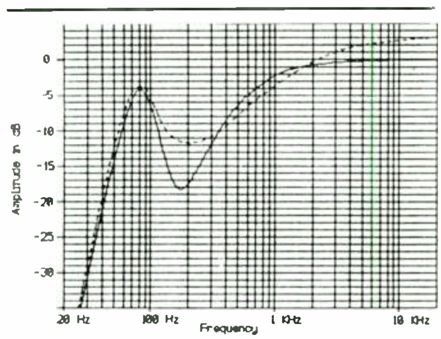


FIGURE 2: Frequency response trend of the KLH 9.

tioning why I have criticized them so harshly. Permit me to explain.

KLH chose to use *segmentation* to correct the frequency response problems inherent in dipole radiators (speakers that radiate sound in front and behind them). This means they used crossovers and shifted the drive to smaller area drivers with increasing frequency. In the case of the KLH 9, they changed from four large bass panels to a single small 4" square tweeter at around 1.5kHz. Such a huge change in radiating area over such a small frequency range doesn't come close to matching the falling frequency response curve for which the segmentation must correct.

Figure 1 shows the general trend of a full-range ESL's frequency response of the approximate dimensions of the KLH 9. The large peak at around 70Hz is the speaker's fundamental resonance. *Figure 2* shows the frequency response trend of the KLH 9 with its single step of segmentation overlaid on *Fig. 1*. The dashed line is the speaker's response. This is an *on-axis* response. Let's analyze what's going on here:

- The poor deep bass performance is due to the severe phase cancellation and radiation resistance losses suffered by all dipoles at low frequencies.

CAVEAT CORRESPONDENTS

Things that go bump in our round file:

1. "I'm thinking of building a 16-in, 8-out console in my basement. What tape recorder should I buy?"
2. "Is my Fisher Z-705 receiver worth updating? Where should I begin?"
3. "Although I forgot to enclose a stamped, self-addressed envelope, please answer the following nine questions based on my experiences building your inverted RIAA kit."
4. "Please forward this (unstamped) letter to Ralph J. whose letter appeared in one of the 1970 issues—don't remember which."
5. "I have a Milhous 10W integrated stereo amplifier and a Gesundheit turntable. Which of the following six cartridges would you recommend?"
6. Queries with no stamped, self-addressed envelope or postal coupons enclosed.
7. Letters without return addresses on them whose envelopes have strayed away somewhere.
8. Illegible hand-written letters scrawled on odd scraps of paper. If you have no access to a typewriter, please try to be sure our typesetter doesn't lose his eyesight and his mind in deciphering your writing. (This is especially important if you want us to publish your classified ad.)

- The midbass peak is due to the speaker's unsuppressed fundamental resonance.
- The large "suck out" in the lower midrange results because the tweeter's area is too small to match the bass panels in the 500Hz to 1.5kHz range.
- The generally falling midrange response is due to the relatively small tweeter area compared to a large-diaphragm, full-range ESL.
- The hot high end is caused by boosting the overall tweeter level to try to compensate for the lack of midrange output. In the bandwidth covered by the tweeter linearly (6kHz and up), the tweeter actually has very linear frequency response.

KLH couldn't do anything about the *deep* bass, but to compensate for the phase cancellation in the *mid*bass, they left the fundamental resonance in place. This made the speaker sound "bassy" at the expense of bass linearity—and it compromised output.

The midrange was a major problem. The solution to this problem in a segmented speaker is to use many segments of different sizes as did Quad. Instead, KLH increased the tweeter output excessively to pull up the midrange, then advised owners to listen to their speakers *off-axis* to try to tame resulting sizzling high end.

Their tweeter was laser-beam directional as are all relatively large planar radiators, and listening off-axis greatly attenuated the highs. Unfortunately, off-axis listening completely destroys any semblance of precise imaging. Also, the high-frequency balance is heavily influenced by the room's absorptive qualities.

The aesthetics of the KLH 9 are wonderful. It is a beautiful-looking speaker. Despite its sonic flaws by today's standards, it was head and shoulders above magnetic speakers of that vintage with regard to detail, transparency, and delicacy—legendary traits shared by all ESLs.

Unless you value the speakers as collector's items, I recommend you gut the enclosures and build new cells and electronics for them. This probably wouldn't be much harder than rebuilding what you have, and the performance difference would be astounding.

I suggest you build two large cells to fill each enclosure per *Fig. 3*. Operate these cells full-range using equalization to correct the frequency response problems. My 1980's articles in *SB* explain how to do this as does my book, but you can get an adequate idea from my 1990 *SB* articles.

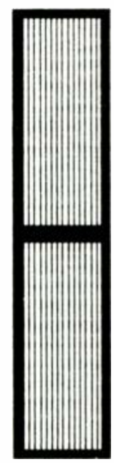


FIGURE 3: Dimension these two cells to fit within the normal KLH 9 enclosure.

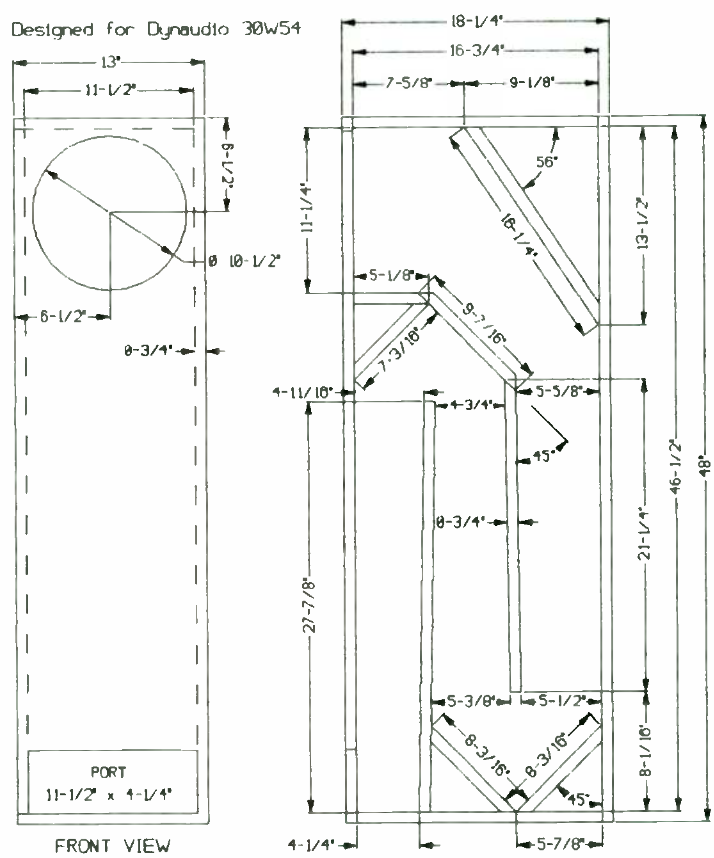


FIGURE 4: A reasonably compact transmission line enclosure capable of electrostatic quality bass when used with a Dynaudio 30W54 driver.

It is impossible to get high output and deep bass simultaneously from a relatively small dipole ESL like the KLH 9. Therefore, I suggest you mate them to transmission line woofers using a 400Hz crossover point. Doing so will dramatically increase output and deep bass. *Figure 4* shows a reasonably compact transmission line enclosure capable of electrostatic quality bass when used with a Dynaudio 30W54 driver.

While you probably agree a magnetic woofer produces deeper bass and more output than an electrostatic woofer, I suspect you believe a magnetic woofer system cannot hope to match the detail you hear from the KLH 9's electrostatic bass. Let me assure you this is not the case. I have done many tests with skeptical "golden ear" listeners and proved to them that the bass from a well-designed transmission line is as clear and detailed as electrostatic bass—while providing *linear* frequency response and high output levels.

If you still have doubts, feel free to visit me and hear my system, which is very similar to what I just described. If eastern Oregon is too far for you to travel, perhaps the Sacramento area would do. My co-author Barry Waldron lives there and I'm sure he would be happy to have you audition his ESL system.

If you have questions, please contact me. In any case, tell me what you do with your KLH 9s.

WATERED-DOWN SILICONE?

Elmer's glue is water soluble, so it can be "watered down" to become "spreadable." This allows it to be spread easily over paper woofer cones, thereby plasticizing them.

How can this be accomplished with silicone? Is there a substance in which silicone can be dissolved sufficiently so you can "siliconize" or rubberize woofer cones as per above? This would also be handy for sealing woofer cloth surrounds and dust caps, thus abating air leaks.

Any response from the readership would be greatly appreciated.

Angel Rivera
Brooklyn, NY 11204

DIGITAL RECORDING

I've been involved with audio sound since 1987 and wish to have whatever is recorded sound as good as the original. Since the advent of a digital recording medium, it seems as though whatever medium's being used to record, it's unlimited in design, as long as the reproducing end is accurate.

I realized this when I was visiting House On the Rock, a tourist attraction near Dodgerville, WI. What impressed me were the real instruments, playing familiar music, with a recorded medium controlling various relays, air compressors, things that pushed and pulled at violin bows, piano keys, wind instrument keys, and percussion instruments.

This in an experience not soon forgotten. On some classical songs, the music becomes very loud, and all the while maintaining exceptional sound quality. My first thought was "Wouldn't it be great to have something like this at home?" It's the closest you could get to being live without having the performers there.

Maybe this could be the next generation of home audio. It sure would be nice and solve those messy calculations of speaker type and cabinet-size matching. Now, all I'd have to do is learn how to tune all those instruments so they sound as good as the original.

Steve Fisher
Ames, IA 50010

COMPUTER HUBRIS

Mr. Tucker's article in *SB* 3/90 (p. 26) was a nice lesson in computer hubris. A slide rule would have told him right away he was in trouble with the MW162 and spared us things like 22.2442331 liters and 33.4647658Hz.

He chose a closed box (V_B = 22l) and took Morel's specs (V_{AS} = 16.5l, Q_{TS} = 0.87). So will we.

Following Small, $\alpha = V_{AS}/V_B = 0.75$. Note that this qualifies as a "wastefully large" box according to Small, who preferred $\alpha = 3$ for economy. Mr. Tucker's knickknacks are being unreasonably crowded.

Continuing, $Q_{TC} = Q_{TS}\sqrt{1+\alpha} = 1.15$. No good; it's too high to risk if you are aiming at 0.9 as was Mr. Tucker. Don't rely on stuffing to help; let it compensate for the ignored volume taken up by the driver. But Mr. Tucker actually measured Q_{TC} = 1.4, about 20% higher still. The reasonable conclusion is that his driver's actual Q_{TS} (unreported) was about 20% high. But, if actual Q_{TS} = 1.0, then Mr. Tucker is lucky he didn't build the 30l boxes he considered. They would give:

$$\alpha = 16.5/30 = 0.55$$

Then:

$$Q_{TC} = Q_{TS}\sqrt{1+\alpha} = 1.0\sqrt{1.55} = 1.24.$$

Still too high. When α is so small, the $1+\alpha$ operation ruins you. Besides, you can't do it on a slide rule.

The Morel 164 with Q_{TS} = 0.53 would be a better bet, but check actual Q_{TS} first. So, in the end, all computations were off and the humps were dispelled by two of those mysterious, unquantified bathroom ventilators. Or can they be modeled?

Derek Wooldridge
Princeton, NJ 08540

continued from page 69

SPEAKER CABINET STANDING WAVES

A common misconception seems rampant among speaker builders that rectangular speaker cabinets are particularly susceptible to internal standing waves. The concept is important and is worthy of our attention because standing waves are detrimental to the speaker's sound.

As an example of popular statements regarding standing waves, typical commercial advertising in *SB* has commended nonrectangular enclosures (Focal's Egg and Transcendental Audio's Un-box, for example) as being beneficial. Also, various *SB* authors[1–4] refer to an absence of standing waves in their odd-shaped cabinets, with the presumption this is beneficial.

While it is perhaps true that certain combinations of height-width-depth ratios of rectangular cabinets are not optimal with respect to enclosure resonances, it is not at all clear that odd-shaped cabinets are any better. In theory at least, nonrectangular enclosures do not prevent the generation of internal standing waves. The purpose of this article is to convince the average speaker builder that this is true, both by application of mathematical theory and by calculation of special cases.

To begin, consider a rectangular enclosure. The cabinet is assumed to have rigid walls. Although this is demonstrably not true in cheaper speaker systems, it really does not significantly affect our results. Also, the effects of internal damping are not of primary influence unless it is extraordinarily dense.

A standing internal wave is one which, once generated, maintains at least a few repetitive oscillations at a fixed frequency without further input of energy. In the jargon of mathematical physics, the spatial characteristics of these standing waves are typically called resonances, normal modes, eigenmodes, or the like, and the associated frequencies are called natural, resonant, or eigen-frequencies.

Each enclosure always has a lowest frequency standing wave and an infinite sequence of higher order ones, the specific details of which are associated with the shape of the cabinet. For the rectangular example, assume the height H is greater than the other lengths. Then, the mode having the lowest frequency is the standing wave or fundamental mode, which exhibits a vertical oscillation.

A horizontal plane across the middle of the enclosure is a node that exhibits no pressure changes, while the upper and lower halves of the cabinet exchange maximum and minimum pressure over one-half the period of the wave. The result is the same as the lowest standing wave in a room, and it is explained in many articles[5,6] and books.[7] The fundamental sloshing mode in your bathtub is a perfect analogy, where the surface displacement of the water is the analog of the acoustic pressure in the cabinet.

This fundamental mode's frequency is:

$$f = 0.5 \times C/H$$

where C is the speed of sound in air, about 1,100 ft./sec. This frequency is independent of the cabinet's width and depth; for a 3′ cabinet, this is about 180Hz.

This is in a typical woofer's operating range, and this mode will be excited efficiently if the driver is near either end of the cabinet. In fact, the next higher longitudinal mode (first overtone or second harmonic) occurs near 360Hz and is optimally excited by a woofer placed in the center or near either end of the cabinet.

Standing waves in the other directions also occur in the cabinet, and they all have frequencies higher than the fundamental. These occur closer together, and the modes are more easily dissipated by damping materials in the cabinet's interior space. Everest gives a good description of these modes and the associated frequencies.[7]

The rectangular enclosure is a particularly useful one because it is easy to construct from stock lumber but, more importantly in the present context, because it is amenable to an especially simple mathematical solution for the standing waves. The mode shapes are sinusoids, and the natural frequencies are related by integer multiples of the cabinet dimensions.

Now, consider a nonrectangular enclosure. Right cylinders having horizontal top and bottom panels, but more complicated geometrical shape around the perimeter are popular, and examples include the pentagon,[2] circle,[3] and hexagon.[8] The important point is that the longitudinal modes in the vertical direction discussed previously are not affected by these nonrectangular enclosure sides. The fundamental frequency is unchanged if the height is larger than the lateral dimensions, and a 3′ high cabinet will exhibit a 180Hz standing internal wave resonance no matter what the particular shape of the sides.

Even the lateral modes, directly affected by the more complicated sides, do not change appreciably with the shape of the sides. As an example that is almost as easy to calculate in theory as the rectangular cabinet, consider the right circular cylinder such as the Transcendental Audio Unbox. It has a diameter D that is larger than the cabinet height. The driver on the side will excite a fundamental mode which, looking down from the top, will have a nodal plane coincident with the diameter. This mode's frequency is approximately:

$$f = 0.59 \times C/D$$

independent of the enclosure height. For a diameter of 1', this frequency is approximately 650Hz, and the side-mounted driver can generate this mode efficiently. This is a well-known mathematical result, with the specific noninteger constant arising because the solutions of the equation of motion of the air in the cylinder are now Bessel functions instead of sinusoids as in the rectangular enclosure.

The water-wave analogy is easy to excite in my circular hot tub as a side-to-side sloshing mode. Higher frequency modes also exist, as in the rectangular enclosure, and examples of the mode shapes are available.[9,10] The natural frequency given above is not identical to the frequency of the first transverse mode given in these references because the boundary conditions for the standing acoustic wave of interest is different from the boundary conditions for the standing membrane waves presented in those references.

Cabinets with no parallel sides are more complicated to analyze mathematically, but even oddly shaped ones like the JBL L250 can be analyzed. This enclosure is distorted from a rectangular shape, with the top being narrower than the bottom, and the fundamental longitudinal mode is much like that in a rectangular box of nearly the same dimensions.

I think by now you get the gist of my argument. The details of the mathematics change, but not the fundamental physics for an enclosure of any shape. The Focal Egg, for example, has standing waves like any other enclosure. The fundamental longitudinal mode is similar to those discussed above. For an Egg of 1' internal height, the fundamental frequency would be a little higher than the cylinder of the same dimension, probably in the vicinity of 700Hz. For other sizes of an egg-shaped enclosure, this value scales inversely with the longer dimension as in the formulas above.

In light of the above discussion, I hope I have convinced you that the purported "advantage" of odd-shaped enclosures does not exist. My opinion is that the "advantage" exists only in the uninformed, or perhaps clever, marketing departments of the commercial speaker manufacturers.

On the other hand, nonrectangular sides do appear to have other advantages. A narrow front panel and curved sides are effective treatments to reduce imaging degradation due to edge diffraction.[10,11] Also, curved panels are stiffer against flexural vibrations than flat ones of the same wall thickness, so curved panels can reduce colorations due to panel motion. These panel motions are combinations of panel resonances, a different type of vibrational mode than the internal standing waves discussed above. But these resonances are another story.

John P. Dugan
Burke, VA 22015

REFERENCES

1. Hope, S., "Custom Cabinet Designs," *SB*, 4/85, p. 34.
2. Coyle, D.P., "Building Pentagonal Enclosures," *SB*, 1/86, p. 28.
3. Allison, C., "The Un-Box Enclosure," *SB* 3/86, p. 19.
4. Davenport, D.W., "Focal Model 280 Loudspeaker," *SB* 3/86, p. 30.
5. Muxlow, P., "Loudspeaker Cabinets," *SB* 2/88, p. 24.
6. Bolt, R.H., "Note on Normal Frequency Statistics in Rectangular Rooms," *J Acoust. Soc. Am.*, 18, 1946, p. 130.
7. Everst, F.A., *The Master Handbook of Acoustics*, Tab Books, 1981.
8. Edgar, B., "The Jordan System 5: A Woodworker's Approach," *SB* 2/84, p. 12.
9. Rayleigh, J.W.S., *The Theory of Sound*, Vol. 1, Dover, 1945, p. 331.
10. Olson, H.F., *Acoustical Engineering*, Van Nostrand, 1957, p. 62.
11. Colloms, M., *High Performance Loudspeakers*, 3rd Ed., J. Wiley. 1985, p. 228.

Classified Advertising

TRADE

MYLAR FILM. ½ mil, genuine DuPont, 48" wide, $1 per running foot; also ESL transformers. ***ROGER SANDERS***, R1, Box 125, Halfway, OR 97834, (503) 742-5023. T3/91

MIT AUDIO CABLES, Van den Hul, Cardas, MAS Masterlink, OCOS, Magnan, including Teflon insulated hookup wire for rewiring components; MIT capacitors; Solen polypropylenes to 200μF!, Wonder-Caps-solder-wire; HOLCO, Vishay & Resista resistors; Music posts & all types of audio connectors; RAM tubes; Tone Cones; NAVCOM products; Shallcross attenuators, many parts & accessories. $1 catalog ($3 overseas). ***MICHAEL PERCY***, Box 526, Inverness, CA 94937, (415) 669-7181. T6/91

CUSTOM PASSIVE CROSSOVERS built to your specifications. No minimum quantity. Only the finest components used. Quick response, satisfaction guaranteed. ***AMS ENTERPRISES***, 1007 Capello Way, Ojai, CA 93023. T2/92

CUSTOM MADE PASSIVE crossovers at reasonable prices. Write your needs to ***CUSTOM SOUND***, Box 3853, Antioch, CA 94531-3853. T5/91

SPEAKER MEASUREMENT SERVICES. Get the data you need to make first class speakers! We can measure Thiele/Small parameters, frequency, impedance, phase-response, and interdriver-offset needed to run crossover-design programs like LEAP, XOPT, CALSOD, CACD, or COMTRAN. Complete computer-optimized active/passive crossover/filter design with optional assembly is also available. ***TCSOUNDS LABORATORY***, Thilo Stompler, 13173 Portofino Dr., Del Mar, CA 92014, (619) 792-7924. T3/91

WHALE COVE AUDIO for SWAN IV drivers and crossovers, Pedal Coupler I or II active crossovers, and the Right Whale Kevlar mid-range, ceramic tweeter and crossover set for either the SWAN IV or the ARIA 5. Box 98, Swan's Island, ME 04685, (207) 526-4343. T2/92

YOUR 50 WORDS
in our private classifieds
ARE FREE
OVER 50: Send money
(20 cents/word)

SPEAKER BUILDER CLASSIFIED ORDER FORM

PLEASE FILL OUT IN CLEAR BLOCK LETTERS OR TYPE. PLEASE SPELL OUT ENTIRE WORD.

1	2	3	4	5	6	7	8	9	10
11	12	13	14	15	16	17	18	19	20
21	22	23	24	25	26	27	28	29	30
31	32	33	34	35	36	37	38	39	40
41	42	43	44	45	46	47	48	49	50

☐ Private For Sale ☐ Private Wanted ☐ If Trade, please indicate # of insertions ______

Private For Sale ads are for *non-profit* use. Individuals selling items for retail/profit should be using **Trade** ads. **Private For Sale** and **Private Wanted** ads are on a one time insertion basis. Ads **must** be resubmitted for renewal. **Ads will not be taken over the phone.**

Definition of a word: A series of letters with a space before and after. Count words, not letters. Any words over 50, 20¢ per word. Trade ads are 70¢ per word and indicate number of insertions. Deduct 10% for a 4× contract in *Audio Amateur* and a 6× contract in *Speaker Builder*, deduct 5% for a 2× contract in *Glass Audio*.

Please remember to include your name, address and telephone number. In order for Private For Sale and Want ads to be **free**, you must be a current subscriber. *Ten dollar minimum for charge card orders.* **Payment MUST accompany first ad.**

Please charge to my MasterCard/Visa:

CARD NUMBER EXP DATE

☐ *Check/Money Order Enclosed* ☐☐☐☐☐☐ Magic No.

NAME

PHONE

COMPANY

STREET & NO

CITY ST ZIP

Electronic

by

ACE AUDIO CO.

• est. 1973 •

#532-5th Street

E. Northport, NY 11731-2399

write for **free flyer**

Crossovers

NEW MATCHED PAIRS of Gold Ribbons, the best ribbon transducer available, build the ultimate speaker that beats IRS/WAMM. Retail $2,000/pair, now 50% off. $1,000/pair. ***SERIOUS LISTENERS***, (415) 365-5032. T3/91

ROTTEN FOAM? Re-surround any speaker! (412) 375-9203. T6/91

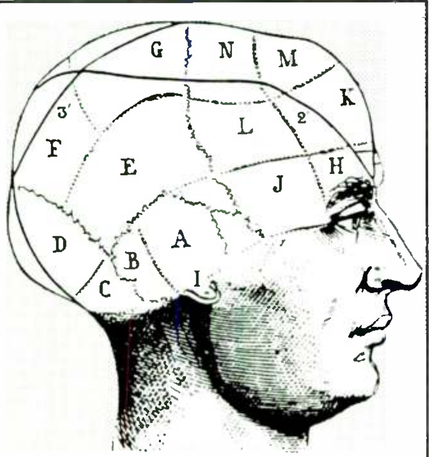
WANTED

Cover for EICO HF35 or HF50; Pr of MET 7, TC50 or SL6 reasonably. Info on Curcio Daniel II preamp, problems? Joe Jennings, 4903A South Orchard, Tacoma, WA 98467, (206) 473-3572

Any information on modification rewiring or improvement to Infinity 4.5 reference speakers will be appreciated. Jonathan Usha, 1261 East 102 St., Brooklyn, NY 11136, (718) 444-5773.

AR remote control, Paul Salamon, 2052 East Blvd. SE, Cedar Rapids, IA 52403, (319) 363-3173.

Technics SB 7070 speaker systems, cabinet must be in good condition. Debbie, (904) 392-0622.



FILL IN THE BLANKS JOIN AN AUDIO CLUB

Learn about the latest equipment, techniques and recordings through group meetings, tours, and newsletters. Ask questions. Share viewpoints and experiences. Stretch your mind.

If there's no club in your area, why not start one? Our club ads are free up to 75 words ($.20 per word thereafter). Copy must be provided by a designated officer of the club or society who will keep it current.

B&K model 707 tube tester; two pair Genalex KT66 tubes; Leak TI/25 plus amplifier. Stan Bungard, 915 Brickner Rd., College Place, WA 99324, (509) 529-8715.

Any information on Bozak CM199-1 or CM199-2 speaker systems. Believe they may use Bozak B199AC 12-inch drivers. Need specs on drivers or systems or any info available. Will pay copy and mailing costs. Don, 112 St James Ave., Merchantville, NJ 08109 or (609) 665-6512 evenings.

One Quad ESL, black only. Cash or trade for VPI TT Isol. L. Larsen (301) 587-8143 evenings.

Do you sell spiders? Voice Coils? Surrounds? Cones? Dust Caps? Adhesives? If you sell these components for speaker assembly, please contact us. Thompson's Electronics, 1122 Oak Street, Eugene, OR 97401, (503) 343-9273.

Replacement parts-information wanted for Daniel Queen omnidirectional tower loudspeaker system. All letters answered. Send all correspondence to Attn: Speaker Parts, PO Box 302, Middlesex, NJ 08846

Audio Control SA3050A ⅓-octave spectrum analyzer or equivalent. Peter Bandurian, (303) 666-7588 evenings.

Who in Los Angeles area has built ribbon speakers and would like to embark on another project? I seek a partner. Eduard, (213) 395-5196.

FOR SALE

Win Labs SDC-10 reference turntable, have original packing crate, as new, top rated turntable, $1,800. Walter Leemans, (403) 282-1862.

Thorens turntable TD-147 Shure 97HE, all accessories, used only 250 hours to tape record, mint condition, $250. Eugene Marshall, 34-13 80th St., Jackson Heights, NY 11372, (718) 478-9011.

JBL L-300 pair, $1,700 or best offer; JBL B-460 18" subwoofer with electronic crossover, $500 or best offer; Gold Sound 15" 4Ω woofer pair, new, in factory boxes, $200. Jim, (702) 746-9513 (NV).

One pair show horns, bass cabinet only, bench, no drivers, $450. Will deliver 100 miles Chicago, IL. Jim Eldridge, (815) 469-3204.

Kenwood passive reverb and light display with depth and time delay controls, $100; SEAS 6½-inch woofer, tweeters and crossover from Dynaco A-10, $75. All items sold plus shipping. Mr. Willes, (800) 227-6121 weekdays from 9-5 central time.

Two Focal T120KF tweeters, used once, $50/pair; four Focal 5N411L, never used, still in boxes, $40. Art, (301) 583-8293 or (301) 893-3463.

Accuton C2-11 tweeters with ultra stiff diaphragms, $240/pair. Bernard Diaz, (415) 930-9104.

Twelve JBL 117H-1 8" woofers, factory sealed, $40 each; four SEAS P14RCY 5" woofers, never used, $20 each; four Dynaudio 24W100 9" woofers, excellent condition, $75 each; two Eton 11-580 11" woofers, excellent condition, $135 each. Clay Jordan, (404) 969-8319 after 5 p.m. EST.

17½" x 11½" x 25" professionally built walnut speaker cabinets, undrilled baffles, perfect, $60/pair; Electro Voice SP15B Radax super fifteen (1965), mint, $45; JBL N2400 crossovers, mint, $40/pair; McIntosh MQ101 equalizer, walnut case, panlocs, manual, pristine, $150. Horace, (703) 578-4929 VA.

Electro Voice ST350B 8oz. tweeters, good condition, $100. David, (614) 267-9476.

Bose 901 Series 5 with equalizer, excellent condition, $595/pair; CV D9s, two pair, $495/pair. Good condition. Derek Galatas, 31125 Marshall Vaughn Rd., Lacombe, LA 70445, (504) 882-3955.

Rare and unusual KEF model K-2 Celeste MK II two-way speakers made in 1966 in England. Solid wood construction, never abused, first class condition, woofer is oval shape, truly collectors' items. Good deal for the right person, cash price, $200/pair, I pay shipping. Ed, (305) 891-2267.

CLASSIFIED INFORMATION

TYPES:

For Sale are personal and not for profit.

Trade feature items for sale or resale at a profit.

Wanted can fulfill your wildest dreams.

Club ads can help you start a club or find new members. Items cannot be offered for sale in "Club" ads.

PRICES:

For Sale, Wanted, and *Club* ads are free to subscribers up to 50 words; 20 cents per word thereafter. Don't forget to count your name, address, and phone number.

Please: only one ad per category per issue. Ads are open to nonsubscribers at 20 cents per word. All overruns and nonsubscriber ads must be prepaid.

Trade ads are 70 cents per word, prepaid. Two tearsheets are sent to the advertiser upon publication. Call for information about classified space ads.

WORDS:

We count words as any collection of letters or numbers with a space on both sides. No abbreviations, please, spell out all words. Ad copy should be clearly printed or typed, preferably submitted on the *Classified Order Form.* Illegible ads will be discarded.

DEADLINES:

Ad copy must arrive by the first of the month of the issue mailing. Ads arriving late will be run in the next issue. Ads are run only once, then discarded.

DON'T PHONE:

Personal ad questions, copy, and copy changes *cannot and will not* be answered by phone. All correspondence must be in writing.

MAGIC:

Please include your Magic Number with your ad copy (located in the upper left corner of your mailing label). If only your name and phone number are to appear in the ad, your address must accompany your copy.

R.S.V.P:

For acknowledgment of your ad, including issue date, include a stamped, self-addressed postcard.

KLH Model 7, $250; Tektronix 502 dual beam scope with C12 Polaroid camera, $200; Dyna PAT-4, $75; two sick Dyna ST-120s, $75 each; MacIntosh 60 output transformer, $200; HH Scott tube amp, 50W mono drives, 0.7Ω to 600Ω, $200; Mattes SSP200 III, 200 w/channel, $100; phase linear 1000, $125; EV Artistocrat clone w/EV 12 TRXB, $150; K-Horn clone (SpeakerLab), $300; Sony PCM F1, $850; AR-1W woofer, $150. Marshall Buck, (213) 559-3946, FAX (213) 836-3763.

IQS 401 FFT spectrum analyzer, IQS impedance analyzer, Apple II+, Apple dual disk drive, Epson MX80 printer, $750 for all. PAC, 120West 5th St., Frederick, MD 21701, (301) 694-7874 or (301) 847-5622.

Two Dynaudio D54AF, new, $110/pair; four AR3A woofers, new, have T/S parameters, $140; Precision TA305F5W dual coil, $30. All plus shipping. Phil Roy, (606) 342-7124.

Four Eclipse 1038R woofers, $100; Soundcraftsman DX4000 preamplifier, $195; AC23 Rane three-way stereo crossover, $225; Swan IV system without active crossover, $600. Chris Manikas, 412 N Second St., Chambersburg, PA 17201, (717) 263-3751.

Below wholesale, $10 misc. electrical parts only, $2 + 50¢ postage and handling. Tom Williams, Box 297, Middlebranch, OH 44652.

Belle Klipsch all horn speakers, walnut, three years old, over $2,800 new, consider splitting for use as center speaker between Klipschorns, sale for $1,950; Thorens TD147 turntable with Shure V15V cartridge, $325; new Electro Voice SM120A midrange horn with 1824M driver, list $280, take $160. All mint condition with original boxes to ship in. Michael Beasley, 332 South Calderwood, Alcoa, TN 37701, (615) 984-0748 or (615) 983- 6414 evenings.

Several pairs of Focal T90K, T120K and 5K013L drivers in new condition. Used only in testing and development of the ARIA 5 and 7 loudspeakers. Will sell at wholesale cost. Joe D'Appolito, 9 W. Knoll Rd., Andover, MA 01810, (508) 470-1346.

SoundLab Dynastat panels with transformers, HV supplies, $650; Gold Ribbon panels, $650; Kepco 400-B lab supply (HV), $75; Scott 330 AM/FM tuner, $60; EPI LS-841 auto speaker, $185. Trades for Ho trains. Ron (201) 543-2971, 7-9 p.m. EST.

Four Dynaudio D-76 mids, $50 each; two pairs of Dynaudio 30W5Y woofers, $120/pair; Dynaudio D28 tweeters, $50/pair; Dynaudio D21 tweeters, $50/pair; two pairs of Swan 305 woofers, $70/pair; four Eclipse 10" woofers, $35 each; JBL 2235H 15" woofers, $200/pair. Bernardo Diaz, (415) 930-9082.

TEAC 2340 4-channel reel-to-reel and model 2 mixer, $400; pair of KEF T33A tweeters, brand new, $45; pair of Eton 4/203-25 midbass drivers, original boxes, $110. Ron Meyer, 1508 Cedar Grove Plantation Dr., Savannah, GA 31419. (912) 356-6193 weekdays or 925-2311 evenings/weekends.

Swan IV satellites (SB 4,5/88) very solidly built (1¼ MDF) SCR electronic parts, crossover mounted outside on the rear baffle in a bi-wire arrangement plus pedal coupler electronic crossover for the Swans, modified, will not separate, $600/all; Kenwood L-09M monoblocks, 300W each, very good condition, $750. Eduard (213) 395-5196.

Subwoofer with nice furniture finish, doubles as bench or coffee table, local only NYC area, $75; pair of Jordan 5" aluminum cone speakers, very rare, $200; Celestion HF 1300 pressure tweeters $25/pair; VIFA C17WH 6½" speakers, $30/pair; Dyna FM-5, $50. Jerry (212) 364-3485.

Lazarus H-1 amp, $850; JBL 2441s, $995/pair; JBL 2380A horns, $250/pair; TAS 1601 15" woofers, $300/pair; JBL N2400 crossovers, $25; Fostek FT-90H Alnico super tweeters, $125/pair; EV T-350, mint, new, $225/pair; Techniques 5HH10, $75/pair. Steve, (203) 397-3886.

TAD 1" compression drivers, 500Hz-22kHz, pure beryllium diaphragms, EC, in original boxes, engineering notes plus planewave tube response included, $360/pair. Dave, (614) 292-5323 8 a.m.—4 p.m. EST.

Community Light and Sound CB594 bass horns can be used with 15 or 18" driver, available with or without drivers, ElectroVoice DL15X professional woofer, 400W, 8Ω, perfect match for above horns, full specs on both available. Have four of each. Don, (609) 665-6512 evenings.

McIntosh MC-2505 50W/channel, C-28 with cabinets, owners manuals, service manuals, and original warranty certificates, $700 or best offer; Kenwood KT700 tuner, $20; all in excellent condition. Dave, home (317) 883-7575, work (317) 451-9381.

Altec 816 horn loaded cabinets, small VT type; 511 horns; 806 drivers, ER-15 woofers, 415A woofers, 604E coaxial speakers, JBL 3115 crossovers, 2425 horn drivers, constant directivity horns, 2 pair for JBL 1" and 2" throats, Marantz 170-DC power amp, Eumig preamp, Soundcraftsman AE2000 analyzer/equalizer. David, (914) 688-5024.

Marantz 10B, $1,250; 8B, $750; SLT-12, $200; 7T, $200; ARC EC-21, $375; Berning TF-10, $750; EA-230, $400; SAE MK1B MK3CM, $450; Magnepan tympani three speakers, $600; PS Audio phono line preamp model 3, $200; Heathkit WM-5, $75; ARC RMV, $100. All A-OK. Lee Sulka, 7233 Tuolumne, Goleta, CA 93117, (805) 685-8130.

Raw speakers: Peerless 831715-12" woofer, ultra-linear CC-line unit, poly 4 layer vc, Fa = 18Hz, $70/pair; MB MCD19e ¾" metal dome, used by high end manufacturers, $40/pair; Audax HD12.9D25 famous 1" dome, $10/pair; Hi-Tech W85 8" woofer, poly, 40oz magnet, 1-1½ aluminum vc, Fa = 34Hz, robust, clean, unused, $20/pair; Piezos: #1016 2x5 horn, $7/pair; dual element line source tweeter, $7/pair; Single drivers: Dynaudio D21, Infinity modified Philips AD5061, Philips ¾" dome tweeter, all 3 for $7, or free with any 3 items ordered above. All units in perfect operating order. MO or check (personal must clear) to: Mike Wayne, 3541 North Overhill, Chicago, IL 60634. (312) 777-7000, ext. 4189 days, (312) 589-2976 (home).

Superphone Rev 2 preamp, almost new, $400; Hafler DH-100 preamps, black, new in box, $170; Dalesford D153 6½ woofers, new, $50/pair; AQ sorb mat, new, $20; everything must go, make offers! Shipping on preamps paid. Walter Fleming, 182 Plain Hill Rd., Norwich, CT 06360, (203) 889-5018, 5pm best.

Three *Speaker Builder* and six *Audio Amateur* binders, as new, $6/each or $50 for all. Peter Bandurian, (303) 666-7588, evenings.

PAS, $65; FM3, $50; with manual and box, $95; Pioneer TV9500 UHF TV tuner, $50; pair of original AR1 woofers, pair of AR1 crossovers. Joe Jennings, 4903A South Orchard, Tacoma, WA 98467, (206) 473-3572.

CLUBS

PIEDMONT AUDIO SOCIETY Audio club in the Raleigh-Durham-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.

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.

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

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.

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

Continued on page 102

Exclusively from
Watters Sound Wave Co(214)991-6994
4320 Spring Valley,Dallas TX 75244
Orca,Dsgn & Mfg Corp.(818)707-1629
1531 Lookout Dr. Agoura CA 91301

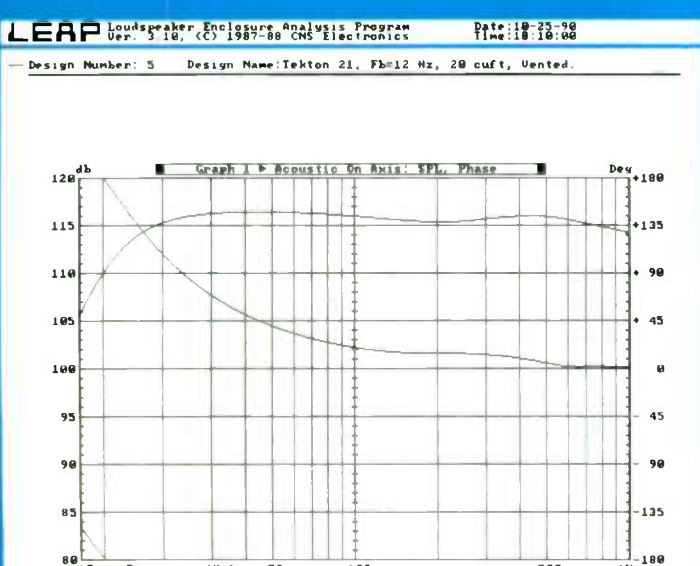
TEKTON 21
6 in. diam VC

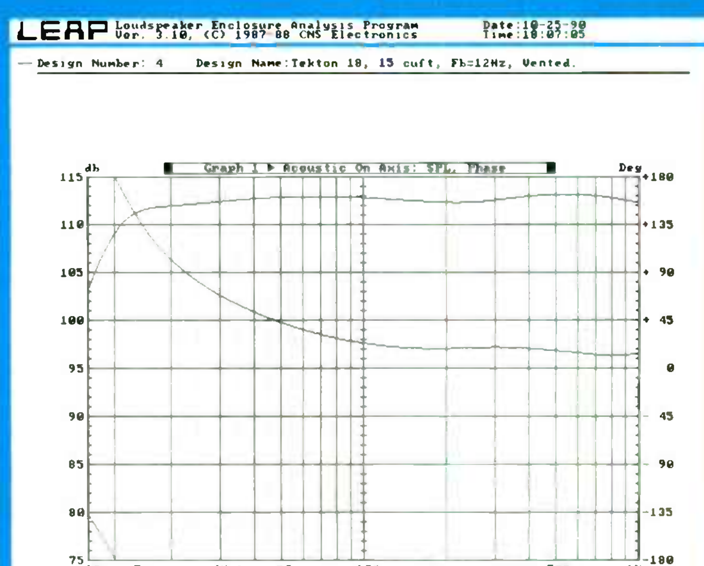
The TEKTON Infra-bass drivers are uniquely featured to offer true extension below audibility. Uncompromising design in collaboration with Kimon Bellas called for machined steel motor assemblies, massive cast frame and immense magnet structure.

High power (over 1.2 KW at 20 Hz) and long throw (close to 1 in. each way) guarantee the deepest and tightest bass available anywhere today.

TEKTON 18
5 in. diam VC

Tm to Orca Dsgn & Mfg Corp.





Continued from page 100

SOUTHEASTERN MICHIGAN WOOFER AND TWEETER MARCHING SOCIETY (SMWTMS). Detroit area audio construction club. Meetings every two months featuring serious lectures, design analyses, digital audio, A B listening tests, equipment clinics, recording studio visits, and audio fun. The club journal is *LC, The* SMWTMS Network. Corresponding member's subscription available. Call (313) 477-6502 (days) or write David Carlstrom, SMWTMS, PO Box 721464, Berkley, MI 48072-0464.

THE ATLANTA AUDIO SOCIETY is dedicated to furnish 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 Denny Meeker, (404) 872-0428, or write: PO Box 361, Marietta, GA 30061.

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 an SASE to: Barry Waldron, 1847 Country Club Dr., Placerville, CA 95667.

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.

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 Dick Bowman, 081 520 6334.

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

NEW JERSEY AUDIO SOCIETY meets monthly. Emphasis is on construction and modification of electronics and speakers. Dues includes monthly newsletter with high-end news, construction articles, analysis of commercial circuits, etc. Meetings are devoted to listening to records and CDs, comparing and A-B-ing equipment. New members welcome. Contact Bill Donnally, (201) 334-9412, RD2, Box 69D, Miller Dr., Boonton, NJ 07005; or contact Bob Young, (201) 381-6269.

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 three thousand "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.

PACIFICNORTHWEST AUDIO SOCIETY (PAS) consists of 60 audio enthusiasts meeting monthly, second Wednesdays, 7:30 to 9:30 p.m. at 4545 Island Crest Way, Mercer Island, Washington. Be our guest, write Box 435, Mercer Island, WA 98040 or call Bob McDonald, (206) 232-8130.

AUDIOPHILES IN THE DAYTON/SPRINGFIELD, OHIO AREA: We are forming an audio club. Please contact me if you're interested in construction, modifications, testing, recording or just plain listening to music. Ken Beers, 1756 Hilt Rd., Yellow Springs, OH 45387, (513) 767-1457.

AUDIO SOCIETY OF MINNESOTA. Audiophiles, music lovers, scratch builders, record collectors, tube freaks, digital freaks — we've got 'em all! Monthly meeting, tours, audiophile concerts, special guests, etc. Now in our 13th consecutive year! Write ASM, PO Box 32293, Fridley, MN 55432.

THE BOSTON AUDIO SOCIETY invites you to join and receive the bi-monthly *B.A.S. SPEAKER* with reviews, debates, scientific analyses, and summaries of lectures by major engineers. Read about Apogee, Nytal, Conrad-Johnson, dbx digital, Snell, music criticism and other topics. Rates on request. PO Box 211, Boston, MA 02126.

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. See you soon!

CHICAGO AREA ENTHUSIASTS WANTED for audio construction club. Call Tom, (312) 558-3377 or (708) 516-0170 evenings for details.

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 the city of Alexandria, Va. Prospective members are welcome but must register in advance in order to be admitted to the meetings. No exceptions please. If interested please call Horace Vignale, (703) 578-4929.

THE WESTERN NEW YORK Audio Society (WNY Audio Society) is an active and growing audio club located in the Buffalo area. We issue a quarterly newsletter and hold meetings the first Tuesday of every month. Our meetings have attracted many local and distant manufacturers of audio related equipment. We are involved in all facets of audio—from building to purchasing at discount prices. For a copy of our current newsletter and information regarding our society, please write to M.A. Monaco, WNY Audio Society, PO Box 312, N. Tonawanda, NY 14120.

Advertising Index

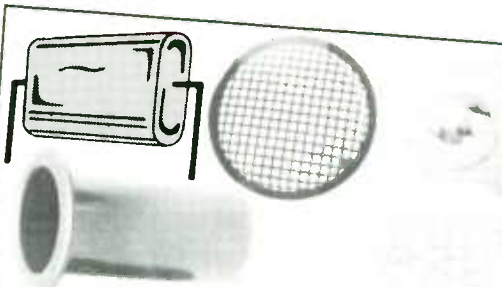
AUDIOPHILE
W I S H L I S T

POLYDAX MDA 100

New diecast frame midrange from Polydax. TPX cone, Norsorex surround and phasing plug. Supra magnet technology. Pure titanium voice coil former. 8 ohm, fs = 215 Hz, response: 450-6,500 Hz. 12.4 oz. magnet. 60 watts nominal. SPL = 92.5 dB 1W/1M. 4" square

$66.00 Each

#SA-280-235



WE STOCK A FULL LINE OF

TUNED PORT TUBES, GRILLS AND CLOTH, CHOKES, MYLARS, CABINET FEET, CORNERS, FOAM, TERMINALS, AND MORE...

Wish List

- ☑ *Polydax Titaniums*
- ☑ *Pyle woofers*
- ☑ *Phillips soft domes*
- ☑ *Pioneer polys*
- ☑ *Eminence M.I.'s*
- ☑ *Thruster drivers*

EMINENCE 12" POLYPROPYLENE

High quality 12" woofer made in the U.S.A. Polypropylene cone with poly foam surround. 2-1/2" voice coil. 4 ohm (3.69 DCR) impedance. 56 oz. magnet. Response: 24-3,000 Hz. fs = 28.5 Hz. SPL = 92 dB 1W/1M. 150 watts RMS. Qts = .48. VAS = 9.15 cu.ft.

#SA-290-187 — Made in U.S.A. — $54.50 (1-3) — $49.80 (4-up)

POLYDAX TITANIUM TWEETER

Titanium is deposited on a polymer dome to combine the advantages of both hard and soft dome technologies. 8 ohm. Ferro fluid cooled voice coil. fs = 1200 Hz. SPL = 90 dB 1W/1M. 50 watts RMS. 4" round. Polydax #DTW100TI25.

#SA-270-047 — $27.50 Each

PHILIPS SOFT DOME PAIR

One of Philips' most popular tweeters. 1" polycarbonate dome with 1" ferro fluid cooled voice coil. 8 ohm. fs = 1,600 Hz. Response: 2,000 - 20,000 Hz. SPL = 92 dB 1W/1M. 10 oz. magnet. 20 watts RMS. Philips #PKAD01624T8FF. 3-3/4" dia. 2-7/8" hole.

#SA-270-220 — $32.80 Pair

PARTS EXPRESS NOW STOCKS PYLE DRIVERS!

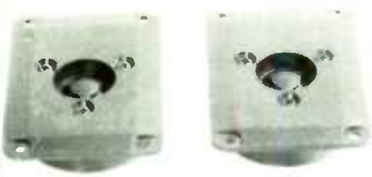
PYLE

Made in U.S.A.

12" LONG THROW WOOFER

Resonant frequency 20 Hz. 2-1/2" voice coil. 70 oz. magnet. 105 watts RMS. Response: 20 - 3,000 Hz. SPL = 97 dB 1W/1M. 8 ohm. Made in the U.S.A.

#SA-292-050 — $62.90 Each



1" SOFT DOME PAIR

High temperature, poly thermal, ferro fluid cooled voice coil. Response: 4,000 - 20,000 Hz. SPL = 90 dB 1W/1M. 8 oz. magnet. 40 watts RMS. 8 ohm. Boxed in pairs. Made in the U.S.A.

#SA-272-010 — $35.50 Pair

12 GA. SPEAKER WIRE

Rope lay construction. Over 250 strands of 36 ga. wire. Extra thick, yet flexible insulation. Made in the U.S.A. (Connectors available).

#SA-100-151 — 39¢ Foot

GOLD PLATED CONNECTORS

We stock a full line of gold plated connectors, speaker pin terminals and spades, RCA and "F", and many cables and adaptors. Send for our full line catalog below.

FREE 148 PAGE CATALOG

We stock one of the largest selections of speaker drivers in the country. Send for your FREE copy of our catalog today.

Parts Express

0 E. First St., Dayton, Ohio 45402
cal: 1-513-222-0173
AX: 1-513-222-4644

CALL TOLL FREE
1-800-338-0531

· 15 day money back guarantee · $15.00 minimum order · We accept Mastercard, Visa, Discover, and C.O.D. orders · 24 hour shipping · Shipping charge = UPS chart rate + $1.00 ($3.00 minimum Charge) · Hours 8:30 am - 7:00 pm EST, Monday - Friday, 9:00 am - 2:00 pm EST, Saturday · Mail order customers, please call for shipping estimate on orders exceeding 5 lbs. · Foreign customers please send $5.00 U.S. funds to receive catalog