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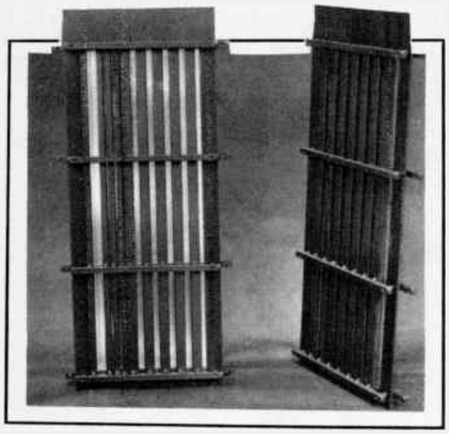
Speaker Builder

THE LOUDSPEAKER JOURNAL

Good News



WHALE COVE AUDIO has been appointed the distributor for Swan's Speaker Systems products. Speaker builders may obtain all products and materials necessary to build the Swan IV loudspeaker system directly from Whale Cove Audio.

The complete kit comprises two Symmetrical Satellites and two Symmetrical Bass enclosures, completely assembled of unpainted 1-inch fiberboard with rounded edges; a grille kit; the Pedal Coupler crossover/equalizer; the Treble Coupler passive crossovers; all drivers, wire, terminals, sealant, and hardware; a reprint of the article from *Speaker Builder*; and further suggestions and instructions.

All kit products are also available separately, including an oak enclosure set, assembled of unfinished 1-inch oak veneered fiberboard with all edges inset with rounded solid oak. The grille kit features removable aluminum frames for each enclosure and charcoal brown foam, with a design that minimizes diffraction effects and high frequency degradation.

The Swan IV active crossover/equalizer uses quality components. An alternative, the Pedal Coupler II is an upgraded rack mount crossover/equalizer designed by Joseph Curcio, using high-quality circuitry and components, and includes a CD input with a level control and an absolute phase reversal switch.

The complete kit is $1,865; add $270 for the oak enclosures, $265 for the Pedal Coupler II. Separate prices are: oak enclosure set, $1,149; fiberboard enclosure set, $879; grille kit, $78; Pedal Coupler, $230; and Pedal Coupler II, $495. Prices do not include shipping.

For information or to purchase products contact Elizabeth Pease Day, Whale Cove Audio, PO Box 356, Lighthouse Rd., Swan's Island, ME 04685, (207) 526-4343.

Fast Reply #FD227

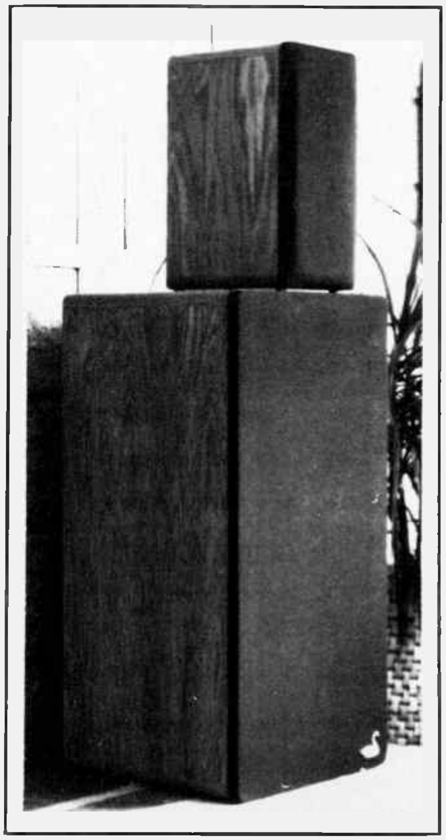
EMINENT TECHNOLOGY now has individual planar magnetic panels for speaker builders, using its linear field transducer design.

The panels are available in two versions. Multiple low frequency panels can operate from below 30Hz to 7kHz; a high-frequency unit's range is from about 60Hz to 20kHz. The radiation pattern of both units is dipolar and phase response is generally ±20° over the operating range, except at the panel's low frequency resonance. Each panel contains 15Ω elements that can be configured to achieve the desired impedance and frequency range grouping across the diaphragm. Each panel has a 7- by 21-inch radiating area and weighs 12 lbs. Multiple diaphragm resonance frequencies are available when more than one panel is used in a system.

These will be sold through specified dealers or direct from the factory; prices per panel are: $280, low-frequency; $300 high-frequency model.

Contact Eminent Technology for an applications manual: 225 E. Palmer St., Tallahassee, FL 32301, (904) 224-5999.

Fast Reply #FD1124



ORIVATION, creators of SchematicMaster software, announces a new phone number, (408) 292-9639, for anyone wishing to speak to David O'Riva concerning this product. An updated version is in progress. For the Orivation Bulletin Board, the number is still (408) 259-2223. For information: Orivation, 2726 Hostetter Rd., San Jose, CA 95132.

Fast Reply #FD228

A new planar loudspeaker highlighted **SUMO**'s exhibit at WCES 1989. The result of more than four years of development, Aria utilizes a single stretched Mylar diaphragm, rather than multiple cone drivers, to radiate sound. The result is an open, transparent quality, and coherent sonic presentation, achieved without crossover network components, equalization, or associated electronics of any kind in the signal path.

Aria is a floor standing loudspeaker, whose operating principle is a drive applied at the center of Aria's diaphragm, which creates a spherical waveform, and results in an accurate soundstage and a broad listening area.

Aria has a retail price of $3,000 a pair.

Andromeda II, SUMO's newest entry in high-end amplification, has a power rating of 200W per channel and peak available current of over 200A. This level of performance ensures absolute stability into demanding and difficult loudspeaker loads. Andromeda II is a low negative feedback, totally balanced amplifier that utilizes proprietary output linearizing circuitry to reduce distortion without intruding on sound quality. Retail pricing will be $1,499.

Great American Sound (GAS), the progenitor of products such as Ampzilla, Son of Ampzilla, Thaedra, and Sleeping Beauty, has been purchased by SUMO's parent, Califone International Inc., and is being structured to operate as a sister company to SUMO. GAS former owners Ed Miller and Adam Zaremba will actively work as part of the venture.

For information contact: SUMO Products Group, 21300 Superior St., Chatsworth, CA 91311, (818) 407-2426; TLX 658234 SUMO CORP WKVG.

Fast Reply #FD180

West Coast audio/video dealers and top manufacturers of high-end equipment, such as Krell, Infinity and Threshold, will be exhibiting at the ***STEREOPHILE*** San Francisco Bay Area High End Hi-Fi Show, April 21–23, at the Dunfey Hotel in San Mateo. In addition to the equipment demonstration rooms, live jazz, gospel and classical music will be featured.

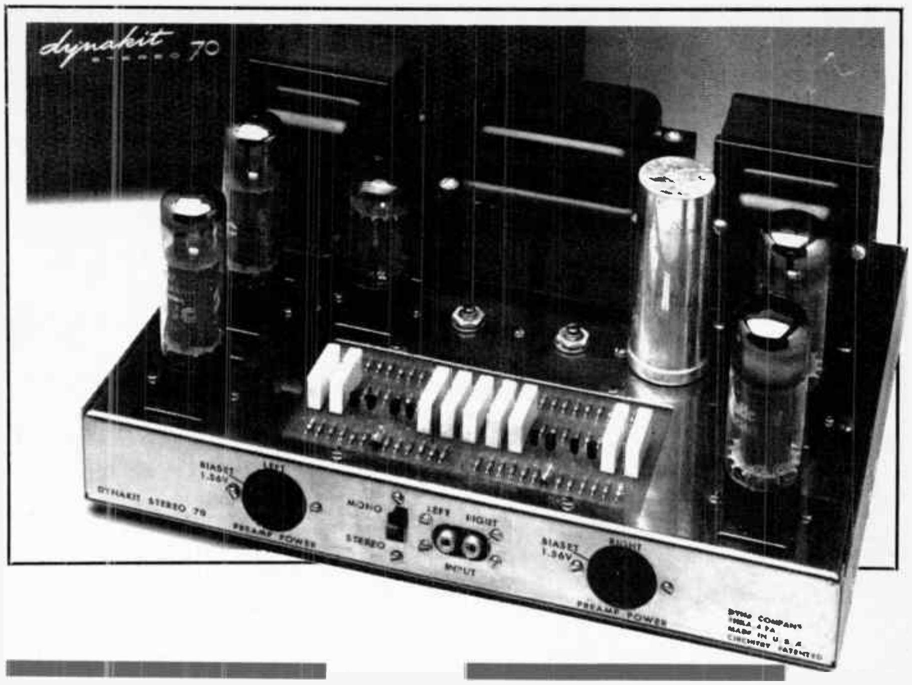
The show is open to the public; tickets, good for all three days, are $15. Contact Richard Ziff Communications, (516) 221-5865 or (212) 661-5300 for information. *Stereophile*, PO Box 5529, Santa Fe, NM 87502.

New from **CONTACT EAST**, the Service Repair Adhesive Tool Kit contains five types of "high-tech" bonding compounds for plastic, glass, metal, ceramic and rubber; plus threadlocking adhesive and a type made for adhering wire to circuit board surfaces. The kit comes in a compact molded case.

Other new products include analog/digital oscilloscopes, static protection products, soldering stations and supplies, test equipment and precision tools.

For information and the 1989 catalog, write or call: Contact East, PO Box 786, 335 Willow St. S., North Andover, MA 01845, (508) 682-2000.

Fast Reply #FD500



OPTION AUDIO, an affiliate of EJ Jordan USA, announces that its affordable, world-class bookshelf monitors, the Syntax 1-A and .5-A series, will soon be available in preassembled and kit form. Please include a self-addressed stamped envelope marked "Syntax" for all requests.

For information, EJ Jordan USA/Option Audio has a new address: PO Box 1411, Wall, NJ 07719, (201) 280-9427.

Fast Reply #FD61

The Hybrid 70 driver board, which updates the Dynaco Stereo 70 vacuum tube power amp to the latest standards, is available from **SUTHERLAND ENGINEERING**. This replaces the driver board with a pure Class A JFET/MOSFET cascode design to drive the tube output stage. Precision metal film resistors and noninductive film capacitors are used throughout.

The Hybrid 70 is factory-assembled and tested and comes with instructions for simple installation: price is $80, plus $5 shipping and handling, from Sutherland Engineering, PO Box 1363, Lawrence, KS 66044, (913) 841-3355.

Fast Reply #FD192

IQS's Bill Elder and Doug Evered are still applying signal analysis technology to audio applications, with new products in that category. The 402 is a successor to the IQS model 401 FFT analyzer. The Series 402 is a software/board-level hardware add-on system for an IBM-PC/XT or AT compatible computer, which provides signal analysis capabilities for acoustics, circuit analysis, vibration, audio, speech, biomedical and educational applications.

The 402 features powerful graphics-oriented MLSSA and Signal Analyst software packages that control hardware and together provide analytical functions that include: FFT, transfer function, impulse response, waterfall (cumulative spectra), PSD, THD, ⅓-octave, reverberation time, energy-time curves, and more. Spectral resolution can be programmed to approach 16,000 lines while delivering a magnitude dynamic range in excess of 80dB. Built-in tunable anti-aliasing filters and input gain are fully software programmable. Software supports all popular color and monochrome graphic modes up to VGA resolution.

The system also includes built-in signal source with which to stimulate systems under test with software-programmable amplitude control. This source generates MLS (maximum-length-sequence), impulse, and step waveforms.

Graphic output to HP-GL compatible single and multi-pen plotters as well as standard laser, ink-jet and dot-matrix printers are supported.

The IQS model 416-SSM-2-xx signal source provides high-resolution arbitrary waveforms for either stand-alone or integrated stimulus/acquisition applications. Features include on-board memory that provides waveform depths up to 65,536 points while delivering 16-bit resolution and 12-bit resolution amplitude programmability. Waveforms are easily loaded from disk and controlled by WaveSource-16, Signal Analyst or custom-created software. Signal Analyst software (included in model 402 and 416 signal analyzers) provides ability to record, store/retrieve to/from disk, and play back waveforms. The optional Sweep Module provides linear/log frequency and amplitude sweep modes for applications requiring slow sweep rates (less than 10SPS).

The model 416-SSM-2 is available stock to 30 days; price is $2,590, and $3,990 with optional module. The model 402 Signal/Spectrum Analyzer is available stock to 14 days; price is $2,495. All inquiries: IQS Inc., 12862-F Garden Grove Blvd., Garden Grove, CA 92643, (714) 539-7842.

Fast Reply #FD178

A&S SPEAKERS announces the winners of its first Audiophile Sound-Off held at its showroom in San Francisco in January: a floor-standing system using Joe D'Appolito's theories with two 7-inch Focal drivers and a Dynaudio tweeter, designed by Mark Carter, Grand Forks, ND; and a two-way design using an 8-inch Eton Kevlar cone and Morel tweeter in a Sonotube enclosure, by Wendell Robbins, Yorktown Heights, NY.

The winners were selected by three judges: Betsy Cohen, audio consultant and Stanford professor; Vance Dickason, editor of *Voice Coil* and author of *The Loudspeaker Design Cookbook*; and Tomlinson Holman, technical director of Lucasfilm. Builders are invited to begin work for the 1990 Audiophile Sound-Off.

Both designs will be exhibited at the *Stereophile* Show and will be available through A&S Speakers, 3170 23rd St., San Francisco, CA 94110, (415) 641-4573.

Fast Reply #FD572

The **Audio Engineering Society** has published *Time Delay Spectrometry*, an anthology of Richard Heyser's works on measurements, analysis and perception, from various publications including the *JAES*, *Audio* and *IREE*.

Editor John Prohs notes the anthology is a memorial to Heyser's work, but is also fundamental material for future developments in audio. The articles are presented chronologically, beginning with "Acoustical Measurements by Time Delay Spectrometry" (*JAES*, 1967). A bibliography and list of related articles by other authors is included.

This 280-page volume is $27 for AES members, $30 nonmembers. To order send payment to: Audio Engineering Society, 60 E. 42nd St., Rm. 2520, New York, NY 10165-0075.

SIDEREAL AKUSTIC has reorganized and relocated and is now shipping its audio-grade capacitor again. These metallized polypropylene types are noted for low ESR, minimum self-inductance and high mechanical stability for freedom from "singing" and microphonic effects.

For more information on the Sidereal-Kap, write or call Richard Smith, Sidereal Akustic, 9974 Scripps Ranch Blvd. #120, San Diego, CA 92131; (619) 578-4226; FAX, (619) 578-4059.

Fast Reply #FD778

Due to a postal error, any mail prior to January 1, 1989 addressed to **DELAC** (Delaware Acoustics), manufacturers of the S10 loudspeaker (*SB* 4/88, p. 5), may have been returned to the sender. Please resubmit any correspondence to the same address: DELAC, PO Box 54, Newark, DE 19711.

SOUND-NET is a new electronic bulletin board information system that can be accessed from any personal computer equipped with a 1200 or 2400 baud modem and communications software, or through the university electronic mail network UUCP.

The system offers industry news, new products and services, tender notifications and business-to-business news. On-line conferences offer a forum for informal debate on such topics as audio, electronics, acoustics, CAD and mathematics. Direct communication with manufacturers and distributors is possible and public domain software can be downloaded from the system.

Sound-Net is nonprofit and sponsored by the Audio Engineering Society. Membership fee for nonmembers is $45; AES members, $35; write Sound-Net, 164 Sunnyside Ave., Suite 100, Toronto, Ontario M6R 2P6, Canada. For more information call Paul Gonsalves at (416) 530-4423 or contact him via UUCP address: dciem!sndnet!paul.

Fast Reply #FD230

SPEAKER BUILDER

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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) **$20**, two years (twelve issues) **$35**. All sets of back issues are available beginning with 1980. Caribbean and Canada add **$4** per year for postage. Overseas rates available on request. Subscribers residing the Western Hemisphere are served by air.

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A Note To Contributors

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

Speaker Builder (US ISSN 0199-7920) is published bi-monthly at $20 per year; $35 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: If undeliverable send to **SPEAKER BUILDER**, PO Box 494, Peterborough, NH 03458.

Fast Reply #FD29

About This Issue

Georgia Tech may not be nationally known as a hotbed of loudspeaker design activity—*yet*, but it may be soon. **Marshall Leach** has made loudspeakers a popular part of the curriculum in Tech's engineering department and a fair sample of the work is available beginning on page 10.

Tom Nousaine is not only our guest editorialist this time but shares his adventures with a passively assisted woofer project beginning on page 16. **Fernando Garcia Viesca** introduces us to the world of digital filters as the audio world daily becomes more digital in character (page 19).

Doug Rauer shows us how he determined the correct arc for his driver array and provides some software to enable us to design arcs for our systems (page 26). Electronic crossovers are nice but they necessitate adding more power amps. **Ralph Gonzalez,** faced with this problem, chose a quick way to combine an active crossover with some minimalist power with surprisingly satisfying results.

If you've wanted to add "surround sound" **Lou Edelman** has advice on how to implement a simple exploration (page 36). **Paul Dwight** was uncertain whether spikes would help his speakers so rather than invest in costly hardware, he fashioned his own and shows you how he did it (page 44).

Contributing Editor **Gary Galo** reviews two disks featuring the Colossus ambience idea as well as two fine CD test discs from France (page 49). **Peter Muxlow** (page 54) surveys ribbon drivers this time. And don't miss our resident **curmudgeon**'s truth table on page 78.

Speaker Builder
THE LOUDSPEAKER JOURNAL

VOLUME 10 NUMBER 2 MARCH 1989

FEATURES

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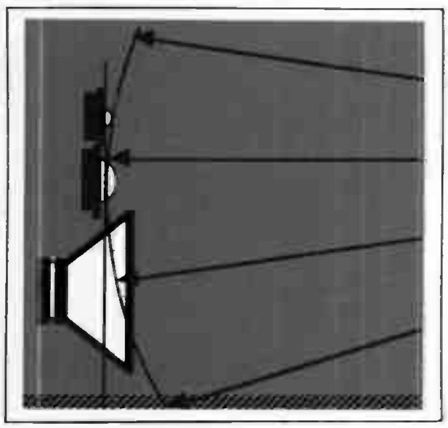

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DEPARTMENTS

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

Julian Hirsch – Stereo Review, Sept. '88

Cambridge SoundWorks has created Ensemble,™ a speaker system that can provide the sound once reserved for the best speakers under laboratory conditions. It virtually disappears in your room. And because we market it directly, Ensemble costs far less than previous all-out designs.

Henry Kloss, creator of the dominant speaker models of the '50s (Acoustic Research), '60s (KLH), and '70s (Advent), brings you Ensemble, a genuinely new kind of speaker system for the '90s, available factory direct from Cambridge SoundWorks.

The best sound comes in four small packages.

Ensemble consists of four speaker units. Two compact low-frequency speakers reproduce the deep bass, while two small satellite units reproduce the rest of the music, making it possible to reproduce just the right amount of energy in each part of the musical range without turning your listening room into a stereo showroom.

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

No matter how well a speaker performs, at home the listening room takes over. If you put a conventional speaker where the room can help the low bass, it may hinder the upper ranges, or vice-versa.

What Henry Kloss tells his friends:

Every time I came out with a new speaker at AR, KLH, or Advent, my friends would ask me, "Henry, is it worth the extra money for me to trade up?" And every time I would answer, "No, what you've already got is still good enough."

But today, with the introduction of Ensemble, I tell them, "Perhaps now is the time to give your old speakers to the children."

Ensemble is a Trademark of Cambridge SoundWorks, Inc.

Ensemble, on the other hand, *takes advantage* of your room's acoustics. The ear can't tell where bass comes from, which is why Ensemble's bass units can be tucked out of the way–on the floor, atop bookshelves, or under furniture. The satellites can be hung directly on the wall, or placed on windowsills or shelves. No bulky speakers dominate your living space, yet Ensemble reproduces the deep bass that *no* mini speakers can.

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

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

Unlike satellite systems which use a single large subwoofer, Ensemble features separate compact bass units for each stereo channel. They fit more gracefully into your living environment, and help minimize the effects of the listening room's standing waves.

cabinets ruggedly constructed for proper acoustical performance. We even gold-plate all connectors to prevent corrosion. An even bigger difference is how we sell it...

The best showroom of all: your living room.

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

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

Call 1-800-AKA-HIFI* (1-800-252-4434)

Our toll-free number will connect you to a Cambridge SoundWorks audio expert. He or she will answer all your questions, take your order and arrange surface shipment via UPS. Your Cambridge SoundWorks audio expert will continue as your personal contact with us. We think you'll like this new way of doing business.

*In Canada, call 1-800-525-4434. Audio experts are on duty Mon.-Fri., 9AM–10PM, Sat., Sun., 9AM–6PM Eastern Time.

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A Peal for Bell

A GUEST EDITORIAL

by Tom Nousaine

Mr. Dell's editorial, "A Modest Proposal" (*SB* 1/89), suggests that the Bell Telephone System was "the best of its kind," but was "more expensive than it needed to be." Well, he's half right. It was, and most likely still is, the best in the world. I agree that it was better before divestiture, but I must disagree with the statement about the price. In 1987, the average cost of unlimited local telephone service in the US was $12.33 a month; about 40 cents a day. Add access charges and it's still only about 50 cents a day—less than people pay for a cup of coffee, a candy bar or to have the daily paper delivered.

And it has become more affordable over the years. Adjusting for inflation, we find that telephone service costs only half as much as it did in 1960. By any imaginable standard, local and long distance telephone service is incredibly inexpensive. So much so that nearly every household has one, and those that don't have a phone can use one any time they wish for a quarter—unless it's a life or death emergency and then it could be free. The primary reason for this value has been a long parade of technology, invention and innovation. These technological advances improved the telephone system and, coincidentally, shaped audio as we know it.

An examination of something as novel as the compact disc format's use of pulse-code modulation (PCM) can be traced back to the work of Harry Nyquist, an AT&T engineer who published most of his work prior to 1930. PCM itself was first used by American citizens in 1962 when telephone companies began installing D1 digital carrier systems.

The CD player employs lasers. The first working laser was demonstrated at Bell Labs in 1958, a precursor to fiber-optic transmission systems in the mid-seventies.

Moving further back in history, we can note that stereophonic sound was first demonstrated at the Labs in 1933. Of course, practically all analog circuitry employs negative feedback, first perfected by H. Black at Bell Labs in 1927. Don't forget solid-state electronics—1947, Bell Labs. In 1952, transistors were installed in the telephone network for the same reasons we use them today in audio—improved performance, lower cost, smaller size, less heat, improved reliability and so on and so on.

Computers? The first Stored Program Control Electronics Switch was installed in Illinois in 1962. The first time many ordinary citizens used a computer was in making telephone calls. "User friendly?" Can you think of a more friendly device than a telephone keypad? On the other end is the most sophisticated electronic network in the world, yet six-year-olds have little trouble using it.

From time to time I find it useful to step back and examine the historical context of our lives. Over the years the telephone system has become such an integral part of our daily routine it is easy to take for granted. As audiophiles, we shouldn't overlook the contribution the telephone industry has made to our avocation. If you think about voice transmission as a basic audio task, you easily see how improving the telephone system dovetails with improving audio reproduction in general.

In fact, I see audio constructors faced with exactly the same equation as telephone network designers. As the advertisements state, "If cost is no object, anybody can build a great speaker." That may be true, but the trick is to build a better speaker for a reasonable outlay. That's what Nyquist and associates have done for us with telephones—great system, low cost. And their technological spinoffs help us build great sounding systems, better than the richest can buy, at reasonable cost.

Coming Next Time

Put an Isobarik into your Thunderbird, updating the Klipsch Cornwall, computer-ready speakers, and a protection switch to keep your system safe. If picket fencing seems unpromising as enclosure material—prepare for a surprise. Mail date: May 26.—E.T.D.

THE AUDIO LABORATORY LOUDSPEAKER SYSTEM

BY W. MARSHALL LEACH, JR.

PHOTO 1: These loudspeakers were constructed of fiberboard according to the author's plans, but the front panel is recessed, the top panel is slightly wider, the port is below the woofer and the base is larger. The drivers are those recommended in the parts list.

By far, the most popular construction projects among Georgia Tech students with whom I have come in contact have been loudspeaker systems—ranging in size from a shoebox to that approaching a telephone booth—infinite baffle, acoustic suspension, vented-box, transmission line and electrostatic systems.

Of all these, my favorite is the vented-box system described here. It is fairly compact, has an extremely smooth and balanced frequency response, is highly efficient for a home system, and can be built for $300–400 a pair.

Building the system has become so popular among students that I have measured the frequency response of more than 100 stereo pairs a year for the past several years. When the end of our academic quarter approaches, so many loudspeakers are brought into the lab to be tested that it is difficult to get through the door.

I constantly get letters and telephone calls for copies of the system plans, which have even found their way onto computer networks. This article presents the plans in much the same form as I give them to students. The emphasis is on construction details rather than theory. I am confident that anyone who builds the system correctly will agree with me that you can build a better loudspeaker system.

BACKGROUND. This vented-box loudspeaker system has been developed over a period of several years in the Georgia Tech Audio Laboratory. Many students have participated in the design. Each time I tested a system in the lab, changes were made when required to improve the performance and the final system design evolved in this way.

The enclosure has an internal volume of 2 ft.^{3} with inside dimension ratios of 0.6:1.0:1.6. These ratios minimize the effects of standing waves inside the box for better frequency response. The bass frequency response is characterized by a fourth-order high-pass transfer function that has a lower half-power cutoff frequency below 40Hz and an electroacoustic efficiency of about 0.8%. The Helmholtz tuning frequency of the port and box is 34Hz. The efficiency is quite high for home systems so that a high-power amplifier is not required. An amplifier rated at about 30W to no more than 100W per channel is adequate.

WOOD MATERIALS. If you are inexperienced at wood construction projects, a reliable woodworking shop that can supply the materials and give assistance in cutting and assembling them is recommended. If you do not need such assistance, many industrial suppliers of construction materials will cut the wood panel according to the cutting diagram in *Fig. 1* so you can take the materials home in a car. All enclosure panels are cut from a single sheet of 4 by 8-foot, ¾-inch industrial grade plywood, fiberboard, or particle board.

Veneer can be purchased at specialty wood stores, but proper veneering techniques are essential for good results. If the veneer is not glued properly, air bubbles can form under it. A method that

ABOUT THE AUTHOR

W. Marshall Leach, Jr., is Professor of Electrical Engineering at Georgia Institute of Technology. He received his BS and MS degrees from the University of South Carolina and his Ph.D. from Georgia Tech, all in electrical engineering. At Georgia Tech, he teaches in the areas of applied electromagnetics and modern electronics. Twice yearly, he teaches a senior elective course, Audio Engineering, which covers loudspeaker and amplifier design. He also supervises the Audio Laboratory at Georgia Tech. His interests are in audio circuit design, electroacoustic design, and writing.

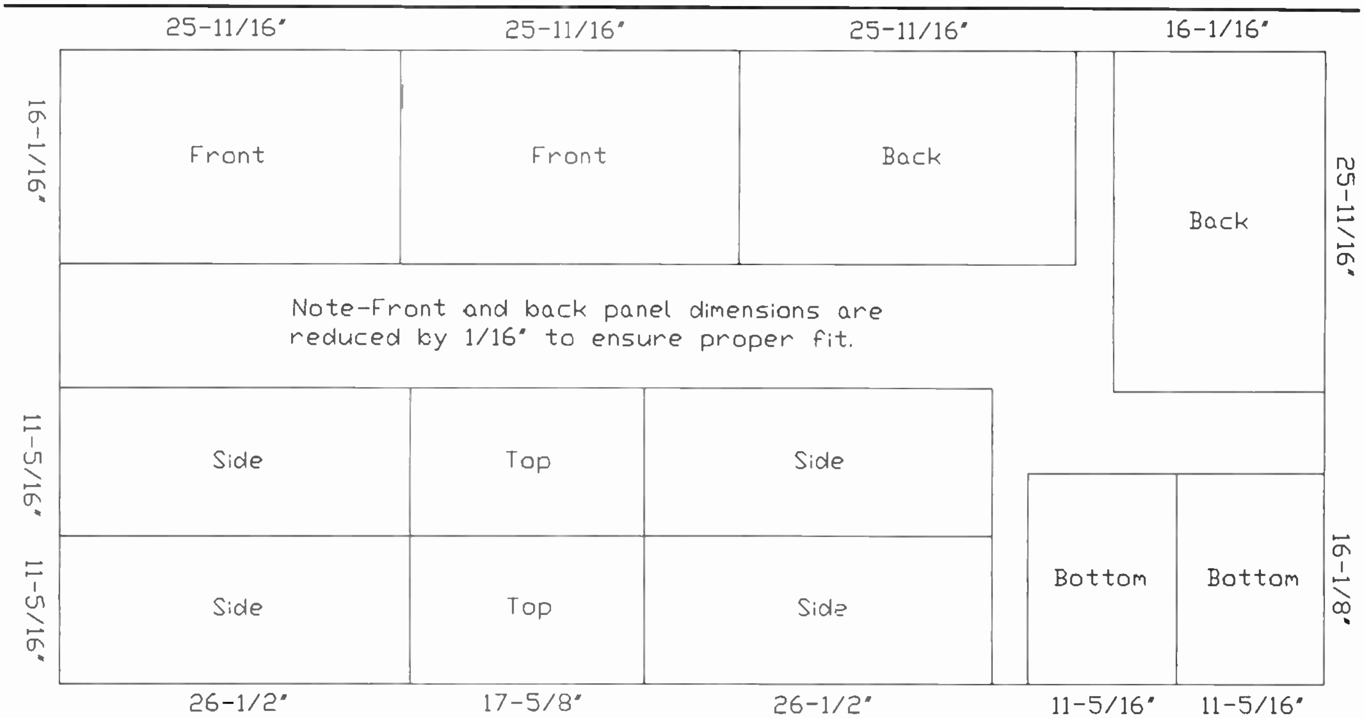


FIGURE 1: Cutting diagram.

WOOD PARTS LIST: TWO ENCLOSURES

Panels	**Dimensions (Inches)**
Fronts and backs (4)	16$\frac{1}{16}$ by 25$\frac{11}{16}$
Sides (4)	11$\frac{5}{16}$ by 26$\frac{1}{2}$
Tops (2)	11$\frac{5}{16}$ by 17$\frac{5}{8}$
Bottoms (2)	11$\frac{5}{16}$ by 16$\frac{1}{8}$
Internal Braces (2 by 2)	
Top-to-bottom (sides)	22$\frac{11}{16}$
Front-to-back (corners)	9$\frac{3}{4}$
Left-to-right (top and bottom)	13$\frac{1}{16}$
Eight of each size required.	
Base (1 by 2)	
Left-to-right	10$\frac{5}{8}$
Front-to-back	8$\frac{5}{16}$
Four of each size required.	

Miscellaneous
Wood glue
Wood screws—1½-inch, #6 or #8 flat head wood screws or dry wall screws
Clear silicone rubber caulk in 10-ounce tubes
Caulking gun
Unfaced fiberglass—1-inch thick
Hookup wire—18- or 20-gauge stranded
Solder-lug terminal strips for crossover network binding posts
Electrical binding posts for rear panel input terminals
Machine screws, flat washers and nuts for mounting drivers
Two 3-inch internal diameter plastic tubes—cut 4 inches long for the ports
Grille frame—rectangular or quarter-round moulding, painted black
Grille cloth—black single knit fabric

will work is to apply yellow carpenter's wood glue with a cloth covered roller to *both* surfaces and allow it to dry for a short time to a "tack dry" state. Apply the veneer to the boxes *before* the glue is completely dry, and use a warm iron to activate the glue. A book on the principles of applying veneer is helpful. *["On Veneering" (Mailbox, SB 1/89, p. 61), describes some basics.—Ed.]*

Industrial grade plywood is recommended for the boxes, braced internally with "two-by-two" fir or pine strips (the cross-sectional dimensions are 1.5 by 1.5 inches). If hardwood or preveneered plywood is used, you must modify the cutting plans so that the joints between the tops and sides can be mitered at a 45° angle with a table or radial-arm saw, and the screws must be installed from inside the box. The rectangular bases for the boxes are cut from "one-by-two" bracing, which measures ¾-inch by 1½ inches wide. *Figure 2* shows the assembly of the boxes. The approximate order of assembly is as follows:

1. Cut panels and bracing;
2. Assemble top, bottom, and sides;
3. Cut holes in front panel for drivers and port;
4. Cut hole in rear panel for electrical terminals;
5. Install front panel on box;
6. Caulk all inside box joints;
7. Drill pilot holes for rear panel and install weather stripping;
8. Install drivers and port on front panel and electrical terminals on rear panel;
9. Install crossover network on bottom panel;
10. Wire crossover network to drivers;
11. Staple fiberglass to all interior walls except the front panel;
12. Connect input terminals to crossover network and install rear panel;
13. Finish box exterior and install grille.

GLUE, CLAMP AND SCREW. The bracing is recessed ¾ inch for the front panel and $\frac{13}{16}$ inch for the rear one. The extra $\frac{1}{16}$-inch thickness allows the weather stripping seal for the removable rear panel. (Note: the dimensions of the front and back panels have been reduced

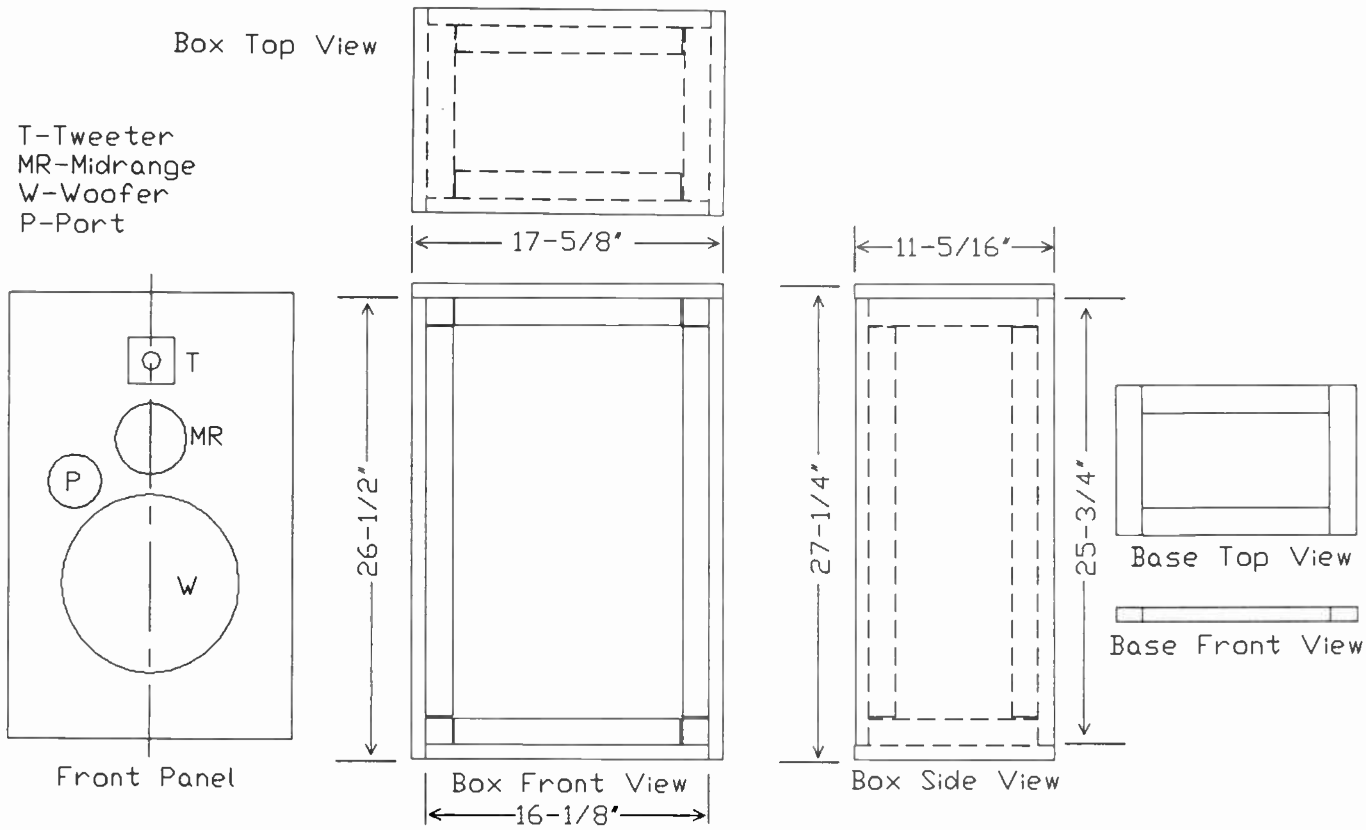


FIGURE 2: Assembly diagram. Note: rear panel bracing must be recessed 1/16" more than front panel bracing to allow for thickness of weather stripping seal.

by 1/16 inch from the exact calculated value to ensure proper fit without force.) The simplest procedure to assemble the top, bottom, and two sides is:

1. Use two corner clamps (Craftsman 6666 or equivalent) to clamp the bottom and one side together at a 90° angle.
2. Apply wood glue to inside joint;
3. Place one corner bracing strip in the joint and coat the two surfaces of the brace that contact the box panels with wood glue;
4. Use two 3-inch C-clamps (Craftsman 66673 or equivalent) to clamp each end of the corner bracing strip to the bottom panel;
5. Mark and drill pilot holes through the side panel and into the corner bracing strip for three wood screws (a wood screw pilot bit for the particular size of screw to be used is recommended);
6. Install screws in three pilot holes (use a drill-powered screwdriver or an automatic return screwdriver);
7. Remove the two C-clamps and tighten screws just installed;
8. Mark and drill pilot holes through bottom panel and into corner bracing strip for two wood screws, centered between adjacent screws installed in the side panel;
9. Install screws in bottom panel;
10. Remove the two corner clamps and repeat procedure until top, bottom, and both sides are assembled—you need not wait for the glue to dry, for the screws will hold the joints together;
11. Install all remaining bracing strips, using the C-clamps, wood glue between the bracing and panels and enough screws to ensure that the bracing fits flat against the panels (the screw holes can be filled with Elmer's Wood Putty).

The internal bracing strips should be cut from fir or pine "two-by-two" bracing (following the parts list), which are actually 1.5 by 1.5 inches. Do not use hardwood bracing, for it is too difficult to screw into. The front panel bracing is recessed by ¾ inch. The back panel bracing is recessed by 13/16 inch. The lengths of the top-to-bottom and left-to-right bracing strips have been reduced by 1/16 inch to ensure proper fit without force. Use caulk to seal any gaps between the bracing strips.

FRONT AND BACK. The drivers mount to the front panel from the *outside* of the box. *Do not* mount them from the inside. Cut circular holes in the front panel for the three drivers and the port. In addition, the appropriate hole for the speaker input terminals must be cut into the back panel. A saber saw can be used to cut the holes, although they may be easier to cut with a hole saw or with a circle cutter in a drill press.

The drivers should be placed close together (about one panel thickness apart) on a vertical line with the tweeter at the top and the woofer at the bottom. The port or vent tube should be close to the woofer and away from the side panels. The port has a 3-inch internal diameter and a length of 4 inches. It mounts flush with the outside edge of the front panel. If you can, use a router

Continued on page 14

COMPONENT LIST

Drivers
Eminence 10290 10" Accordion Edge Woofer
Peerless K040MRF/8 4" midrange
Philips AD11600/T8 1" dome tweeter

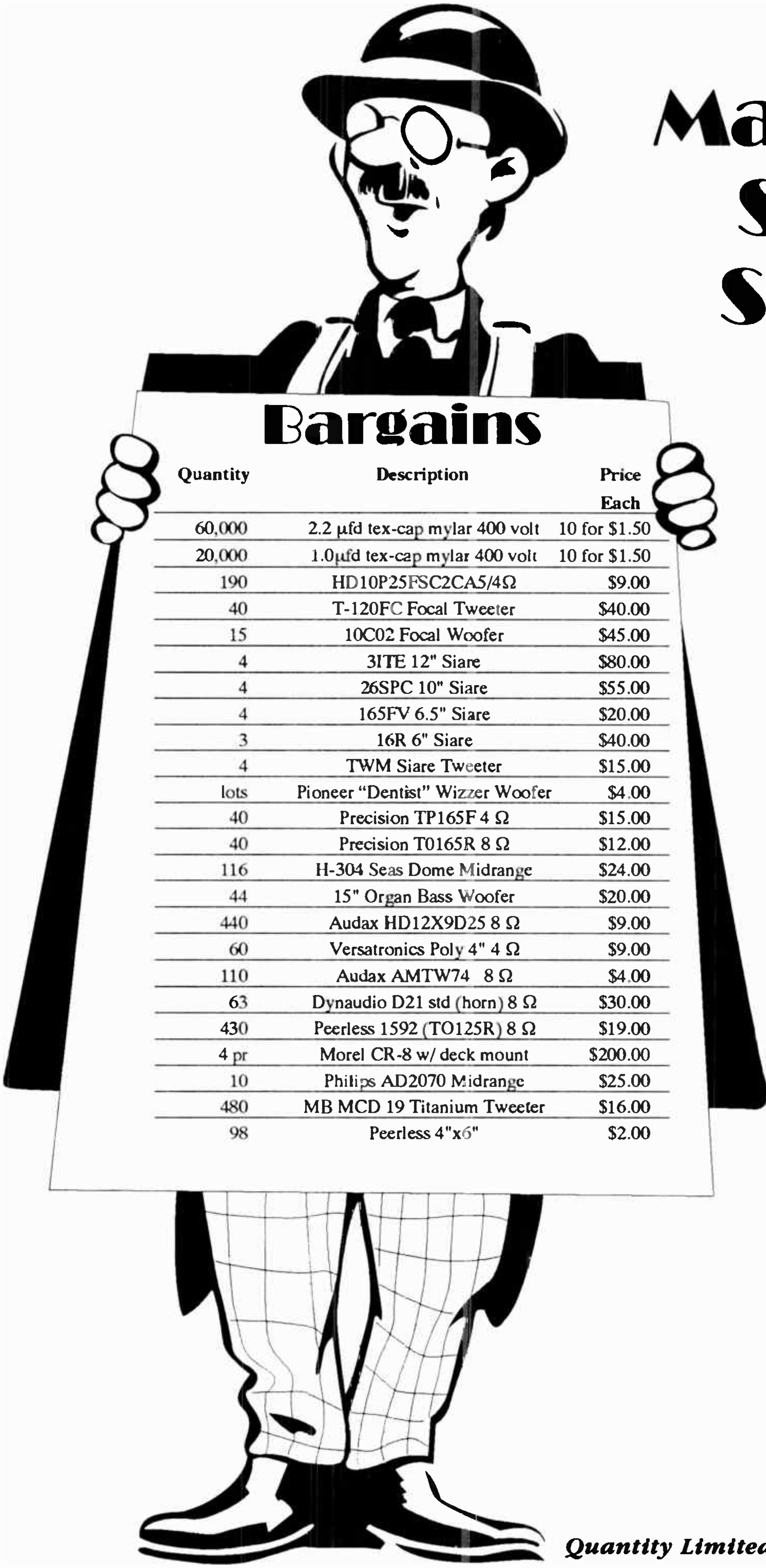
Capacitors
50µF 50V (or greater) non-polar electrolytic
30µF 50V (or greater) non-polar electrolytic
4µF 50V (or greater) Mylar

Inductors
3mH air core
6mH ferrite core
0.68mH air core
0.5mH air core

Resistors
3Ω 10W
15Ω 10W

Two of each item are required

Madisound Spring Special '89



Bargains

Quantity	Description	Price Each
60,000	2.2 µfd tex-cap mylar 400 volt	10 for $1.50
20,000	1.0µfd tex-cap mylar 400 volt	10 for $1.50
190	HD10P25FSC2CA5/4Ω	$9.00
40	T-120FC Focal Tweeter	$40.00
15	10C02 Focal Woofer	$45.00
4	31TE 12" Siare	$80.00
4	26SPC 10" Siare	$55.00
4	165FV 6.5" Siare	$20.00
3	16R 6" Siare	$40.00
4	TWM Siare Tweeter	$15.00
lots	Pioneer "Dentist" Wizzer Woofer	$4.00
40	Precision TP165F 4 Ω	$15.00
40	Precision T0165R 8 Ω	$12.00
116	H-304 Seas Dome Midrange	$24.00
44	15" Organ Bass Woofer	$20.00
440	Audax HD12X9D25 8 Ω	$9.00
60	Versatronics Poly 4" 4 Ω	$9.00
110	Audax AMTW74 8 Ω	$4.00
63	Dynaudio D21 std (horn) 8 Ω	$30.00
430	Peerless 1592 (TO125R) 8 Ω	$19.00
4 pr	Morel CR-8 w/ deck mount	$200.00
10	Philips AD2070 Midrange	$25.00
480	MB MCD 19 Titanium Tweeter	$16.00
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Continued from page 12
to recess the drivers so they mount flush with the front panel.

CAULKING. After assembling the box (minus the rear panel), seal all inside joints with a permanent caulk. Clear silicone rubber caulk is the best, but is the most expensive. One to two 10-ounce tubes (the type that requires a caulking gun) will do the job. The inside joints between the front panel and the port should be caulked. In addition, apply a thin bead of caulk to all sealing surfaces of the drivers and the input terminals before installation. The drivers should be carefully secured to the front panel with machine screws, flat washers, and nuts.

CROSSOVER. Install the crossover network (*Fig. 3*) on the inside bottom panel of the box using glue or caulk. Two or more solder-lug terminal strips screwed to the panel can be used for solder binding posts to wire the crossover network. All electrical connections inside the box should be soldered carefully with rosin core solder only. Any corroded terminals should be cleaned with steel wool or a wire brush before soldering. I recommend 18- or 20-gauge stranded wire for all wiring. Single strand wire should *not* be used.

Double-check the polarity of each driver connection to the crossover network. The positive reference input terminal to each driver is indicated by a red dot or red wire. This terminal must be connected as specified on the circuit diagram for the crossover network. Do not overheat the tweeter terminals, for the internal connections can break.

The inductor wire has a reddish-brown shellac insulation which must be scraped off each end with a knife before soldering. Inductor resistances should measure 2.2Ω for 6mH, 1.1Ω for 3mH, 0.54Ω for 0.68mH, and 0.44Ω for 0.5mH (all values ± 10%). Too low a resistance value can affect frequency response. Resistance can be increased by adding series 10W resistor if required.

FIBERGLASS. All inside box surfaces except the front panel should be lined with 1-inch thick fiberglass. This can be purchased from Radio Shack (#42-1082). The fiberglass should be stapled to the box walls with a staple gun. To prevent the staples from cutting through the fiberglass, use a small rectangular strip cut from an index card, folded double between each staple and the fiberglass. Any leftover scraps of fiberglass can be stapled to the front panel over the back of the midrange and the tweeter. It should not cover the woofer or come near the port opening. If construction type fiberglass is used, the paper backing must be peeled off.

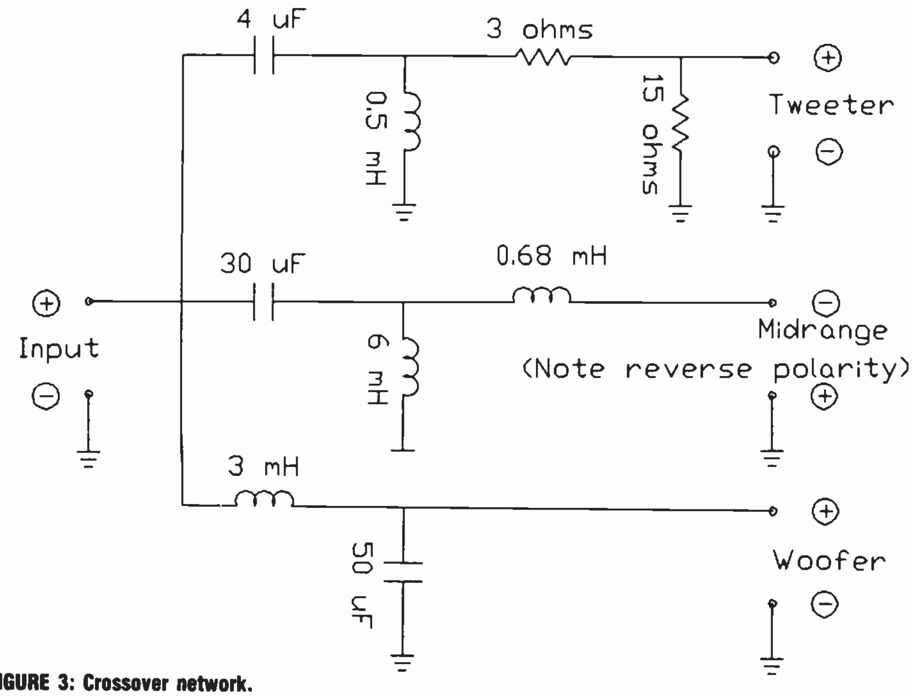

FIGURE 3: Crossover network.

REAR PANEL. To provide access to the inside of the box in case repairs are needed, the rear panel is not glued in place. To ensure an airtight seal when the rear panel is screwed to the bracing, use foam weather stripping between the rear panel and the bracing. I recommend 3/16-inch by 1¼-inch wide, adhesive-backed closed-cell foam tape, such as that made by Macklanburg-Duncan Co. or by W.J. Dennis Co. The bracing for the rear panel must be set into the box by an amount equal to the thickness of the panel (¾ inch) plus the thickness of the compressed foam tape (about 1/16 inch). The length of the front-to-rear corner bracing strips specified in the Parts List allows for the 1/16-inch thickness of the weather stripping.

The electrical input terminals on the rear panel can be either 5-way binding posts or those specifically designed for loudspeakers. If you use the 5-way type, recess them to avoid damage during transport. Cheap 5-way binding posts break off easily. Good quality ones are made by the H.H. Smith Co. (model 899). I do not recommend gold plated ones, for they are expensive. I prefer a recessed "cup" type terminal with two integral 5-way binding posts that is specifically designed for loudspeakers.

THE GRILLE. This box design has the front panel flush with the front edges of the top, bottom and side panels. A grille frame can be made with rectangular or quarter-round moulding strips to cover these edges, then black single-knit fabric that can be purchased at most cloth stores can be stretched around the grille frame and stapled to the back. The grille frame can be installed on the front panel with standard grille mounting hardware, or four small blocks of wood can be attached to the front panel over which the grille frame fits.

An alternative box design would have the front panel and grille recessed so that the front edges of the top, bottom, and side panels are exposed. This design requires that these edges be finished with veneer tape when the final finish is applied. If you prefer this design, you must increase the appropriate dimension of the top, bottom, and side panels so that the internal box dimensions will not be changed.

QUALITY PARTS. The loudspeaker drivers are standard models from Eminence, Peerless, and Philips. Substitute drivers must not be used. I recommend Mylar capacitors. To prevent breaking the capacitors' leads, do not bend or flex the leads close to the capacitor body. If necessary, capacitors can be paralleled to achieve the value specified. The better inductors are wound with a higher gauge wire and exhibit less series re-

Continued on page 70

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A PASSIVELY ASSISTED WOOFER SYSTEM

BY TOM NOUSAINE

PHOTO 1: Finished system—a three-way design.

We all want low bass and small enclosures. Here's a design based on a Society of Automotive Engineers (SAE) paper published in 1986, which I believe offers the fewest trade-offs.[1] The basic system uses a modified Precision TA305F to produce 25–30Hz in a 1.2 $ft.^3$ box, with roughly the same sensitivity as a compound dual woofer, but without the cost of a second driver. A second driver restores sensitivity to approximately original levels, and without wasting the piston area of the second driver. Thus a 2.4 $ft.^3$ box gives performance below 30Hz with good sensitivity.

The alignment, called CG3, devised by Clark and Geddes uses a large series capacitor to boost the voltage drive to the speaker below system resonance. The drivers must have a low resonant frequency and a fairly high mechanical Q, since the system Q_{ms} is to be 10. The electrical Q_{es} of the alignment is targeted at 1.1. The basic idea is to increase the cone's mass about 80%, for a driver of about 25Hz F_s, and a Q_m of 4.5 or greater. This gives a Q_m of about 7; F_s, 21Hz; and Q_e, approximately 0.53.

When you install the modified driver in a box of F_{sb} 42Hz, the F_3 will be 27Hz with a 1,900μF capacitor added in series. The finished system can be trimmed for Q_m by adding stuffing to the box.

In practice, the Precision TA305F's electrical characteristics fit this alignment closely. I have constructed two different stereo pairs and both work pretty much as expected. I discovered using a capacitor of smaller size also works well.

I will describe my first design in detail and then the second in general. First, I purchased four Precision TA305F 12-inch woofers. The electrical parameters are shown in *Table 1*. The drivers were all within 10% of each other in specs. I measured within the same tolerances with the second set of four I bought.

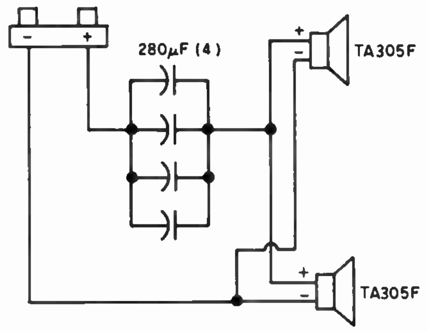

FIGURE 1: The alignment uses a large series capacitor to boost the voltage drive to the speaker below system resonance.

CONE SYRUP. I modified each driver by adding mass to the cone until the F_s was 21Hz. To do this I placed a ring of small lead fishing sinkers symmetrically around the dustcap and then trimmed the F_s by adding Liquid Rubber to increase mass and glue the sinkers to the cone. You can do this while the driver is hooked up to your signal generator and voltmeter. The liquid rubber is swell, pouring from the tube with a consistency of syrup. With the driver lying on its back the goo naturally flows evenly around the dustcap. If you use lead sinkers or shot you will need two or three tubes for all four drivers; using Liquid Rubber without sinkers will require about six tubes.

Allow to dry overnight and then recheck the F_s. Because fluid evaporates, the F_s will increase slightly, so touch up the F_s with a little more goo. Then, I matched my drivers to get a Q_m as consistent as possible between each pair. I

TABLE I

DRIVER PARAMETERS

#	R_e	F_s	Q_m	Q_e	Q_t	V_{as} ($ft.^3$)	V_b ($ft.^3$)	Q_{ms}	Q_{es}	F_{sb}	F_3
1	5.6	27	4.7	0.34	0.31	6.6					
2	5.5	26	4.8	0.35	0.33	5.7					
3	5.6	27	4.5	0.34	0.31	5.2					
4	5.6	28	4.5	0.37	0.34	6.7					
1 + 2	3.0	21	7.6	0.54	0.51	6.7/3.12	2.1	9.6	1.1	43	27
3 + 4	3.0	21	7.3	0.57	0.53	7.2/2.72	2.6	9.5	1.1	40	26

ABOUT THE AUTHOR

Tom Nousaine, Director of Capital Recovery for Ameritech Services in Chicago, has worked in telecommunications for over 17 years. He presently writes product reviews for Car Stereo Review *and* The Sensible Sound *and is Secretary of the Chicago AES. He has been active in loudspeaker design and construction since joining SMWTMS and subscribing to* TAA *in the seventies.*

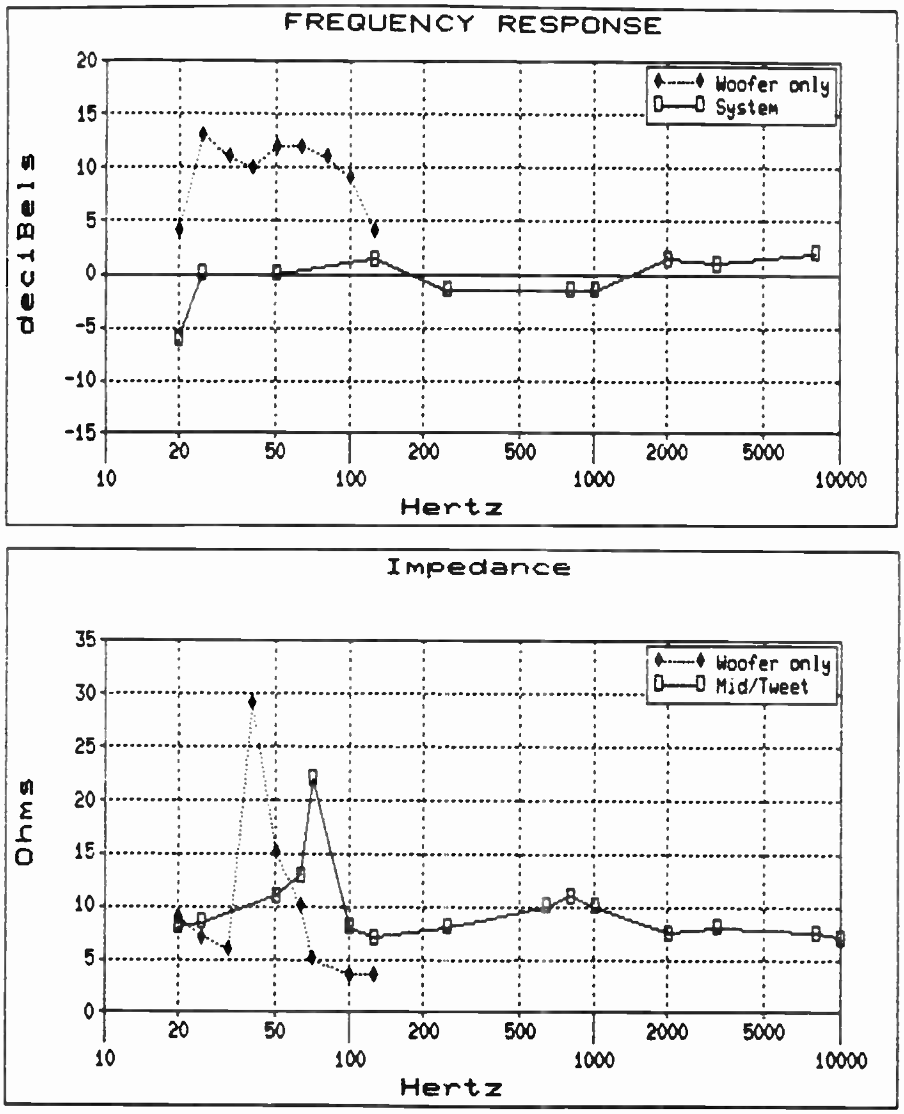

FIGURE 2: Impedance and frequency response for the first system.

matched drivers 1 and 2, and numbers 3 and 4.

Then I measured each pair as a set. Notice that the F_s remains the same, the V_{as} is significantly reduced to about 7 ft.^{3} for two drivers, and the Q_m is about 7.5. From this data we can determine the appropriate alignment.

SYSTEM ALIGNMENT. Because this alignment was a first-time experiment, and the woofer section was to be part of a large three-way system, I designed the cabinet considerably oversized just in case this wasn't going to work. The alignment calls for a box Q_{es} of 1.1; I determined the box volume for one at 2.6 ft.^{3}, or slightly larger than anticipated, and the other was right on target at 2.1 ft.^{3}. I installed the drivers and then glued Styrofoam blocks inside until the F_{sb} was 40 and 43Hz, respectively. I measured the Q_{ms} and Q_{es} for each system after installation, but before adding the series capacitors. Q_{es} was right on target, but Q_{ms} was somewhat below design requirements. This doesn't seem to affect performance in the finished product.

Using the formula, $C1 = 0.234/(R_e \times F_{sb})$, we need a 1,800$\mu$F capacitor. I used four nominal 280μF nonpolar electrolytics, with good results, available from Madisound for $3 each. The measured value of the four parallel caps is 1,212μF.

The bass section contains the two Precision 12-inch woofers, passively assisted, and driven by a 115W amplifier below 150Hz using an electronic crossover. The midrange/tweeter section is driven by another 115W amplifier above 150Hz through a passive 18dB/octave filter at 2.1kHz. I used Dynaudio drivers, a D28 tweeter and a 17W75 7-inch midrange, which is enclosed in a separate, interior 0.75-ft.^{3} cabinet. The system half-power bandwidth is 27Hz.

I constructed the cabinet from high-density ¾-inch MDF, using glue, drywall screws and caulk sealer. I mounted the four drivers on a removable front panel, fastened and sealed to the cabinet with wood screws and foam weather stripping, and attached the drivers, using gaskets, with T-nuts and ¼-inch machine bolts. The cabinet was finished with plywood veneer, using two coats of hand-rubbed varnish over dark wood stain.

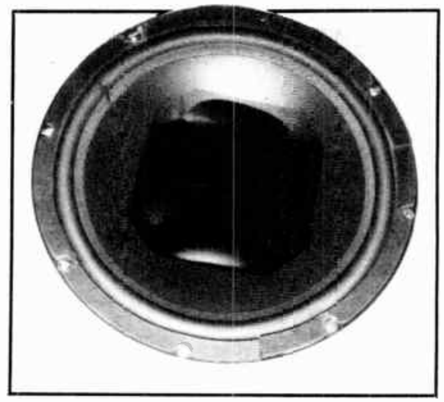

PHOTO 2: "Gooped" woofer with black contact paper cover.

PERFORMANCE. For my second set, I built a separate woofer cabinet to the correct calculated size, derived from the electrical specs of two TA305F pairs, modified with the added mass. In this case, I tricked myself and the box was not large enough to obtain a Q_{es} of 1.1. Instead I obtained 1.3. But my Q_{ms} was right on target. In any case, the system's bass performance was approximately the same as my first design, although I never listened to them together in the same room at any time. The woofers' impedance and frequency response graphs for my first system are shown in *Fig. 2*, using closely miked pink noise. The combined woofer pair's SPL is approximately 89dB, 1W/1m.

I biamplified both sets of woofers and used electronic crossovers of 150–200Hz. The woofers perform well up to 200Hz. The minimum impedance reaches 3.5Ω at 100Hz. They are easily driven with most good quality amplifiers such as the Hafler DH-200/220 series. Although they won't snowflake my listening room at high levels, compared to my 16Hz system, these little babies add a wonderful sense of ease and lower register impact, lacking in many systems that are considered full-range. The design requires a larger enclosure than a compound woofer but you get back your sensitivity by not wasting the output of the second driver.

PHOTO 3: Alternative woofer design with the author's satellites.

MATERIALS

4	Precision TA 305F 12-inch woofers
8	280μF nonpolar electrolytic capacitors
2	dual banana 5-way binding posts
5 ft.	14-ga. zip cord
4-6	tubes Liquid Rubber (black or white)*
	Lead shot*

*Available at ACE Hardware.

STEP BY STEP. 1. Modify four 12-inch TA305F woofers by adding mass until F_s is 21Hz.

2. Match driver pairs so Q_m is equal between the two sets.

3. Measure Q_e, Q_m, and V_{as} for paired sets.

4. Design your CG3 system:

$$Q_{es} = 1.1/Q_e = \sqrt{(V_{as}/V_b) + 1}$$

0.5 = F_{sb}/F_s, and F_3/F_{sb} = 0.64. Typical V_b = 2.1 ft.^{3}; F_{sb} = 43Hz; F_3 = 27Hz.

5. Construct cabinets: typical size is 27¾ by 13¾ by 16 inches. Use MDF or high-density particle board. Glue all joints and fasten with drywall nails. Cut two woofer holes. Mount dual banana jacks on the rear.

6. Install woofers: use four 280μF nonpolar electrolytic capacitors inside the cabinet. Mount the woofer with T-nuts, bolts and foam gaskets. Attach

Continued on page 71

THE DIGITAL FILTERS

BY FERNANDO GARCIA VIESCA

Filter networks have been used in audio since the beginning of electrical sound reproduction. In the early stages, filters were composed of passive inductive and capacitive components. Circuit losses, difficulty of winding high-quality inductors, hum pickup, impedance effects and bulkiness were pitfalls with which the audio engineers had to live.

In the late forties, George Philbrick introduced a mysterious circuit labeled *operational amplifier*. Few engineers understood the potential of such a circuit, only that it could be used for servo applications.

A breakthrough occurred in 1965, when Bob Widlar at Fairchild designed a monolithic operational amplifier, the now famous 709. Despite numerous shortcomings, its small size and low cost enabled engineers to design working cir-

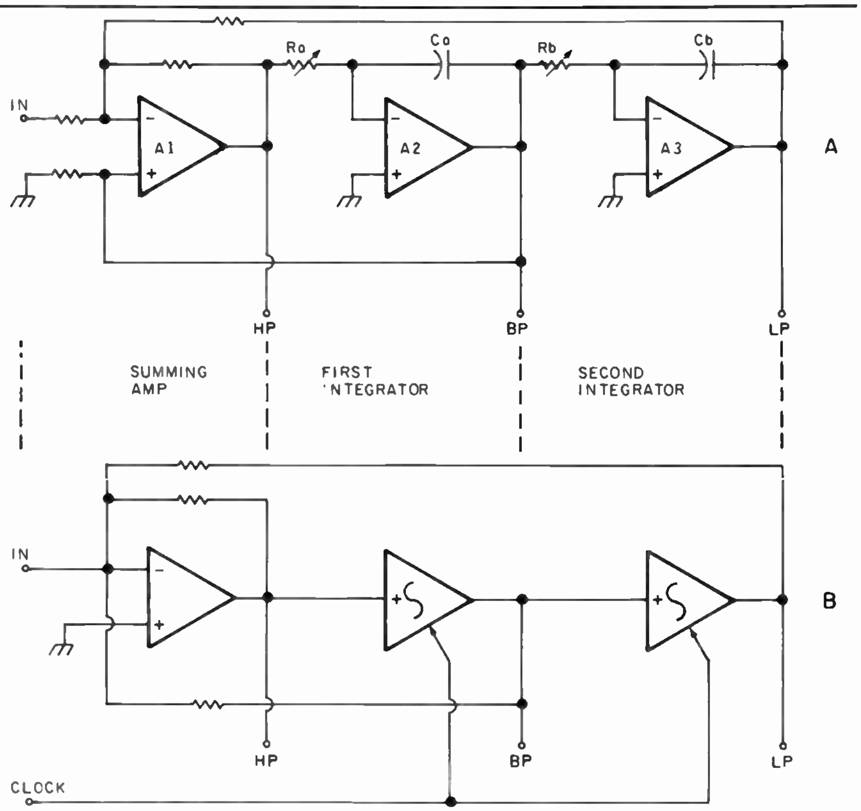

FIGURE 1: Conventional state-variable universal filter (a) made of three stages; (b) in a switched capacitor filter; two are replaced by clock-tuneable integrators.

ABOUT THE AUTHOR

Fernando Garcia Viesca who resides in Mexico, is an electrical engineer with broad experience in industry. For many years he worked for Sylvania/ GTE. He has written for a wide variety of electronics journals, and has a long-standing love for audio.

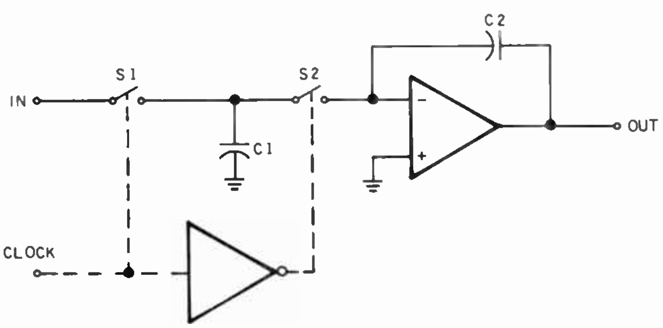

FIGURE 2: A tuneable integrator is made of two non-overlapping switches and two capacitors.

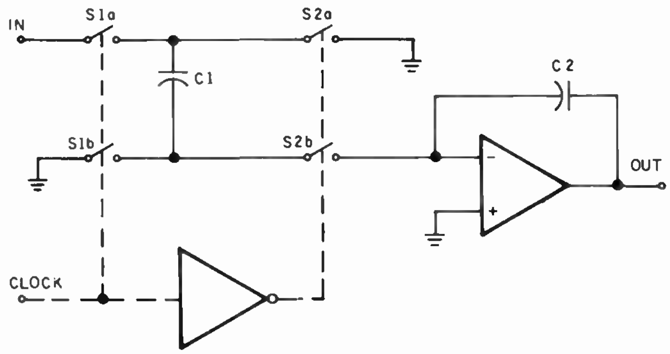

FIGURE 3: Actual integrator has two additional switches to provide a noninverting integrator function.

cuits attainable only in theory until then. Active filters appeared, allowing buffered inductorless filters, with many advantages already discussed in these and other publications. Op amp technology advanced in leaps and achieved considerable feats.

Then, less than a decade ago, National Semiconductor introduced the MF10, a monolithic switched capacitor filter. All filter responses could be achieved without any reactive component. Tuning the filter was easily achieved by modifying the clock frequency, and adjusting the circuit's Q was easier.

Even though these filters are no longer new in the IC scenario, they have not found widespread use in high-quality audio applications. I wondered why, since they offer several attractive features.

In this article I explore the filters' different characteristics, to determine whether or not the digital filter could find widespread use among *SB* experimenters.

First, I'll briefly describe how these filters work.

SWITCHED CAPACITOR. To understand how the digital filter evolved, let's look at a universal state-variable, second-order filter. This filter configuration is very useful because by tapping the signal at different points, all filter responses are achieved. Also, the cutoff frequency is adjusted with a pair of resistors, and Q is independent of this frequency.

As shown in *Fig. 1a*, this active filter has a summing stage (A1) and pair of integrators (A2 and A3, and associated resistors Ra, Rb and capacitors Ca, Cb). Those RC time constants tune the filter.

These are substituted in the digital filter by a pair of switched capacitor integrators, shown to the right in the simplified diagram of *Fig. 1b*. The integrator's time constant may be tuned with the clock frequency. In this way, a pair of switches and capacitors replace a variable resistor and a capacitor. The schematic of the tuneable integrator is shown in *Fig. 2*.

Switches S1 and S2 are driven out of phase and in a break-before-make fashion. When switch S1 is closed, capacitor C1 stores a charge equal to $V_{in} \times C1$. In the next clock state, S1 opens and S2 closes, and therefore all the charge is fed to capacitor C2. The amount of this charge is equal, of course, to $V_{in} \times C1$.

Since we know from our physics courses that electrical current is defined as an amount of charge conducted over a period of time, then the current charging capacitor C2 is:

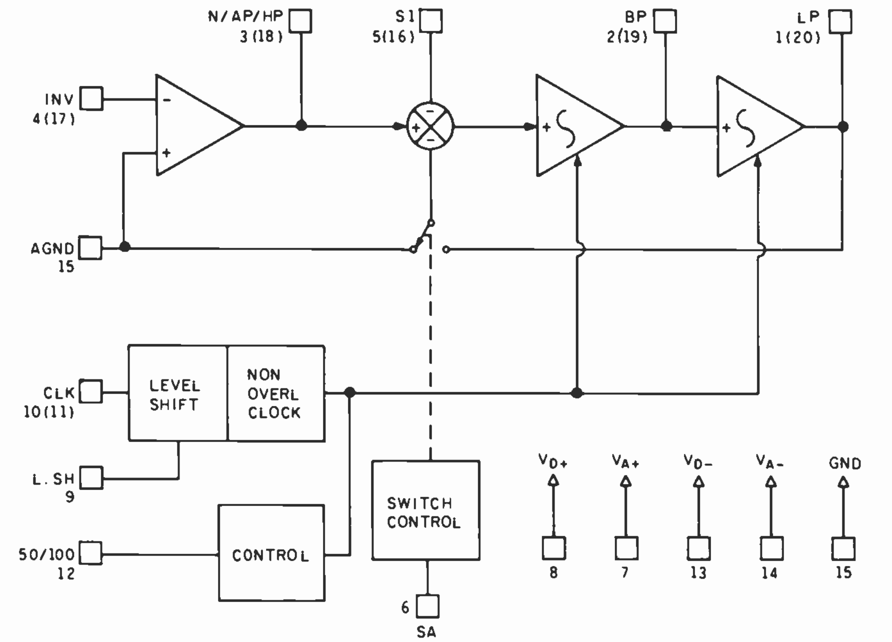

FIGURE 4: Block diagram of a monolithic switched-capacitor filter, the MF10. Numbers in parentheses indicate second section.

$$Ii = dQ/dT = (V_{in} \times C1)/T$$

Since the period is the reciprocal of the clock frequency, this equation may be rewritten:

$$Ii = V_{in} \times C1 \times F_{CLK}$$

The effective resistance as seen from V_{in} to the op amp's inverting input is therefore:

$R_{eff} = V_{in}/Ii$; or by substituting Ii:

$$R_{eff} = 1/(C1 \times F_{CLK})$$

The value of this resistor may be "tuned" by changing the clock frequency.

An integrator's time constant can be defined as:

$$t = C2 \times R_{eff}; \text{ or } t = C2/(C1 \times F_{CLK})$$

In a monolithic circuit, the ratio of C2/C1 is easily controlled, reproducible, and with a close thermal match we may assume it is constant. Therefore, the only parameter that affects the time constant is the clock frequency. A frequency-tuneable integrator has been achieved with this elegant circuit.

Actual integrators within the IC use a more elaborate switching scheme, to provide a non-inverting integrator function, as shown in *Fig. 3*. S1a and S1b are closed together to charge the capacitor. In the next cycle, S2a and S2b are closed. You may note the opposite capacitor plate is grounded, inverting the voltage applied to the op amp's summing node.

National Semiconductor manufactures four different IC switched capacitor filters in IC form. The MF4 and MF6 are respectively fourth- and sixth-order low-pass filters. They are extremely easy to use, but since they are dedicated single-function filters, we'll review only the second-order universal filters, the MF5 and MF10. All filter responses and parameter adjustments may be realized using these devices with different ancillary resistors. The MF10 is a dual version of the MF5. A block schematic is shown in *Fig. 4*.

Besides the summing stage and the pair of tuneable integrators, these ICs incorporate a couple of analog elements that help achieve all filter configurations with a minimum of external components. First, in a three-input summing stage, where two inputs are subtracted from the first op amp's signal, one of the inverting inputs is brought to the outside so that certain connections result in certain filter responses. The other input is routed via the second feature, an internal analog switch to either analog ground or the low-pass integrator. This analog switch is controlled by the Sa/b pin. When this pin is tied to the Vd+ supply, the switch is connected to the low-pass output. Likewise, to switch to analog ground, the pin should be tied to the Vd− supply.

The ratio of the clock frequency to the center frequency is controlled by the 50/100/CL pin. The actual ratio is a trifle smaller (49.94 ±0.6%, or 99.35 ±0.6% at a Q of 10), and it increases/decreases slightly with an increase/de-

Continued on page 22

Continued from page 20
crease of the Q, as shown in *Fig. 5*. These values are for the MF10; for the MF5 they are slightly different.

As with all circuits handling bipolar signals, a split supply of ±5V is recommended. It may be powered by a single supply following the same recommendations of their op amp counterparts; analog ground should be tied to a noise-decoupled, low impedance point, centered at Vdd/2.

Operating Modes. A universal filter must provide all available filter responses, (bandpass, low-pass, and so on) and provide alternatives such as adjustable f_{CLK}/f_o ratios. This is done, as stated before, by three means:

- Logical level of the Sa pin; the internal analog switch is connected to ground or the low-pass output.
- Type of signal connected to the inverting summing node S1.
- Resistor connections.

The switched capacitor filter may be operated in several ways, or modes, depending on the required outputs and features. *Table 1* provides a summary. Unfortunately, I cannot give all the equations and circuit topologies needed for all the modes because I do not wish to unnecessarily lengthen this introductory article. For equations and circuit topologies, consult the references.

The circuit topology for the test is shown in *Fig. 6*; a mode 3 operation.

Equations for these circuits are:

High-pass gain = $-R2/R1$
Band-pass gain = $-R3/R1$
Low-pass gain = $-R4/R1$
Quality factor = $K \times (R3/R2)$
Cutoff frequency = $K \times F_{CLK}/50$, or $K \times F_{CLK}/100$;

where $K = \sqrt{R2/R4}$

TABLE I

MODES OF OPERATION

MODE	RESPONSES	# RESISTORS	ADJ. F_{clk}/F_o	NOTES
1	BP, LP, N	3	No	
1a	BP, LP	2	No	May need input buffer. Poor dynamics for high Q
2	BP, LP, N	3	Yes, above $F_{clk}/50$ or $F_{clk}/100$	
3	BP, LP, HP	4	Yes	Best general purpose mode. Universal state-variable filter.
3a	BP, LP, HP, N	7	Yes	Includes resistor tuneable notch
4	BP, LP, AP	3	No	Inverting all-pass
5	BP, LP, AP	4		Flatter all-pass response
6a	LP, HP	3		Single pole
6b	LP	2		Single pole inverting and noninverting outputs

BP = Band-pass; LP = Low-pass; HP = High-pass; N = Notch; AP = All-pass

Compare these simple equations with those needed for a conventional op amp filter. Although I know many shortcuts for the latter type of filters, it is difficult to beat the digital filter in design simplicity.

You may round out the actual ratio to 50 or 100, but for optimum cutoff frequency accuracy, use *Fig. 5*.

PROs and CONs. There is no free lunch, the saying goes, and digital filters are no exception. We may be tempted to glamorize their advantages, but we must also understand the pitfalls, to avoid a badly engineered circuit. In my experience, worst case situations are often my first-trial and didn't-work situations (a corollary of Murphy's Law, I guess). But first I'll describe the advantages.

Tuneability. To tune a conventional active filter you must modify at least one analog value (resistance, capacitance, and so on) per filter order. In a two-pole filter, matching two resistors precisely over the full tuning range is difficult. And in higher-order filters this is quite an impossible task. In contrast, the digital filters are tuned with a varying clock frequency only, that may come from a voltage-controlled oscillator (VCO), phase-locked loop (PLL), or programmable divider. Very high order tuneable filters are feasible since all sections track within a fraction of a percent. Other difficult applications, such as choosing unusual frequencies or a frequency-tracking filter, are easily achieved, and Q adjustment is independent of cutoff frequency.

Stability. A high-accuracy clock will yield a high-accuracy cutoff frequency, also. Temperature drift and aging are considerably improved over conventional filters.

No reactive components. A great advantage over a conventional RC filter is the absence of capacitors. High-precision, quality resistors are readily available and relatively inexpensive, but unfortunately, capacitors are more expensive and bulkier and have a nasty habit of drifting and aging. Filter accuracy may be degraded, a problem in higher-order filters.

Minimum overall parts. For manufacturability and reliability, plus lower cost (or to achieve a higher performance circuit for the same cost), a circuit with fewer parts is preferred to one with more.

These filters, much like all the digital devices that convert or manipulate analog signals, operate by sampling the analog waveform. This creates the filter's pitfalls.

Sampling steps. The output is not continuous but is a series of small steps, as

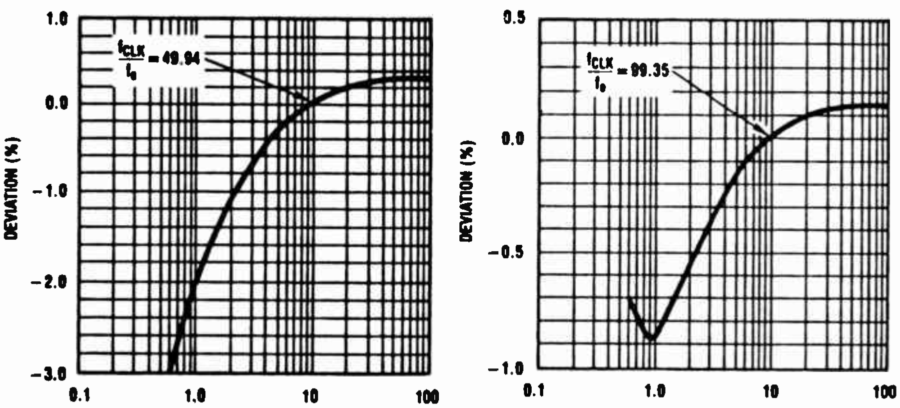

FIGURE 5: Actual clock ratios are modified by the filter's Q. Graphs show actual ratios for (a) 50:1 and (b) 100:1.

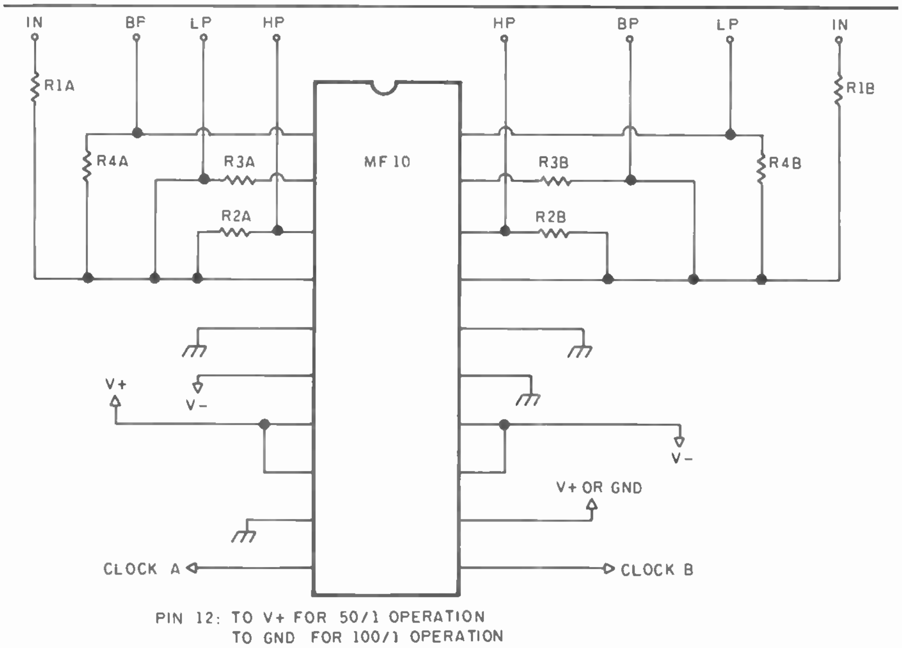

FIGURE 6: Mode 3 operation allows stereo high-pass, low-pass and band-pass outputs in a single IC. In a stereo operation, clock A and clock B are tied together.

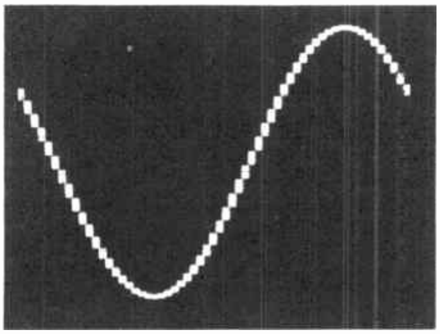

FIGURE 7: Sampled sine wave: input frequency, 1kHz; clock frequency, 40kHz.

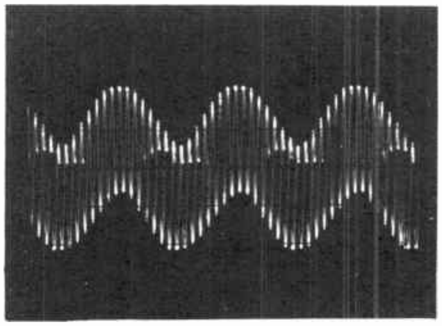

FIGURE 8: Alias distortion: clock frequency equals input frequency equals 100kHz.

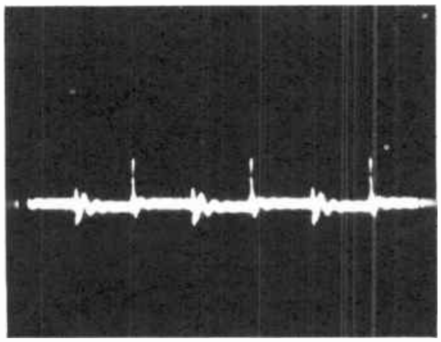

FIGURE 9: Clock noise feed-through. Spikes are 70mV high.

shown in the waveform samples (*Fig. 7*); obtained by feeding the filter with a frequency very close to the clock frequency. Though I deliberately chose an extreme case, the cutoff frequency was made very low in value and the clock ratio pin was set for a 50:1 ratio, this shows a common pitfall that you may run into if you are not aware of the filter's limitations. In the extreme frequencies where the sampling steps could be bothersome, a smoothing low-pass filter can be used at the output. But in low-frequency applications, and whenever the 100 mode is selected, the output should be considered continuous.

Aliasing. The same considerations that apply to all analog-to-digital converters are valid here; foremost is the use of an anti-aliasing filter if your input frequencies come near half the clock frequency. The filter must provide sufficient attenuation above that frequency to place the signals' amplitude below the noise floor.

Alias distortion, as shown in *Fig. 8*, is a hard distortion compared to that produced by slew-rate limiting in conventional linear filters (can any type of distortion be called soft?).

Clock noise. All sampled systems suffer from noise spikes that are coupled to the signals since the switches open and close at the clock rate. Since this rate is much higher in frequency than the signals of interest, it may also be removed by the same low-pass filter used for smoothing the output steps, also clock-frequency related. Clock noise differs from the sampling steps, that is, its amplitude is constant regardless of the input signal's amplitude. Worst case S/N

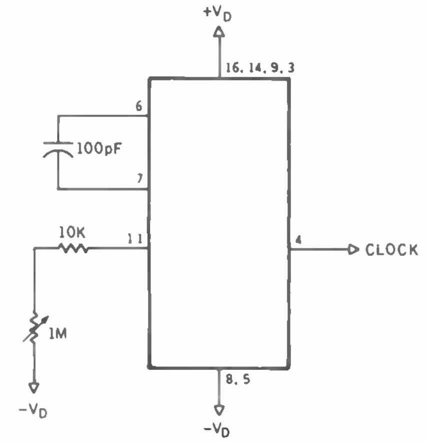

FIGURE 10: Offset nulling for the MF10 and MF5. A bias current is fed into the inverting terminal of the summing amp.

would appear at low levels, as shown in *Fig. 9*.

Offset voltage. Any active filter will have an offset voltage from its op amp(s). The switched capacitor filter has an additional cause of offset voltage; the charge injection of the MOS switches. Therefore, the offset voltage is greater in the 100:1 ratio.

This offset voltage may be nulled by conventional op amp techniques, for example, injecting a small current into a summing node. *Figure 10* shows how this may be achieved for the MF5/MF10 filters. This is different from National's *Linear Databook* approach, but I find it easier and this yields a lower noise level. The MF6 filter has a dedicated pin for nulling, and the offset voltage may also be servo-mechanically removed by an external op amp integrator.

Thermal noise. Any device integrated into monolithic silicon is small, by design. The very small capacitors used in a monolithic digital filter are no exception. A low capacitance value implies a large equivalent resistance value. But large valued resistors generate high thermal noise. Therefore, the noise floor in this type of filter is considerably higher than in a conventional filter.

Aperture effect. In an ideal sampling system the sampling pulses should be infinitely narrow. This is not possible in practice, and the pulses must have a small width. This will cause a low-pass filter aperture effect. Fortunately, for the available filters this effect is minimal because the clock frequency is far from the center frequency. But it can be a problem in high-pass outputs.

TEST RESULTS. As you see from the *Table*, mode 3 is the best general purpose type. This mode also resembles our familiar state-variable filter, therefore for

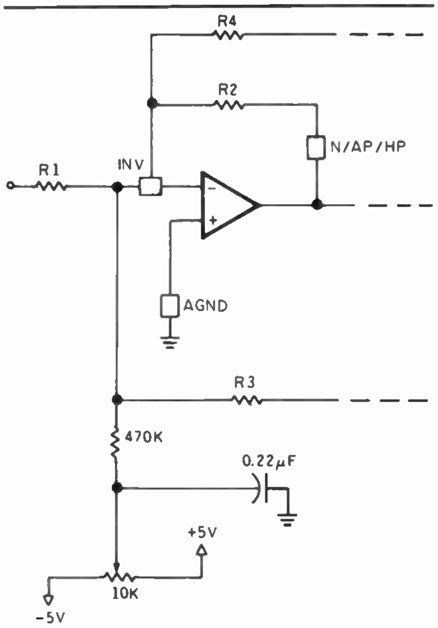

FIGURE 11: A simple 50% duty-cycle oscillator; maximum frequency is around 1.3MHz.

comparision purposes, I built this circuit.

The circuit (*Fig. 11*) had a cutoff frequency of 1kHz and a Q of 0.7. The IC works best with a 50% duty-cycle clock, therefore I generated this clock from the VCO of a CD4046 PLL. This VCO produces a neat, 50% duty-cycle square wave with a minimum of external parts, is easily sweepable, and may be limited with the inclusion of another resistor for a minimum frequency different from zero for a control voltage input of zero. The other PLL's functions are not used.

Of course, you may use your favorite clock circuit here. Unfortunately, 555 timers do not work for the highest clock frequencies, and 50% duty cycles are not readily achievable.

I measured the test results with a 1kHz, 1V test signal.

Signal to noise: 64.5dB
Signal to noise, 30kHz post filter: 82.1dB
Distortion: 1.68%
Distortion, 30kHz post filter: 0.12%
Offset voltage, 100/1, LP output: 320mV
Offset voltage, 50/1, LP output: 230mV
Peaking: 0.8dB
Max. voltage swing before clipping: 2.65V RMS

Now we see that the 30kHz, third-order output post filter greatly enhances the filter performance. Also, the offset voltage is excessive for DC-coupled applications, and it should be nulled.

The maximum voltage swing performance will disappoint those who prefer at least 6V RMS swings. Unfortunately, the low supply voltage of the filter disallows any improvement in this area.

The specs appear fine, but I like to use the best measuring equipment available for a particular test: a trained human ear.

I invited two musician friends to evaluate the sound using a high-end, biamplified system. My original electronic crossover is a two-pole, Butterworth filter, employing the TL074 op amp, which is a good quality amplifier. I checked the filters so that gains, crossover frequencies, and so on, were similar.

Since dynamic headroom is lower for the digital filter, we kept the levels low enough to avoid overloading. We set up the two filters to make A/B subjective listening comparisons, employing different musical selections. The digital filter was set for a 100/1 ratio. No anti-aliasing or post filters were used on the first trial.

Since 100kHz is an extremely high frequency to be reproduced by any tweeter, much less to be heard by anyone, we supposed the slight hiss heard with the digital filter was caused by a higher noise figure of the filter's op amps.

We also heard some raspiness in the highs. Although we supposed that it may be caused by aliasing of some very high harmonics, an anti-aliasing filter did not improve the sound quality. We did not believe any substantial harmonics existed above 50kHz.

So we suspected the next culprit, sampling of the input signal. A post filter, which improved the S/N and distortion figures, not only suppressed the raspiness, but lowered the hissing noise somewhat.

The sound was also slightly muffled. This may be caused by the lower digital filter's dynamic headroom. Though our levels were low, averaging about 0.7V RMS, I am sure many musical transients have peaks several times as large as the average musical level, and those peaks may have clipped.

CONCLUSION. Unfortunately, the switched capacitor filter does not live up to the performance expected for the circuits employed by audiophiles.

The reason is simple; these ICs were designed to be as versatile and universal as possible, and because integration into monolithic silicon will always require certain compromises, various specs required by the high standards of audio reproduction may not be met. That doesn't mean switched capacitor filters will not find their way into audio design. A better performing circuit may be built using discrete components, for example, using high-performance audio quality op amps, or a higher sampling frequency. Some discrete JFET switches may operate at frequencies as high as 10MHz. This could mean a 500:1 clock-to-input frequency ratio, which could be considered continuous. You could completely eliminate aliasing and post filters.

But this does not mean you shouldn't try the actual filters. Complying with the filter's limitations may allow you to build circuits that otherwise would be very difficult, if not impossible, to build.

NOTE: Micro Linear Corp. has an identical pinout switched capacitor filter IC with a maximum clock frequency of 7.5MHz, part #ML2111.

REFERENCES

1. *Handbook of Operational Amplifier Circuit Design*; McGraw Hill, 1976, p. 12-7; description of conventional universal state-variable filter.
2. *Linear Applications Databook*, National Semiconductor Corp., 1986, p. 908; this application note (AN307) describes the basic design criteria and considerations when using the MF10 filters.
3. *Switched Capacitor Handbook*, National Semiconductor Corp., 1985; a thorough discussion of digital filters. This comprehensive book is required reading for anybody doing serious work with the filters. Complete theory, examples, applications, tables and design pitfalls are covered.
4. *Linear Databook*, National Semiconductor Corp., 1982, pp. 9-212; this is the minimum information you will need should you work with digital filters.

THE LISTENING ARC ALIGNMENT

BY DOUGLAS K. RAUER

I first began building speakers in the seventh grade, after a school friend, with help from his father, introduced me to audio. I went right to work creating some "fine" cabinets in wood shop, and purchased a pair of 8-inch drivers with whizzer cones from Allied Radio. That project sure improved the sound of my little RCA phonograph.

In time, I learned more and more about the art of speaker building. Each year we gain bits of new information from sources such as *Speaker Builder*, and try to work the new pieces into our latest audio creation. I would like to inform *SB* readers of another piece of useful information.

Aligning driver diaphragms is nothing new to experienced builders. *Figure 1* illustrates what we've been told about aligning the acoustical centers of driver diaphragms in a vertical line, which is parallel to another vertical line representing the listening position. If the listening position could be an infinite distance from the speaker plane, the two lines would indeed appear to be parallel. But, the listening position isn't an infinite distance from the speaker plane; and the listening position isn't a line, it's a point. If we could build the perfect loudspeaker, it too would be a point (or point source).

FIGURE 1: Driver alignment for a theoretical listening position.

Since we listen to our speakers at a finite distance, determined by the walls of our house, the true speaker "plane" becomes an arc. As shown in *Fig. 2* this arc is created by multiple equidistant lines from the listening position, or the ear. Sounds emitted from any point along the arc will arrive at the ear at the same time. If this listening arc is superimposed onto the speaker plane (*Fig. 3*), we can see the drivers are no longer aligned.

This problem can change depending on how listeners choose to position themselves; that is, slouching down, sitting up straight, or sitting in a different location.

If you carefully read speaker reviews, the reviewers may convey how they position the speakers to obtain the best results. One way may be to tilt the speakers backward, to obtain the best imaging. Tilting realigns the drivers so they fall into the listening arc. Several articles in *SB* discuss techniques for tilting the speakers if you prefer a permanent arrangement. If you don't care for the

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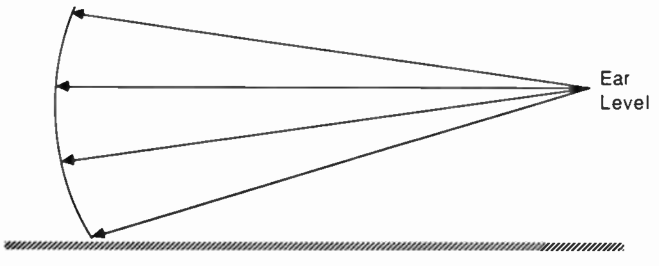

FIGURE 2: The listening arc.

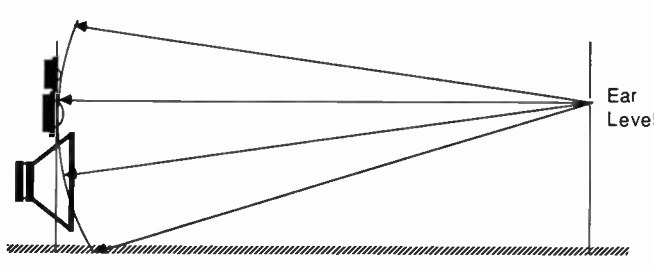

FIGURE 3: The listening arc, which should correspond to the speaker "plane" for proper alignment.

```
100 CLS
110 PRINT "      THIS PROGRAM CALCULATES THE DISPLACEMENT ON A CURVE."
120 PRINT "            FROM THE LISTENERS SITTING POSITION."
130 PRINT "                TO THE FRONT OF A SPEAKER."
140 PRINT " "
150 M=6                         :REM - SET THE STARTING LINE NUMBER OF THE TABLE
160 KON=.25                     :REM - ARC LENGTH INCREMENTS
170 P=22/7                      :REM - SETTING THE FACTOR Pi
180 STP=1/KON                   :REM - CALCULATE THE NUMBER OF STEPS PER INCH
200 INPUT "      WHAT IS THE DISTANCE TO THE SPEAKER IN FEET";DST
205 PRINT " "
210 INPUT "      WHAT IS THE LISTENING HEIGHT IN INCHES      ";HGT
215 PRINT " "
220 INPUT "   WHAT IS THE HEIGHT OF THE TOP DRIVER IN INCHES";TOP
225 PRINT " "
230 INPUT "WHAT IS THE HEIGHT OF THE BOTTOM DRIVER IN INCHES";BTM
235 PRINT " "
240 LPRINT " "
245 LPRINT "     CALCULATING THE ARC DISTANCE, TO ALIGN DRIVER DIAPHRAMS."
250 LPRINT " "
255 LPRINT "     THE DISTANCE FROM THE SPEAKER TO THE LISTENER IS ";DST;" FEET."
260 LPRINT " "
265 LPRINT "     THE LISTENING HEIGHT IS ";HGT;" INCHES."
270 LPRINT " "
300 LNG=DST*12                  :REM - TURN FEET INTO INCHES
310 CALC=INT((TOP-HGT)*STP)     :REM - CALCULATE THE HEIGHEST ARC POINT
320 AL=CALC*KON                 :REM - SET THE MAXIMUM ARC LENGTH
330 GOTO 810
400 DEG=AL*(360)/(2*P*LNG)      :REM - CALCULATE THE ARC TO LISTENING ANGLE
410 RAD=DEG*(P/180)             :REM - CONVERT DEGREES TO RADIANS
420 R=LNG/COS(RAD)              :REM - CALCULATE THE RADIUS OF THE ARC
430 DD=(R-LNG)*COS(RAD)         :REM - CALCULATE DEPTH OF ARC OFF THE VERTICAL
440 AHT=(R)*SIN(RAD)            :REM - CALCULATE THE ARC HEIGHT
450 DELTA=(R-LNG)*SIN(RAD)      :REM - CALCULATE THE HEIGHT DELTA
460 HD=AHT-DELTA                :REM - CALCULATE THE HEIGHT DISPLACEMENT
470 DRHT=HGT+HD                 :REM - CALCULATE THE DRIVER MOUNTING HEIGHT
480 NUM=DRHT-BTM                :REM - SEE IF AT THE BOTTOM POINT YET
500 LPRINT USING "######.######";AL,DRHT,DD,DEG    :REM - PRINT THE RESULTS
600 AL=AL-KON                   :REM - DECREASE THE ARC LENGTH
610 IF NUM<0 GOTO 900           :REM - STOP IF AT THE BOTTOM POINT
700 M=M+1                       :REM - POINT TO NEXT LINE ON THE TABLE
710 IF M<59 GOTO 400            :REM - SEE IF AT THE END OF THE PAGE
720 M=0                         :REM - SET LINE POINTER TO ZERO
800 LPRINT CHR$(12)             :REM - TOP-OF-FORM
810 LPRINT "    ================================================================"
820 LPRINT "         ARC        DRIVER          DEPTH          ARC"
830 LPRINT "       LENGTH       HEIGHT          DELTA        DEGREES"
840 LPRINT "    ----------------------------------------------------------------"
850 GOTO 400                    :REM - GO BACK AND NUMBER CRUNCH
900 LPRINT CHR$(12)             :REM - LAST TOP-OF-FORM
910 END
```

FIGURE 4: Program listing.

aesthetics of this arrangement, then why not incorporate the listening arc concept into your next design? Two designs offering pleasing visual effects are stepped driver mounting, which can be hidden by the grille frame, or cabinets with sloped fronts.

Your own listening room situation and speaker design concept results in a unique listening arc. I have developed a simple BASIC program that will give you a numerical representation of the listening arc, and will allow you to alter several variables to tailor your speaker design or listening habits. *Figure 4* shows the program listing, which I wrote on an IBM PC. I have provided enough REM statements for you to understand what the program is doing, and allow for personal customizing. You may wish to have everything printed out on the screen before committing it to paper, or you may wish to change the incremental steps along the arc length. For adventurous programmers, a plotting routine would be a nice touch.

Figure 5 illustrates a sample run for a possible speaker design. For convenience, I have changed the arc length, from ¼-inch to ½-inch increments, to reproduce it here. This hypothetical

ARC LENGTH	DRIVER HEIGHT	DEPTH DELTA	ARC DEGREES
5.000000	35.998560	0.104150	2.386364
4.500000	35.498950	0.084360	2.147727
4.000000	34.999260	0.066651	1.909091
3.500000	34.499510	0.051042	1.670455
3.000000	33.999690	0.037494	1.431818
2.500000	33.499820	0.026041	1.193182
2.000000	32.999910	0.016653	0.954545
1.500000	32.499960	0.009368	0.715909
1.000000	31.999990	0.004166	0.477273
0.500000	31.500000	0.001038	0.238636
0.000000	31.000000	0.000000	0.000000
-0.500000	30.500000	0.001038	-0.238636
-1.000000	30.000010	0.004166	-0.477273
-1.500000	29.500040	0.009368	-0.715909
-2.000000	29.000090	0.016653	-0.954545
-2.500000	28.500180	0.026041	-1.193182
-3.000000	28.000310	0.037494	-1.431818
-3.500000	27.500500	0.051042	-1.670455
-4.000000	27.000740	0.066651	-1.909091
-4.500000	26.501060	0.084360	-2.147727
-5.000000	26.001450	0.104150	-2.386364
-5.500000	25.501930	0.126027	-2.625000
-6.000000	25.002500	0.149974	-2.863637
-6.500000	24.503180	0.175996	-3.102273
-7.000000	24.003970	0.204105	-3.340909
-7.500000	23.504880	0.234291	-3.579546
-8.000000	23.005930	0.266565	-3.818182
-8.500000	22.507110	0.300910	-4.056818
-9.000000	22.008440	0.337329	-4.295455
-9.500000	21.509920	0.375841	-4.534091
-10.000000	21.011570	0.416419	-4.772728
-10.500000	20.513390	0.459083	-5.011364
-11.000000	20.015400	0.503812	-5.250000
-11.500000	19.517600	0.550608	-5.488637
-12.000000	19.019990	0.599506	-5.727273
-12.500000	18.522600	0.650447	-5.965909
-13.000000	18.025410	0.703478	-6.204546
-13.500000	17.528460	0.758571	-6.443182
-14.000000	17.031740	0.815736	-6.681819
-14.500000	16.535260	0.874974	-6.920455
-15.000000	16.039030	0.936279	-7.159091
-15.500000	15.543060	0.999652	-7.397728
-16.000000	15.047360	1.065078	-7.636364
-16.500000	14.551950	1.132588	-7.875000
-17.000000	14.056810	1.202160	-8.113636
-17.500000	13.561970	1.273779	-8.352273

FIGURE 5: Calculating the arc distance to align driver diaphragms. The distance from the speaker to the listener is 10 feet; the listening height is 31 inches.

speaker design is 42 inches high; the acoustical center of the tweeter is mounted at 36 inches; the midrange, at 31 inches; and the woofer, at 16 inches. Assuming the tweeter and midrange domes have the same diaphragm depth offset, and the woofer has a 1½-inch offset, let's ignore the listening arc and step the woofer 1½-inches on the speaker baffle. This design aligns the driver diaphragms to a vertical line.

In our example, the listener is seated 10 feet from the speakers at an ear height of 31 inches. The midrange happens to be at ear height and is at the point of the curve tangent to the vertical. The tweeter's location at 36 inches reveals the diaphragm offset is 0.1 inch. The woofer's offset is 0.94 inch, in addition to the 1½-inch offset at which it is already mounted.

Raising the listening height to 33 inches improves the tweeter/midrange position, but the woofer's offset increases to 1.2 inches. Lowering the listening position to 29 inches decreases the woofer's offset to 0.7 inch, yet increases the tweeter/midrange offset to 0.2 inch.

You can see how a small change in ear level can affect the listening arc, and where your speaker drivers lie on the arc. If you're designing a new speaker system, adjusting each driver's mounting position on the vertical plane can greatly affect the offset distance. Vertical mounting differences are more apparent for the woofer, and this positioning is critical if you intend to operate the woofer in the midrange frequencies. In the hypothetical design, if the woofer is mounted at a height of 22 inches versus 16 inches, the offset would be 0.33 inch, rather than 0.94 inch. This is a considerable improvement, which can help simplify the cabinet design.

Using this process, you can design an ideal cabinet and driver mounting arrangement to your personal measurements and liking; or you can compromise with your design by adjusting your listening position. Of course if you just built new speakers or you're satisfied with your old ones, you can always use this program to custom design your next sofa or easy chair.

The author's BASIC program is available on an IBM PC compatible, 5¼ inch disk. To obtain a copy contact Old Colony Lab; price is $17.50.

LETTER WRITERS AHOY . . .

We need your cooperation in the matter of your welcome letters to authors and other readers. Please enclose a stamped and addressed envelope if you expect a reply. If the author/reader lives outside the USA, please include two International Postal Reply coupons (available at your post office) instead of stamps on your envelope.

In questioning authors, please leave room in your letter for replies which should relate to the article, be framed clearly, and written legibly. Please do not ask for design advice or for equipment evaluations.

Letters to authors or other readers cannot be acknowledged, unfortunately. Any letter which does not comply with the requests above will not be answered.

KITS • COMPONENTS

What's Included?

Kits include all the parts needed to make a functioning circuit, such as circuit boards, semiconductors, resistors and capacitors. Power supplies are not included in most cases. Unlike kits by Heath, Dyna and others, the enclosure, faceplate, knobs, hookup wire, line cord, patch cords and similar parts are not included. Step-by-step instructions usually are not included, but the articles in *Audio Amateur* and *Speaker Builder* are helpful guides. Article reprints are included with the kits. Our aim is to get you started with the basic parts—some of which are often difficult to find—and let you have the satisfaction and pride of finishing your unit in your own way.

PREAMPS

DAVE VORHIS

KP-8: VORHIS LAST PAS MOD. [4,5:82] Precision parts, 1% polystyrene capacitors and metal film resistors, gold-plated phono jacks and B+ regulator board included. Does not include balance, volume control, or 71k capacitors. Each **$220**

FRED GLOECKLER

KP-6: MIKE PREAMP. [3:82] Gloeckler's adaptation of Advent's MPR-1. All parts, input transformer, gain switch and PC board for two channels. Case, connectors and batteries not included. Each **$75**

JOSEPH CURCIO

KS-6: CURCIO VACUUM TUBE PRE-PREAMP. [5:84] Contains board, all parts, jacks, tubes & sockets for amplifier/regulator. With master power supply, including board and transformers. No enclosures supplied. Each **$135**

KT-1: CURCIO TUBE PREAMP. [2:85] Contains board with all parts, bridge, relay & socket, tubes & sockets, heatsinks, and switches. Power supply with transformers included. Each **$365**

KV-3: AUTO MUTE. [1:86] Turn-on delay for power amp and speaker protection. Each **$18**

ERNO BORBELY

KT-2: BORBELY PREAMP. [4:85, 1:86] A highly sophisticated preamp using the best technology and design ideas available. KT-2 includes five sections for each channel of the stereo preamplifier. Each section is built on its own board. In addition to the two boards of the amplifier proper, there is the tape buffer, line amplifier and power supply. The kit comes complete with two function switches and volume control. Included are two toroidal power transformers, gold-plated connectors and all precision passive and active components. An article reprint is included. The kit does not contain hook-up wire or cable, solder, housing or knobs. Each **$650**

KT-2 SECTIONS. (1-5) Available separately. Inquire for pricing.

KT-2/KW-3: With supply upgrade to KW-3. Each **$795**

KW-3: BORBELY IMPROVED POWER SUPPLY. [1:87] This single channel, low impedance supply was designed for the exacting requirements of Erno Borbely's moving-coil preamp (2:86, 1:87). The design utilizes polypropylene caps and 1% metal film resistors. LM317/337s are used in the preregulator and Signetics NE5534 in the op amp regulator. The kit includes a low profile 24V toroidal transformer, 4¼" x 5½" circuit board and all board mounted components. Chassis and heatsink are not included. Each **$135** Two or more **$125**

SULZER PREAMP POWER

KL-4A: OP AMP PREAMP VOLTAGE REGULATOR. [2:80] Sulzer ±15V regulator, 30mA capacity. All parts and board. Requires ±21V filtered DC input. No transformer, filter caps, or rectifier. Each **$40**

KL-4B: SULZER RAW DC SUPPLY. Triad F-91X transformer, diodes and two 5,000µF @ 35V capacitors. Will power two KL-4A supplies. Diagram supplied for construction. Each **$48**

KL-4C: SULZER RAW DC SUPPLY. Same as KL-4B but contains ILP ±22V toroidal transformer. Each **$60**

KL-4D: ILP ±22V toroidal transformer only. Each **$55**

POWER AMPLIFIERS

WILLIAM Z. JOHNSON

KH-6: AUDIO RESEARCH ST-70-C3 MODIFICATION KIT. [4:77] Kit supplies ALL parts: wire, tubes, controls, assembled circuit board and hardware to modify Dynaco's Stereo-70 stereo power amplifier to *Audio Research Corporation* levels of performance. The transformers, power switch and cord, fuse post, chassis and a few of the screws and nuts are the only parts of the Dynaco ST-70 unit used. Fully detailed construction booklet with step-by-step instructions. Each **$420**

KH-6B: AUDIO RESEARCH ST-70-C3. Modification kit construction booklet. Each **$5**

ERNO BORBELY

KP-3A: BORBELY 60W MOSFET AMPLIFIER. [2:82] All parts for one channel (except "L" choke) including board, driver and output heatsinks. Each **$90**

KP-3P: BORBELY 60W MOSFET AMP POWER SUPPLY. [2:82] Two channels. Transformer and all parts except chassis and large filter caps. Each **$80**

KP-3PC: BORBELY 60W POWER SUPPLY. [2:82] Power supply with two 10kuF @ 75V output filter caps. Each **$88**

KS-1: BORBELY SERVO 100 MOSFET POWER AMP. [1:84] One channel includes all parts, board and output heatsinks. Each **$150**

KS-3: BORBELY DC 100 MOSFET POWER AMP. [2:84] One channel includes board, all parts, and output heatsinks. Each **$160**

KS-3PA: SERVO 100, DC 100 POWER SUPPLY. [2:84] Two channels, includes two 10kuF @ 75V caps, one bridge, fuses and one 500VA toroidal transformer. Each **$175**

KS-3PB: SERVO 100 or DC 100 MONO POWER SUPPLY. [2:84] One channel with all parts and 225VA toroidal transformer. Each **$125**

KS-3TA: SERVO 100, DC 100 TOROIDAL TRANSFORMER, 500VA ONLY. Each **$120**

KS-3TB: SERVO 100, DC 100 TOROIDAL TRANSFORMER, 225VA (mono). Each **$75**

KS-1M: COMPLETE BORBELY SERVO 100 MONO AMP with KS-3PB POWER SUPPLY. [1:84] Each **$275** Two or more, Each **$260**

KS-1S: BORBELY SERVO 100 STEREO AMP with KS-3PA POWER SUPPLY. Each **$450**

KS-3M: BORBELY DC 100 MONO AMP with KS-3PB POWER SUPPLY. [2:84] Each **$285** Two or more, Each **$270**

KS-3S: BORBELY DC 100 STEREO AMP with KS-3PA POWER SUPPLY. Each **$470**

REG WILLIAMSON

KK-13CH: WILLIAMSON 40/40 [4:79] Complete power amp, two channel without power supply. Toshiba outputs. Each **$170**

KK-13P WILLIAMSON 40/40 [4:79] Power supply for two channels. No heatsinks or transformer. Each **$35**

KA-2T: AVEL-LINDBERG 40/3018 TRANSFORMER for the 20/20 or 40/40. Toroidal with flying leads, 50V@1.8A. Each **$65**

BOAK-JUNG-AMER

KM-8: ST-150-BJ-1 DYNA 150 MOD. [2:81] All parts except power supply kit, (see KL-1D), two channels, all resistors and capacitors for the amplifier circuit mods. Each **$90**

POWER AMPLIFIERS

JAMES BOAK

KJ-8A: DYNACO MARK III POWER SUPPLY MOD. [1:78] All parts needed to add a solid state regulated power supply to the Dynaco Mark III, including heatsink. The kit also includes all parts necessary to add a balance and adjustable, independent bias control for the Mark III. Each **$38**

KL-1P: BOAK REGULATED POWER AMP POWER SUPPLY. Pass Class A version. [1:80] All parts except large, flat heatsink for either positive or negative half of Pass power supply. Two required for one two-channel Pass amp (4:78). Board included. May be built as either (+) or (−) supply. Requires Pass DC supply components. Each **$85**

KL-1W: BOAK REGULATED POWER AMP POWER SUPPLY. [1:80] Williamson 40/40 or 20/20 amplifiers. All parts except large, flat heatsink. Positive supply. Use Pass power supply (KJ-5-4) for best results and increased power output. Each **$75**

KL-1D: BOAK REGULATOR KIT [2:81] Revised regulator for use with ST-150 MOD KM-8. All parts for regulator including new board. May be built as either positive or negative. One channel only. Includes heatsink. Each **$100** Four for **$320**

NELSON PASS

KJ-5-7: CLASS A 40W AMP A40. [4:78] This bipolar design from Nelson Pass of Threshold Corporation is as rugged as it is clean. Kit contains two boards (3" x 3") and all parts for two channels, including eight heatsinks and all parts for one ± 44V @ 8A (KJ-5-4) stereo power supply.
Each **$465**
West of Rockies, Each **$475**

PASS-CITATION MOD

KM-7: MOSFET CITATION 12 MOSFET MODIFICATION. [2:81] Complete kit, two channel, all parts to modify Harman-Kardon's Citation 12. Each **$170**

K. LANG

KV-2S: LANG CLASS A MOSFET POWER AMP. [2:86] A compact 20W per channel design from Germany which utilizes 4 push/pull power MOSFETs. Especially well-suited for multi-amp systems. Kit includes two circuit boards, all parts, two power supplies (± 24V @ 100mA; ± 15V @ 8A) with toroidal transformers and article reprint. Stereo pair. Each **$285**

KV-2DM: LANG CLASS A MOSFET POWER AMP. [2:86] Includes two amplifier channels and two each of the input and output power supplies for dual-mono operation (each channel independently powered). Each **$350**

CROSSOVERS

*For KC-4A choose **ONE** frequency from those listed below. For KC-4B choose **TWO** frequencies from the list. NO OTHER FREQUENCIES ARE AVAILABLE FOR STOCK KITS.*
60, 120, 240, 480, 960, 1920, 5k or 10kHz.

KC-4A: ELECTRONIC CROSSOVER, KIT A. [2:72] Single channel, *two-way*. All parts and C-4 circuit board. Includes new LF351 ICs. Suitable supplies include KE-5 and KF-3. Each **$14**

KC-4B: ELECTRONIC CROSSOVER, KIT B. [2:72] Single channel, three-way. All parts and C-4 circuit board. Includes new LF351 ICs. Each **$18**

KK-6L: WALDRON TUBE CROSSOVER: Low pass. Single channel, 18dB/octave, Butterworth, [3:79] includes three-gang pot, level control, 12AX7 tubes, board and three frequency range determining capacitors. *Specify ONE frequency range per kit please.* (Hz.): 19-210; 43-465; 88-960; 190-2100; 430-4650; 880-9600; 1900-21,000.
Single channel, Each **$60**

KK-6H: WALDRON TUBE CROSSOVER: High pass. Single channel, 18dB/octave, Butterworth, [3:79] includes three-gang pot, level control, 12AX7 tubes and three frequency determining capacitors. Please specify **one** of the frequencies above. No other can be supplied. Each **$62**

KK-7: WALDRON TUBE CROSSOVER POWER SUPPLY. [3:79] All parts, including board, transformer, fuse, semiconductors, line cord, capacitors. Will power four tube crossover boards (8 tubes), one stereo bi-amped circuit. Each **$110**

SBK-A1: LINKWITZ CROSSOVER/FILTER. *Speaker Builder's* [4:80] first kit, including all parts and board for one channel of the three-way crossover/filter/delay. 24dB/octave at 100Hz and 1.5kHz and 12dB/octave below 30Hz, with delayed woofer turn-on. Board is 5½ × 8½". Requires ± 15V supply, not included. Use the Sulzer supply KL-4A with KL-4B or KW-3. Per channel **$75** Two channels **$140**

SBK-C1A: ELECTRONIC CROSSOVER. [SB 3:82] 30Hz filter adapted as a single channel, two way crossover. Can be 6, 12 or 18dB per octave. Includes WJ-3 (F-6) PC board, 4136 IC, quality parts. Choose frequency of 60, 90, 120, 250, 500, 1k, 2k, 5k or 10k. Power supply required. Can use the KL-4A/4B or KW-3. Each **$32**

SBK-C1B: THREE WAY, SINGLE CHANNEL CROSSOVER. [*SB* 3:82] Contains two each SBK-C1A. Choose high & low frequency. Each **$60**

SBK-C1C: TWO CHANNEL, COMMON BASS CROSSOVER. [*SB* 3:82] Contains two each SBK-C1A. Choose one frequency. Each **$64**

SBK-C2: BALLARD ACTIVE CROSSOVER. [SB 3,4:82] Three-way crossover with variable phase correction for precise alignment. Kit includes PC board ($5^3/_8$ x $9^1/_2$"), precision resistors, polystyrene & polypropylene caps, crossover point at 400Hz & 35kHz. Requires ± 15V DC power supply—not included. Can use KL-4A/4B or KW-3.
Two channel **$150**

CD PLAYERS

KW-1: MAGNAVOX CD PLAYER MODIFICATION. Improves frequency response. Includes two Signetics NE5535, two Panasonic HF series 330μF capacitors and four 3.92k, 1% metal film resistors. Each **$12**

KW-2: MODIFICATION. As above, but with two AD-712 op amps in addition to the NE5535s. Each **$16**

KX-1A: DISC STABILIZER. Set of 3 Mod Squad Tiptoes, 3 Sorbothane feet and 25 AudioQuest CD Rings. Each **$70**

KY-1: BEERS' BUDGET CD MOD. [1:89] Kit provides POOGE-4 improvements without additional wiring or circuit boards. Complete parts for assembling amplifier modules and replacing DAC components. Article reprint included. Soldering skills required, not recommended for beginners. Each **$95**

CDPV/2 PIERRE VERANY TEST DISCS. More than 100 demo tracks and equipment tests are contained on this 2 CD set. Detailed instruction booklet is included. In addition to list price, please add $2.25 shipping and handling. Set **$34**

MIXING AND RECORDING

KB-2R: THE 4 + 4 MIXER. [2:71] All parts for eight mikes/inputs plus two extra faders for two added line-only inputs. Rotary fader pots, ICs & sockets and PC board included. Each **$90**

KB-3: 4 + 4 MIXER POWER SUPPLY. + 12V DC, all parts. Each **$16**

KF-2: GATELY EQUALIZER [2:75]. Single channel, three-section equalizer. EG-1 circuit board, all parts including pots and prime LM301s. *Panel not included.* ± 15-18V power supply required.
Single channel kit **$88**
Two kits, as above **$164**

KF-3: GATELY POWER SUPPLY. Regulated ± 18V, 100mA per side. EG-2 circuit board, all parts, includes transformer and heatsink. Each **$52**

KH-1: THE WILLIAMSON SUPER QUADPOD. [1:77] Ambience decoder or encoder following the Blumlein system for recording. Parts for one four-channel board. 25V DC supply needed. Each **$38**

FILTERS & SPEAKER SAVER

KF-6: 30Hz RUMBLE FILTER. [4:75] Two channel universal filter card supplied with WJ-3 (F-6) circuit board and all basic parts, 1% metal film resistors and 5% MKM capacitors for operation as an 18dB/octave 30Hz rumble filter. Kit may be adapted as two- or three-way single channel crossover with added capacitors and resistors. Each **$30**

KH-2: SPEAKER SAVER AND OUTPUT FAULT DETECTOR. [3:77] Two-channel kit provides turn-on and off protection. Fast optocoupler circuitry prevents damage from transients. Additional parts provided offer speaker protection in the event of amplifier failure. Kit includes circuit board, all board-mounted components, power supply parts and article reprint. Each **$65**

SYSTEM ACCESSORIES

KC-5: GLOECKLER 23-POSITION LEVEL CONTROL. [2:72] All metal film resistors, shorting rotary switch & two boards for a two channel, 2dB per step attenuator. Choose 10k or 250kΩ. Each **$48**

KG-1: GATELY PEAK OVERLOAD INDICATOR. [2,3:76] Detects signal peaks and warns of overloading levels with LED. Use with or without meter. Firing sensitivity is adjustable down to 170mV. Needs ± 15V from system. Two channels. Each **$16**

KH-8: MORREY SUPER BUFFER. [4:77] All parts & board for two channel output buffer to isolate tape outputs in your preamp from distortion originating in a turned-off tape recorder. Many uses for this versatile matchmaker. Requires power supply. Each **$22**

KL-2: WHITE DYNAMIC RANGE & CLIPPING INDICATOR. [1:80] One channel, including board, with 12 indicators for preamp or crossover output indicators. Requires ± 15V power supply @ 63 mils.
Single channel, Each **$58** Two channels **$110** Four channels **$198**

KM-10A: SWITCHBOX. [2:81] ($6^3/_8 \times 4 \times 2$") black with white letters with 5-position DP silver contact, steatite switch, five inputs, and one output with RCA jacks, nickel-plated, and five ground posts. Each **$45**

KM-10B: Same as 10A but with 12 gold-plated jacks. Each **$55**

KP-4: BOAK HEADPHONE AMPLIFIER. [3:82] Two-channel headphone amplifier with two regulated power supplies as in KP-4B. Limited quantities. Each **$195**

KP-4B: BOAK HEADPHONE VOLTAGE REGULATOR. All parts for 12V or 24V regulator for one channel. Does not include transformer, rectifier, or filter capacitors. Each **$26**

KP-5: AUDIO SWEEP MARKER ADDER. [2:82] All parts and circuit board (no power supply) for adding two adjustable markers to a swept audio signal, 20Hz-100kHz. Each **$32**

SBK-D1: NEWCOMB PEAK POWER INDICATOR. [SB 1:83] All parts & board. No power supply required. Each **$7** Two for **$11**

SBK-E2: NEWCOMB NEW PEAK POWER INDICATOR. [*SB* 2:84] All parts & board, new multicolor bar graph display; red, green & yellow LEDs for one channel. No power supply needed.
Each **$14** Two for **$22**

AIDS & TEST EQUIPMENT

KE-2: REGULATED POWER SUPPLY. [4:74] ± 15V @ 1.5A. Lab quality device but excellent for powering system components. Includes board, all board mounted parts plus two LM395K regulators. Transformer and filter caps not included. Each **$48**

KE-5: OLD COLONY POWER SUPPLY. ± 18V @ 55mA balanced, includes all parts, board and transformer. Not regulated. Each **$20**

KH-7: PRECISION 101dB ATTENUATOR. [4:77] All switches, 1% metal film resistors to build Gloeckler's prototype. Chassis, input/output jacks not included. Each **$65**

KJ-6: CAPACITOR CHECKER. [4:78] All switches, ICs, resistors, 4½" D'Arsonval meter, transformer and PC board to measure capacitance, leakage and insulation. Each **$90**

AIDS & TEST EQUIPMENT

KK-3: THE WARBLER OSCILLATOR. [1:79] Switches, ICs, transformer, and PC board for checking room response and speaker performance without anechoic chamber. Each **$70**

KL-3: INVERSE STEREO RIAA NETWORK. [1:80] Two channels, 1% polystyrene capacitors and metal film resistors, gold jacks, cast aluminum box, solder lugs and alternate 600 or 900Ω R_2'/C_2' components. Each **$45**

KL-6: MASTEL TIMERLESS TONE BURST GENERATOR. [2:80] All parts with circuit board. No power supply. Each **$24**

KM-3: CARLSTROM/MULLER SORCERER'S APPRENTICE/PAUL BUNYAN. [2,3:81] Includes six boards and all parts including power supplies. No chassis or knobs. Two article reprints. Each **$290**

KP-2: TWO TONE INTERMODULATION FILTER. [1:82] All parts, circuit boards, 1% resistors included. Each **$26**

SBK-D2: WITTENBREDER AUDIO PULSE GENERATOR. [*SB* 2:83] All parts, board, and power supply included. Each **$80**

SBK-E4: MULLER PINK NOISE GENERATOR. [*SB* 4:84] All parts, board, 1% MF resistors, capacitors, ICs, and toggle switches included. No battery or enclosure. Each **$35**

KV-4: HANSEN CIRCUIT-SAFE CHECKER [*CS* 1:86] All parts supplied, except test leads and battery. Each **$20**

PARTS

Tubes

Siemens 6DJ8	$ 9.50*
Siemens 12AX7	$ 9.50*
Mullard 12AU7	$ 7.00*
National 6CA7 matched pair	$32.00*
National 6L6GC matched pair	$28.00*
TRW socket SKT9R pc board mount, 9 pin	$ 2.15*

Integrated Circuits

Signetics NE5534	Each	$4.75*
Signetics NE531 (8-pin DIL)	Each	$4.25*
Signetics NE5535 (8-pin dual FET op amp)	Each	$3.70*
Signetics LF351N (8-pin op amp)	Each	$1.25*
Analog Devices AD-711 (8-pin DIL single FET op amp)	Each	$2.40*
Analog Devices AD-712 (8-pin DIL dual FET op amp)	Each	$3.00*
Motorola MC34081P input op amp	Each	$2.50*

Connectors

SCXT7: ROYCE AUDIO PLUG. RCA type phono plug custom made for Old Colony. Five part construction with excellent strain relief. Heavy 24K gold plate, accepts cable diameter up to 0.23".
Pair **$18*** Two or more pair Each **$17***

SCXT8: ROYCE AUDIO JACK. Counterpart to SCXT7. Mounts from front of panel (up to ³⁄₁₆" thick, ¹⁄₁₆" if with insulators) in ⁵⁄₁₆" hole. Nylon insulators are included. Pair **$16***
Two or more pair Each **$15***

PHONO JACK A. Mounts in ⅜" hole from rear of panel (up to ¹³⁄₆₄" thick). External hex nut ensures tight installation. Gold-plated hardware included. Pair **$7.50***

NYLON INSULATING WASHERS. One flat/one shoulder, 10 pairs per set. ⅜" size—Fits Phono Jack A **$1.50***
¼" size—Suitable for ¼" phono jacks **$1.50***

SB7550B: PHONO PLUG. Gold-plated, fully shielded. Features spring strain relief. Accepts cable diameter up to .24" (such as Neglex 2534).
Pair **$7.50***

SCBPG: BRASS GOLD-PLATED BINDING POSTS. Red and black. 30A, 1000V AC, five-way. Pair **$6.50***

SCBNG: GOLD-PLATED BANANA PLUGS. Stackable, beryllium copper type. Leads held by internal set-screw. Red and black.
Pair **$6.50***

***DISCOUNT: $25-$75 = 10%, $75 and up = 15%**

PARTS

Connectors

SCSLG: GOLD-PLATED SPADE LUGS. For ¼" post, accepts 10-12 gauge wire. Solder or crimp. Pair **$1.50***

INDIUM PLATED SCREWS. 10/32 x ½"Indium over chrome over brass. Indium provides superior electrical power contacts on large electrolytic terminals (POOGE-2, 4:81). Four **$4.75***

& Cable

518: APATURE SPEAKER CABLE. This heavy 12-gauge oxygen-free copper, linear crystal cable has an ultra flexible clear jacket. Terminate with SCBNG or SCSLG. Twin Lead, per foot **$1.60***

2534: NEGLEX AUDIO CABLE. Low capacitance, high performance interconnect made with OFHC wire by Mogami. Copolymer insulated with spiral shield. Available in blue, black or yellow (specify with length). Per foot **$1.25***

2477: NEGLEX SPEAKER CABLE. Low impedance, high definition cable made with Mogami OFHC wire and copolymer insulation. Per foot **$2.50***

TK2477: TERMINATION KIT. For 2477 cable, includes four gold-plated spade lugs and insulating sleeve. Per pair **$2.50***

2515: NEGLEX HOOK-UP WIRE. Oxygen-free copper, super flexible, 18-gauge with cross-link polyethylene insulation. 25 foot spool in red or black. Specify. Each **$10.00***

Potentiometers

Alps offers the finest in precision rotary controls. All mounting hardware is included.

VR-50K: Stereo volume control, no detents. Two sections matched within ± 2dB. Size: body 25H x 25.3W x 27D (pins project 5mm above height measurement). Shaft length 13mm beyond mounting bushing. **$32**

VR-100KB: Stereo volume control, 41 detents, log taper, tracking error less than 2dB. Dimensions as above. **$32**

VR-100KJ: Stereo volume control, log taper, 31 detents. Taps at 14k for a "loudness" control. Heavy cast aluminum 34 x 33.5 x 40mm case. **$22**

VR-100KD: Balance control, single center detent. **$32**

Et cetera

MSLD-32: OLD COLONY SOLDER. Manufactured by Alpha Metals, our product features a highly activated organic flux (2.2%) with a water-soluble resin derivative of rosin which allows post soldering residues to be removed with a water wash. An article reprint, *Soldering: The Basics* by Marc Colen is included. 63/37 Tin/Lead, .032" diameter. 30 foot spool is 2 oz. Each **$7.50***

KM-12: RESISTOR SAMPLE DESIGN PACK. Metal film resistors, ¼W, ½W, ± 1% tolerance. A total of 119 resistors in 31 of the most used values. **A $25 value.** Only **$15***

KM-11: RESISTOR SENIOR DESIGN PACK. Metal Film Resistors, ¼W, ½W, ± 1% tolerance. 108 values (total E24 series) distributed according to Old Colony's customer preferences over 18 months: 31 packs of 3 each; 28 packs of 5 each; 33 packs of 10 each; 12 packs of 15 each and 3 packs of 20 each. A total of 803 resistors. Regular value $165. Each **$125***

KM-6: CRAMOLIN CONTACT CONDITIONER. Kit consists of one two-dram vial of Red for old contacts, one two-dram vial of Blue for new contacts, lint-free applicators and full instructions for use. This military grade contact cleaning compound dissolves and removes oxides and their effects on all non-soldered contacts in audio systems from cartridges to speaker terminals. Each **$20***

KM-9: WILLIAMSON RECORD CARE KIT. Deep cleaning and static removal treatments. (2,4:81) Contains ½ ounce of de-static liquid and 4 ounces of powder for making up coating cleaner. User needs isopropyl alcohol and distilled water as well as glycerin and a surfactant to make up the needed treatments. Full instructions included with each kit. De-static treatment for 1,000 disks, deep cleaner for 80 disks. We regret that due to postal regulations, this cannot be shipped outside of the U.S. Each **$12***

KM-9R: CLEANER REFILL. 4 ounces, Each **$6.95***

KM-9S: DE-STATIC LIQUID. refills ½ ounce. Each **$6.95***

HDHFT: HI-FI TIPS. Imported for Old Colony. Solid brass ⅞" high conical feet for components and loudspeakers. Includes self-adhesive pad. Each **$3.00**
10 or more Each **$2.50**

***DISCOUNT: $25-$75 = 10%, $75 and up = 15%.**

OLD COLONY'S KIT ORDER FORM

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AN EASY IC POWER AMP/CROSSOVER

BY RALPH GONZALEZ

While somewhat out of fashion with audiophiles, IC power amps can be the perfect solution for some applications. They offer wide bandwidth and low noise and hum, in extremely simple, compact, and inexpensive circuits. These are ideal for speaker builders who want to get their feet wet in the construction of active circuits, cheaply and painlessly.

The power amp design I present may be used specifically for adding a subwoofer to a low-power stereo or to a video system, although it can also be used full-range. It includes a gain control, low-pass filter (optional), and channel-combining stage (optional). For example, you can use it to tap a signal from the left and right speaker outputs of your television or receiver, and power a single 8-12-inch subwoofer. This won't draw any additional power from your television or receiver outputs. If you use the speaker outputs, be sure the two channels have a common ground. To check this, use an inexpensive multimeter to ensure the resistance between the left and right "minus" speaker terminals is near zero ohms. Of course you can also use line-level sources.

The only active components are two LM383 high-current IC power amps, available from Radio Shack for $2.99 each. These are popular in automotive applications, and are rated at 8W into 4Ω, when properly heatsinked. I've bridged them as suggested in Radio Shack's *Semiconductor Reference Guide*, to provide 16W continuous output, although somewhat higher short-term power may be available.

As shown in *Fig. 1*, Radio Shack recommends capacitors C1 and C2 to avoid supersonic oscillation. Possibly due to

ABOUT THE AUTHOR

Ralph Gonzalez has a B.S. in mathematics from the University of Delaware and is completing his Ph.D. in systems engineering at the University of Pennsylvania. He and his wife, Maureen, live in Delaware and he is an assistant professor of computer science at Rutgers University. In addition to his audio interests, Mr. Gonzalez started Delaware Acoustics, a fledgling speaker company, and plays fretless bass guitar.

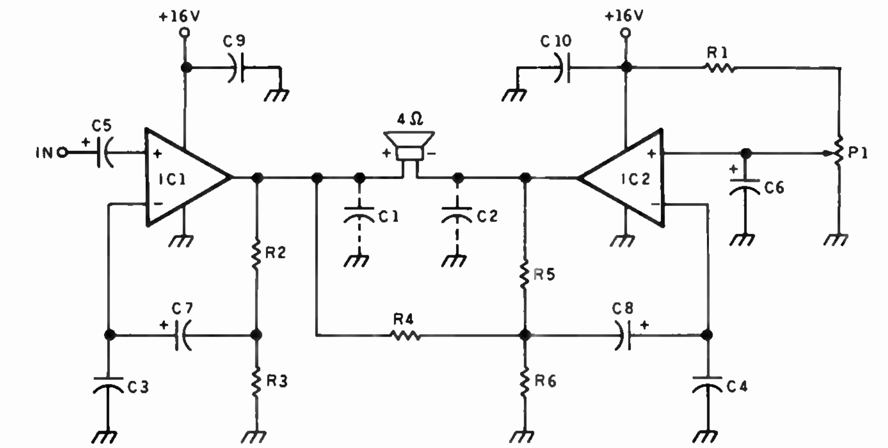

FIGURE 1: Schematic (from Radio Shack).

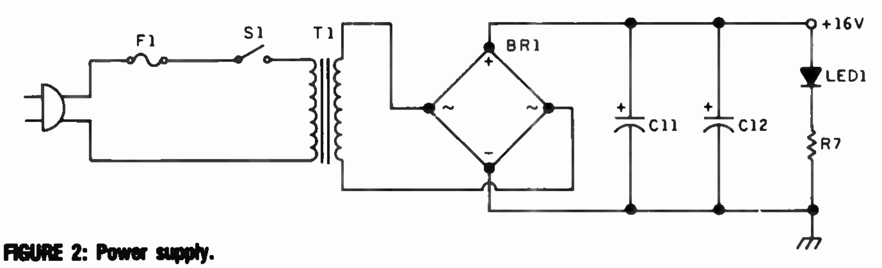

FIGURE 2: Power supply.

PARTS LIST

R1	1MΩ
R2, 4, 5	220Ω
R3, 6	2.2Ω
R7	2-5kΩ
R8, 9	4.7kΩ
C1, 2	0.2μF (see text)
C3, 4	0.2μF ceramic or film (see text)
C5, 6	10μF electrolytic
C7, 8	470μF electrolytic
C9, 10	0.2μF ceramic or film
C11, 12	4,700μF/20V electrolytic
C13	1μF NP electrolytic or film (see text)
P1	100kΩ PC board potentiometer (linear taper)
P2	5kΩ panel-mount potentiometer (audio taper)
T1	12V/3A power transformer (discard center-tap)
BR1	50PIV/4A full-wave bridge rectifier
IC1, 2	LM383 audio amplifier IC
F1	3A fast acting fuse
S1	SPST switch

Misc: TO-220 heatsinks and grease, fuseholder, hook-up wire, LED, pre-drilled general purpose PC board and standoffs, feedthrough barrier strip, knob, power cord, metal cabinet, vinyl grommets.

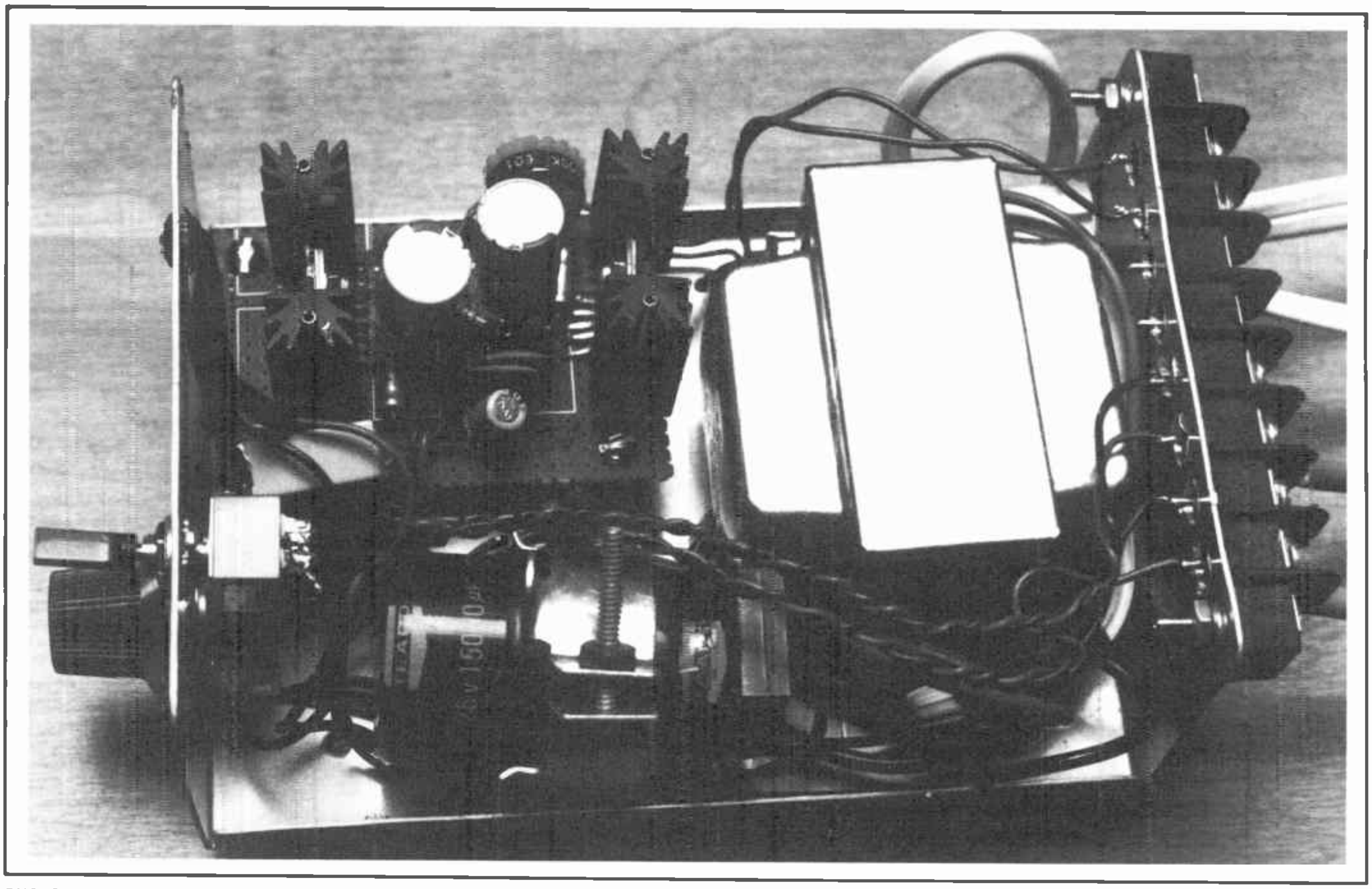

PHOTO 1: A simple IC power amp.

my inexperience with circuit layout and grounding, I had to *remove* these and add capacitors C3 and C4 to prevent oscillation. Oscillation is a plague in designing active circuitry, and can best be avoided by keeping inputs and outputs physically separated, by keeping component leads short, by referring all grounds to a single physical point ("star" grounding), and by adding small capacitors at the ICs' V+ and V− terminals as shown. If you don't have an oscilloscope, you can tell your circuit is oscillating because the heatsinks will get very hot, quickly. Disconnect power immediately if this occurs.

Use your multimeter to adjust P1 for minimum DC over the speaker terminals, and to test the power supply, shown in *Fig. 2*.

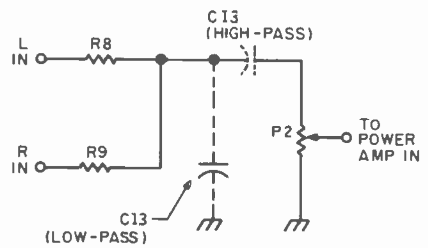

FIGURE 3: Crossover—with channel summing and volume control.

Figure 3 shows a simple volume control/channel-combining/low-pass circuit for a subwoofer operating below 95Hz. This circuit can also be used with a different power amp, though you'll have to adjust the capacitor value if the amp's input impedance is less than about 50kΩ.

Figure 4 shows the pin configuration for the LM383.

You can also use pairs of these amplifier/crossover combinations to experiment with active crossovers in your stereo system. To change the crossover frequency (F), recalculate C13 in *Fig. 3* from:

$$C13\ (\mu F) = 95/F$$

To make a first-order, *high-pass* crossover, place C13 in *series* with the signal as shown in *Fig. 3* and use:

$$C13\ (\mu F) = 21/F$$

To use the circuit with only a single input, short the unused one to ground. Finally, to use it full-range, eliminate C13 altogether.

You can get all the parts from Radio Shack for about $35. If you parallel four 10Ω resistors, you can get near the 2.2Ω value. You can use RS #276-1363 for C11 and C12, or substitute a single 10,000μF or larger cap if you can find it in Radio Shack's capacitor assortment (20V or higher). Make sure all electrolytic capacitors are used with the correct polarity, as shown. For better heatsinking, use pairs of heatsinks back-to-back, with heatsink grease to aid conduction.

Plastic Package

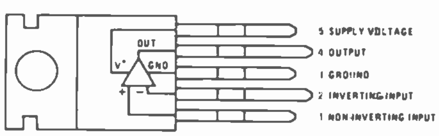

FIGURE 4: Pinout configuration—LM383.

I soldered most of the circuitry onto a small pre-drilled circuit board. *Photo 1* shows how everything can be squeezed into a very compact box (RS #270-252). Good luck!

RESULTS. My father is using this miniature amp/crossover to power a 15-inch subwoofer for his Sony stereo television. Since the bass of the television's detachable speakers is so puny, the subwoofer effect is dramatic. This gives a "movie theater" feel to shows like *Star Trek* (you can now hear the ship's engines), makes sporting events more believable, and gives music true "weight."

Continued on page 71

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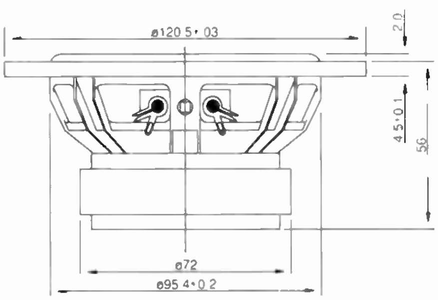
MP 12 VC

4.5" High Fidelity midrange unit

Chassis: glassfibre reinforced plastic, black
Surround: soft PVC, black
Cone: polypropylene, black
Dust Cap: soft PVC, black
Mounting holes: 4 x 5 mm, equispaced on PCD 111.5 mm.



The MP 12 VC bears some resemblance with the famous 11 F -M, but is a completely new midrange unit. The chassis is made by a specially developed moulding tool which offers the availability of custom-made integral chassis fronts at a reasonable tool cost.

The cone, the surround and the dust cap have been made from plastic material with excellent dynamic behaviour.

For this reason no additional coating of the cone is necessary, and an improved consistency between units has been achieved.

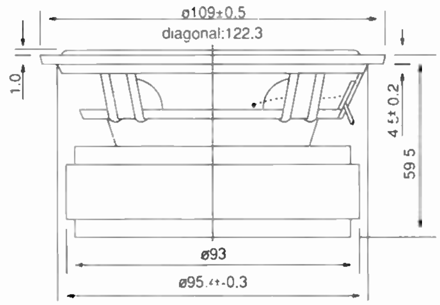
Due to its regular, smooth frequency response MP 12 VC may be used with crossover networks of low complexity, and will produce a clean, natural sound.

Recommended frequency range	300-5000	Hz
Nominal power (DIN 45573)	80	W
Characteristic sensitivity (lm, lw)	89.5	dB SPL
Operating power (DIN 45500)	4.5	W
Voice coil diameter	26	mm
Flux density	1.0	T
Recommended enclosure volumes		
Closed cabinet	1.5-3	litres
Weight	0.65	kg
Magnet weight	0.25	kg
Effective diaphragm area	62	cm
Free air resonance	110	Hz
Qts	0.58	

P 11 RCY

4.5" High Fidelity mini woofer

Chassis: magnesium, injection moulded, black
Surround: rubber
Cone: polypropylene, black
Dust Cap: soft PVC, black
Mounting holes: 4 x 5 mm, equispaced on PCD 139 mm.



The P 11 RCY is based upon the legendary 11 F- GX. The cone and the dust cap have been made from plastic materials with excellent damping properties.

Consequently, the sound quality and the consistency between units have been markedly improved.

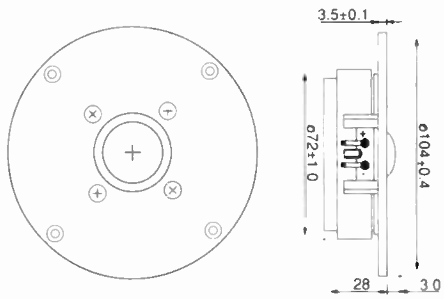
P11RCY may be used in very small two way systems producing an astonishingly deep bass and a clean, neutral midrange sound.

Recommended frequency range	45-4000	Hz
Nominal power (DIN 45573)	50	W
Characteristic sensitivity (lm, lw)	86	dB SPL
Operating power (DIN 45500)	10	W
Voice coil diameter	26	mm
Flux density	1 25	T
Recommended enclosure volumes		
Closed cabinet	2.5-4	litres
Bass reflex cabinet	3-6	litres
Weight	1 1	kg
Magnet weight	0 42	kg
Effective diaphragm area	55	cm
Free air resonance	50	Hz
Qts	0 24	

25 TF

4.5" High Fidelity Dome Tweeter

Chassis: glassfibre reinforced plastic, black
Diaphragm: soft dome, textile, coated
Mounting holes: 4 x 4.5 mm, equispaced on PCD 92 mm.



The 25 TF is a soft dome tweeter where the diaphragm material has been coated on the rear side prior to the forming operation.

This gives a vast improvment in consistency compared to the normal coating methods.

The diaphragm and coating materials show a very high degree of stability against changes in air temperature and humidity.

Recommended frequency range	3000-25.000	Hz
Nominal power (DIN 45573)	50	W
Characteristic sensitivity (lm, lw)	91.5	dB SPL
Operating power (DIN 45500)	2.8	W
Voice coil diameter	26	mm
Flux density	1.8	T
Effective diaphragm area	7	cm
Free air resonance	1000	Hz
Magnet weight	0 25	kg

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EASY SURROUND SOUND

BY LOUIS EDELMAN

Now that you have finally become aware how much weight, impact and solidity a four-channel, surround-sound setup can provide, don't degrade the quality of your stereo system by adding distortion-generating rear-channel processors, and image-blurring, muddying auxiliary speakers.

Axiom: As stated and demonstrated by Linn Products, any extra speaker-type transducer in a listening room will cause a muddy reproduced sound. So how can you get surround sound without extra speakers? You can't—the extra speakers are required, of course, but you must take steps to gain their benefits without allowing their presence in the listening environment to cause too much sound quality degradation.

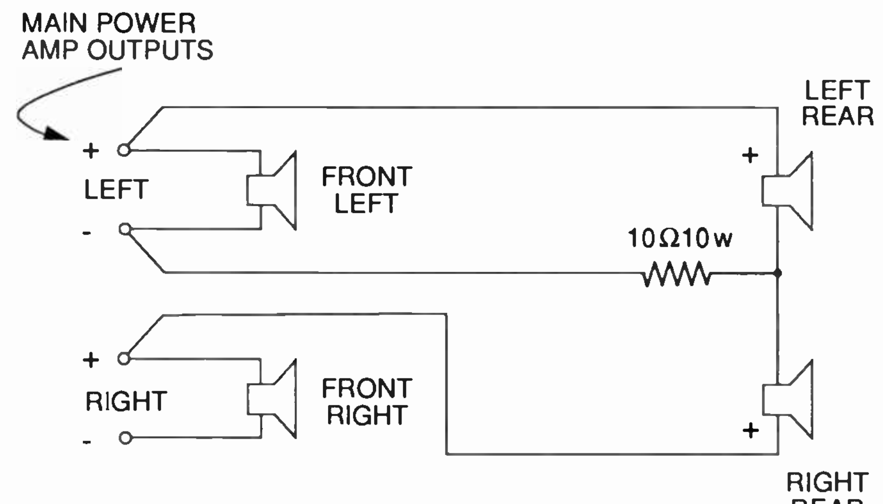

FIGURE 1: The Dynaquad "matrixing" arrangement.

SPEAKERS. Smaller sized extra speakers are less likely to muddy the sound. Therefore, instead of a bookshelf-sized, or larger, pair of rear-channel speakers, use the smallest high-quality speakers you can find. The ideal rear-channel speakers might therefore be a pair of top-quality 5-inch or smaller, automotive type speakers.

You could place these speakers on the floor near the corners of the listener end of the room, or build tiny boxes to dress up their appearance. Baffle-type mounting is unnecessary, since any bass they produce would probably detract from the system's overall sound quality (I assume it is good to begin with). The midrange and high end of these extra speakers must be high quality, however, but it does not have to be quite as good as that of the main speakers. They will still be capable of adding all the extra space and dimensionality you are hoping to attain, if they are reasonably good quality.

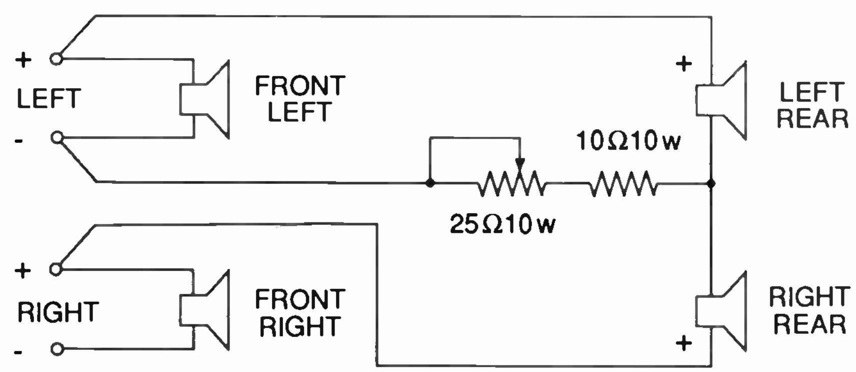

FIGURE 2: Adding a level control.

EASY SURROUND SOUND. First, consider the stereo component system you have strived so hard to improve, and perhaps in which you have invested a great deal of money. Since all currently available surround-sound processors add noise and distortion to your system, why even consider spending more than a thousand dollars to add noise and distortion to your system? You don't need to throw away even a few hundred dollars in attempting to achieve a satisfactory four-channel sound setup.

You can obtain the rear-channel signals the easy way, using the old Dynaquad matrixing arrangement. Connect the power amplifier's positive output terminals of the left and right channels to the positive terminals of the left and right rear-channel ambience speakers; and connect the negative terminals of the rear-channel speakers together, as shown in *Fig. 1*.

Tie their junction point to one end of a 10Ω, 10W power resistor, and connect the resistor to either one of the power amplifier's ground (negative) terminals, as shown. Now, sit back and listen.

EFFICIENCY. If your rear-channel units are more efficient than the main speakers, or seem too loud when heard from a normal listening position, add a 25Ω, 10W potentiometer in the ground return of the rear-channel drivers, as shown in *Fig. 2*. Simply connect it like a rheostat, as shown, and adjust the level of the rear-channel speakers just enough to add a full-bodied, airy, more spacious quality to the sound. If your attention is directed toward the ambience speakers themselves, they are too loud.

AMBIENCE AMP. If your rear-channel speakers aren't as efficient as the main ones, use an extra power amp to drive them. Use the outputs from the front-channel's power amplifier, as shown in *Fig. 3* to feed the inputs of a rear-channel amp.

Take an ordinary interconnect cable with RCA phono plugs and cut it in half. Extract the inner conductor from the shield at the cut end, and connect it to a positive output terminal of the main power amplifier. Connect the braided shield of the cable's cut end to the negative terminal of the same channel. If the interconnect cable that you have selected has *two* inner conductors and a separate shield, use the two inner conductors as the signal leads, and cut off the braided shield. Do the same with the other half of the cut interconnect cable, to provide

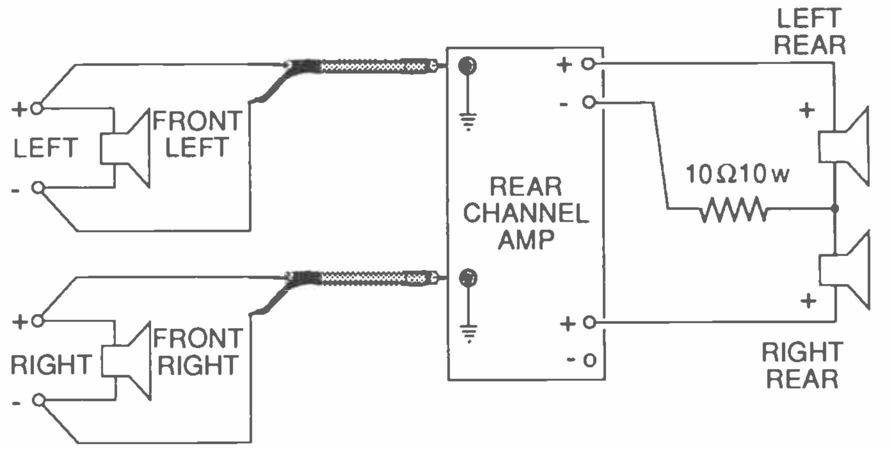

FIGURE 3: Adding an extra power amp.

the rear channel amp's other input. Be sure the negative terminal is connected to the ground side (outer conductor) of the phono plug.

If the rear-channel stereo amplifier is located any distance from the front-channel power amplifier, you can run ordinary lamp cord or small diameter speaker wire from the main amplifier's output terminals to the doctored interconnect cables.

The type of wire you use for this connection is not at all critical. The outputs of the main amplifiers, driving the inputs of the rear-channel amplifiers, act as an almost perfect, near-zero source impedance generator, driving a very high impedance load. This results in almost perfect transfer of a signal, and adds no extraneous noise. Therefore, the signal cable going to the rear-channel amplifier inputs can be plain old small-gauge zip cord, connected to cheap phono plugs, even if your system employs something as exotic as MIT's Music Hose or other fancy, expensive speaker leads and interconnects.

Do the Dynaquad matrixing at the outputs of the rear-channel amplifiers, right at their speaker connections, as shown in the figure. Actually, it seems silly to even refer to this uncomplicated connection arrangement as a matrix. Simply connect the positive leads of the rear-channel speakers to the positive leads of their amplifier channel outputs, and tie the speakers' negative leads together. Then connect this common tie point through a 10Ω (approximately) resistor to either ground connection at the rear-channel amplifier's output; the same arrangement as with the single-amplifier.

Now set the rear speakers' output levels with the input level controls. If the rear-channel amp doesn't have level controls, use a pair of 50kΩ potentiometers in a small junction box placed before the inputs, as shown in *Fig. 4*. Don't worry about the quality of the necessary phono jacks at this point; the circuit impedances are such that no signals will be degraded. Use quality potentiometers, however.

If you use two separate front-channel amplifiers, the hot signal leads coming from outputs should be isolated by send-

Continued on page 41

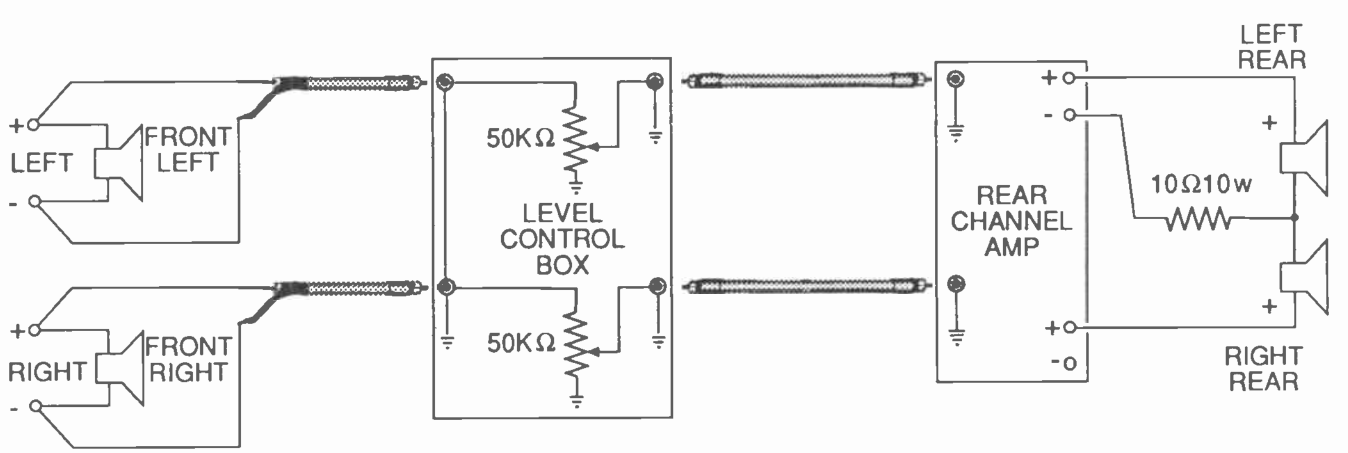

FIGURE 4: Adding a "black-box" level control for the rear amplifier.

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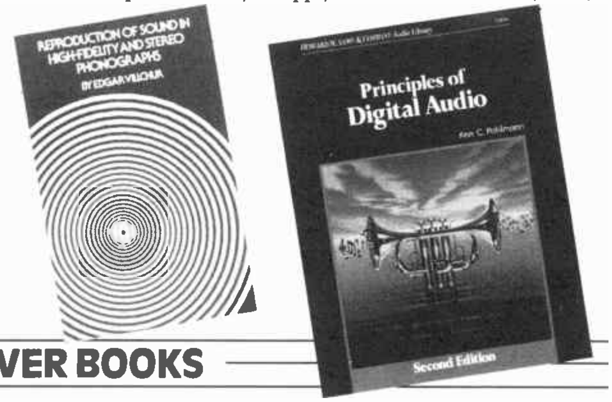

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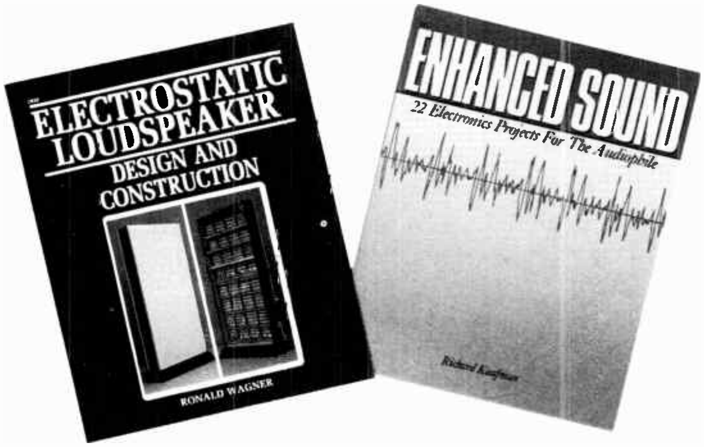

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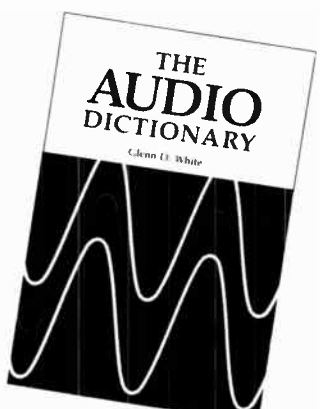

UNIV. of WASHINGTON PRESS

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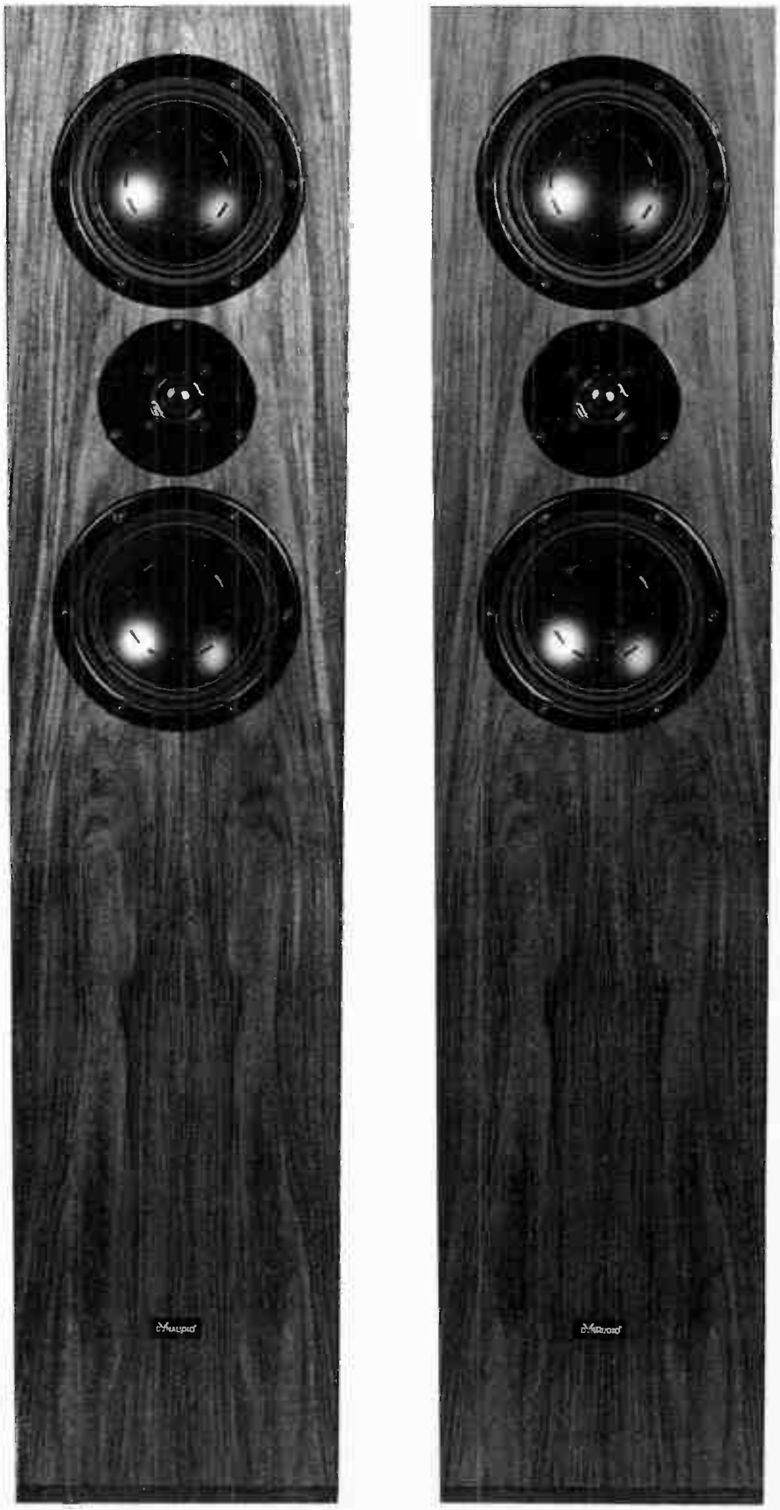
DYNAUDIO®

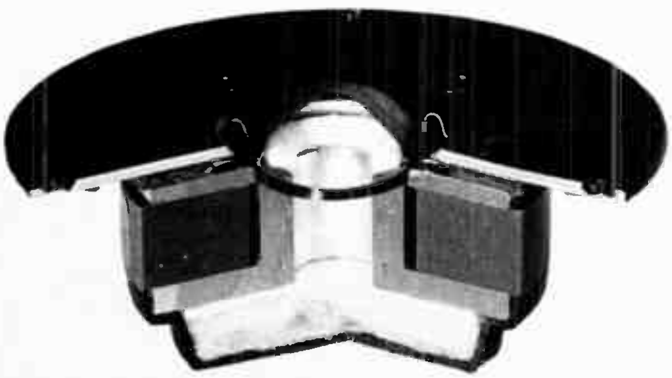
TECHNOLOGY UNLIMITED

The Twynn

The idea behind the TWYNN is the realization of a system utilizing the combination of our most popular models, the woofer-midrange 17 W-75 and the dome tweeter D-28 AF. These two units have achieved worldwide acclaim due to their extraordinary price to performance ratio. The result of these efforts is the TWYNN system that sets standards within its class due to avant-garde technology and detailed attention to sound fidelity.

As the name TWYNN implies, this DYNAUDIO System is equipped with two woofer midrange drivers arranged in a vertical array with the high frequency unit at the center point. Among the benefits of this arrangement, often referred to as the D'Appolito configuration, is stable three dimensional imaging, even when the listener is off avis from the speaker's radiating surface. Thus a listener who does not have an optimum hearing position will be able to distinguish a balanced and homogenous sound spectrum. In addition, the acoustically effective diaphragm surface is twice as large as traditional systems. This makes it possible to develop a slim cabinet design and still achieve superior pulse response.



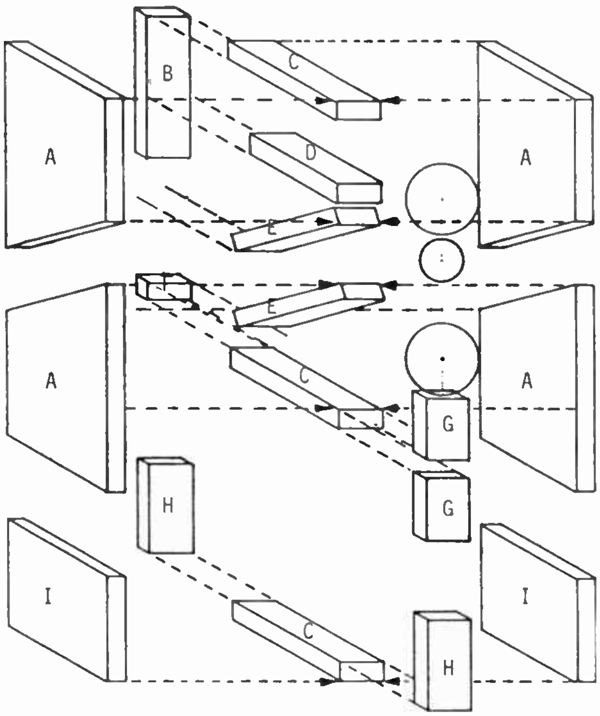


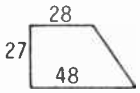
Closed cabinets with two chambers

The cabinet of the TWYNN has been designed for operation with the DYNAUDIO 17 W-75 as a closed system with optimum damping characteristics made possible through the use of the DYNAUDIO Variovent. Each woofer has a closed cabinet of its own for DC Coupling without internal woofer interaction. Moreover, these enclosure volumes are unequal, creating the advantage of a shifted resonance frequency of the two drivers and the ability to tailor the sound to precise musical standards.
The resonance frequency of the system is determined not only by the size of the cabinets, but by the internal damping properties. Here the amount, arrangement, and the type of damping material is important. Still, the careful builder should be able to accurately assemble the TWYNN from the directions which follow.

Three drivers for a two way system

The Dynaudio 17 W-75 woofer possesses a very large voice coil which makes possible outstanding drive characteristics. Deformation of the cone, a problem that often occurs with drive units possessing voice coil of small diameters, is effectively controlled.
Another advantage of the 17 W-75 is its frequency response in the upper transmission range. Above 2500 Hz its frequency characteristic possesses an extremely sharp roll-off of more than 48 db per octave. This eliminates the need for filters with sharp slopes and wide phase shifts. As a result, tweeter matching is not difficult and results in a homogenous blending of upper and lower frequencies. The D-28 AF with its relatively large diaphragm surface develops outstanding dynamic range even at 2500 Hz and complements the powerful woofer capabilities.

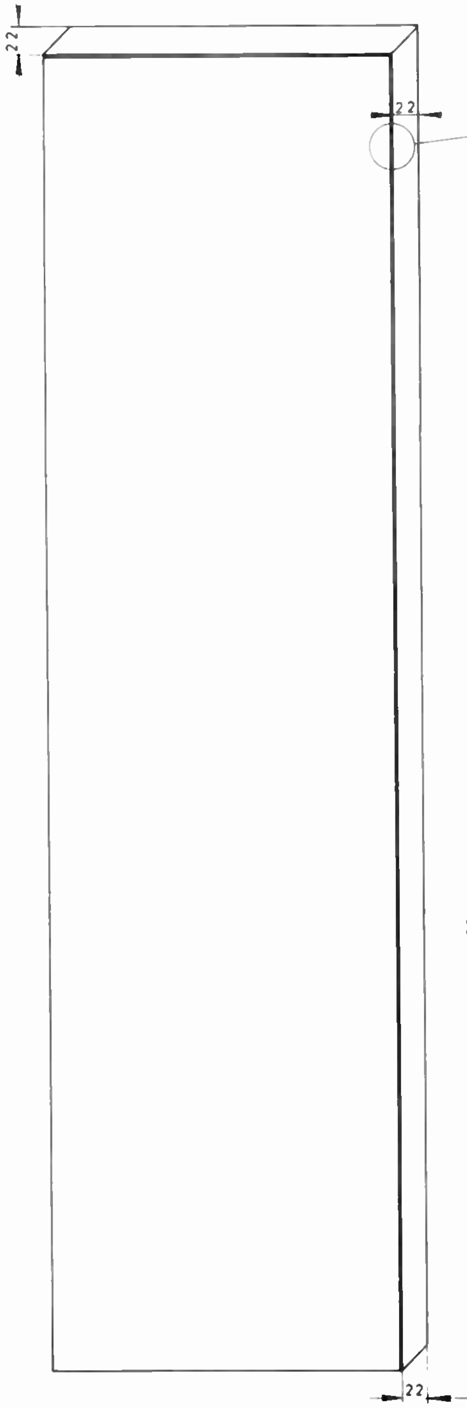


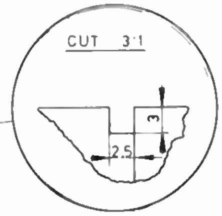


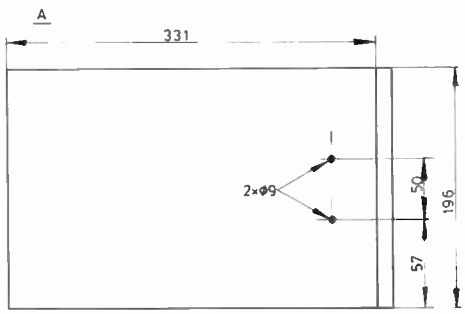
A:

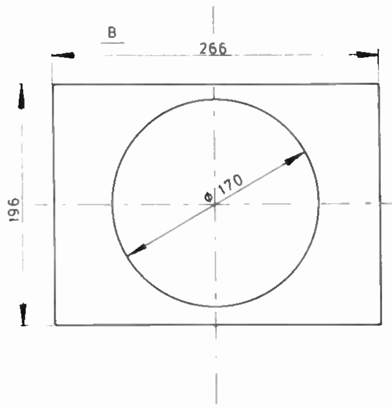
50mm ROCKWOOL

		E:	32 x 10
B:	37 x 10	F:	5 x 10
C:	27 x 10	G:	15 x 10
D:	22 x 10	H:	20 x 10
		I:	25 x 27



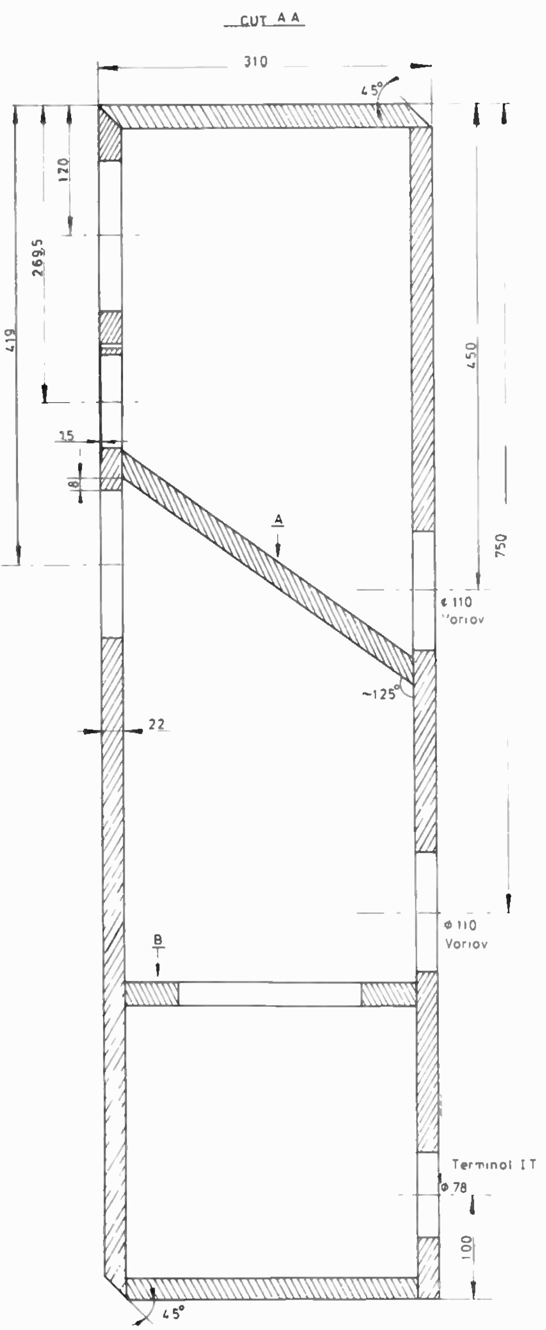


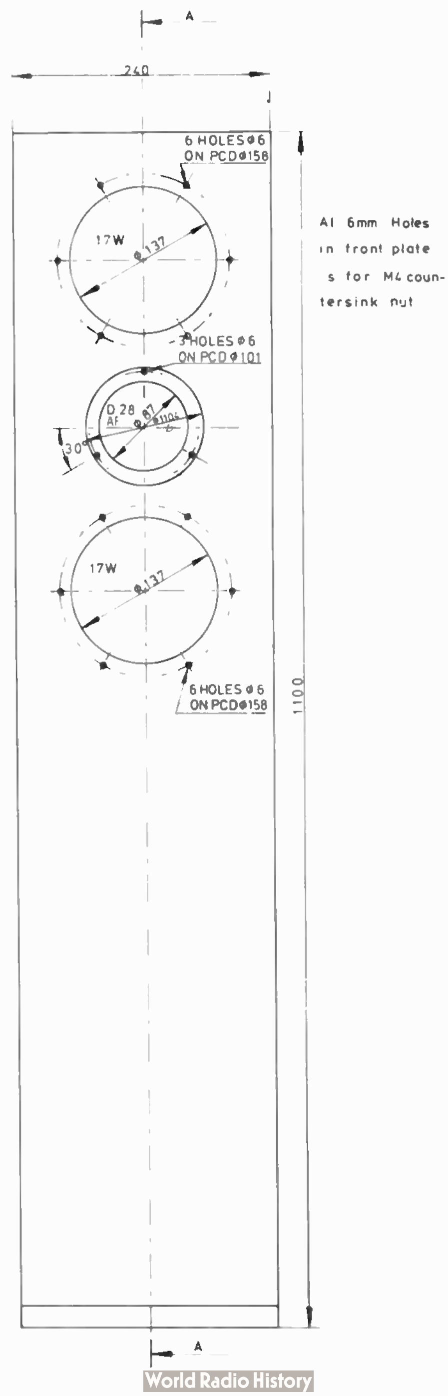




The cabinet

The standards the TWYNN sets from an acoustic and technical point of view is matched by the conception and execution of the cabinetry as well. The design of the entire cabinet was deliberately developed for problem free construction by kit builders, but careful assembly is a necessity.
DYNAUDIO has taken the careful finish of the cabinet surfaces into consideration when creating TWYNN cabinets at the factory. With these units the owner can choose from completely varnished cabinets, veneered cabinets both finished and unfinished. This last option allows a wide variety of finishes because the veneers possess a neutral grain. We suggest a few methods to illustrate the many possibilities.
1) Coat the cabinet with clear varnish that results in a light colored oak veneer. 2) Stain the neutral oak veneer with water stains to produce a wide range of colors. Here you may also select other colors apart from oak veneer colors. Reddish hues like mahogany, or pure colors like white, grey, or black produce interesting combinations. 3) Coat the cabinet in various colors with wood wax (for example, Miniwax) is especially easy and produces no odors and little cleanup. 4) Coat the wooden surfaces with colored varnishes. Depending of the primer, you can influence the strength of the visible grain of wood.





Perfection by self assembly

If you assemble your loudspeaker system yourself, you have additional opportunities for fine tuning. Each listening room has individual acoustic characteristics. On preassembled systems it is usually assumed that the room has average absorbtion levels. Therefore, it may well be that the tweeter sounds dull in a heavily damped room even though that was not the design intention. Conversely, rooms with significant reverberation produce the opposite effect. To solve this problem, we have integrated a method for adjusting the treble level of the filter network of the TWYNN the same method that is included in all DYNAUDIO kits. Changing the individual filter components produces emphasis or attenuation of the treble level with minimal influence on transient response. Thus the TWYNN can be optimized to all listening rooms. You may try and test the possibilities then choose the one that delivers the best results.

List of components for 1 pair of TWYNN loudspeakers

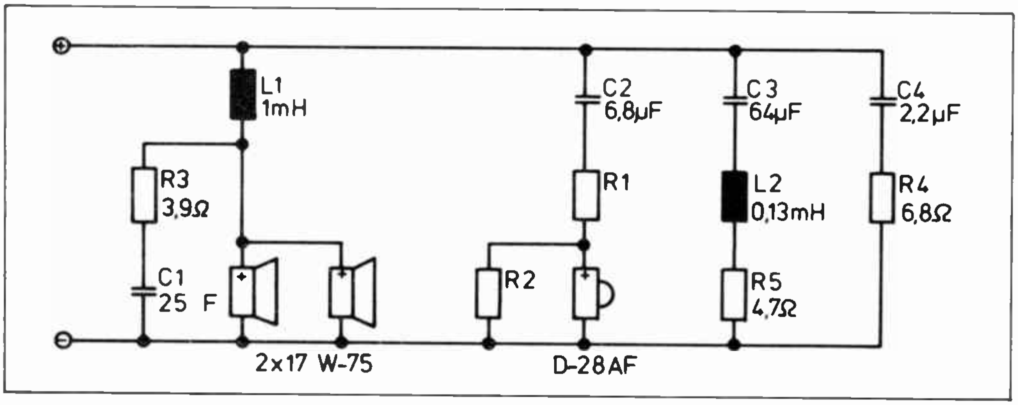
Woofer	4	17 W-75
Tweeter	2	D-28 AF
Crossover	2	DF TWYNN
Damping material	2	DD 1000
Accessory	1	DS TWYNN
Enclosure	2	Original TWYNN enclosures (completely assembled) or self-designed enclosures

DYNAUDIO

Tips for final assembly

The following suggestions assume that you have assembled the wood parts of the cabinet. All holes necessary for cables and terminals must be drilled. The plate nuts should be mounted as well.
First of all, cut the damping pads according to directions. The individual parts of the damping material can be keyed together inside the cabinet so no additional attachment is required. For damping the lower chamber it is useful to insert the damping parts through the cutout of the 17 W-75 and to adjust them to their respective positions through the cutout of the lower variovent. Next the crossover has to be mounted and the cables attached. For this, follow the instructions that accompany the filter. Then attach the crossover to the rear wall, through the cutout of the upper 17 W-75.

The corresponding damping pad covers the crossover from above. During the next step, the cables are pulled inside. Please remember to allow that the cable leads can be attached to the drive units. The cable leads must be identified; confusing the cables or failure to maintain polarity will have disasterous consequences.
Now mount the variovents. Coat them with a layer of the adhesive, insert them into the respective mounting holes and attach them with the help of a rubber mallet. Do the same with the input connection terminal. Now you may start mounting the drive units. When soldering the leads observe the polarity of leads and drive units. The baskets should be sealed with adhesive and mounted firmly to the baffle.



The TWYNN filter

The crossover filter of the TWYNN is of particular interest. While the use of a six db, first order crossover is not extraordinary, only the sophisticated design of the DYNAUDIO drive units makes it feasible to use this kind of filter circuitry. That it works exceptionally well is due in large part to the complimentary characteristics of the 17 W-75 and the D-28 AF. These units have a well matched frequency response pattern so that the filter only must make slight adjustments for the desired results. The final system presents the power amplifier with a load it can drive with ease, and it effectively suppresses cable reflections beyond the audio range. Finally, a special impedance network is inserted in parallel with the frequency network, so that the impedance response is extremely linear in both output and phase even to frequencies below 50 Hz.

Crossover modifications:

Depending on the damping characteristics of the listening room it may be advantageous to slightly vary the treble level.

	R 1	R 2
High-frequency roll-off	2.2	15
Normal	1.2	15
High-frequency emphasis	1.2	omit
Extreme high-frequency emphasis	0	omit

Other constructions published: MYRAGE, AXIS, PROFIL, XENNON, PENTAMYD, JADEE

DYNAUDIO

Loudspeakers, assembled filters, variovents, audio wool, and inputs are available at a cost of $500.00 per pair.

Cabinet materials can be obtained from local building material suppliers.

After construction, you will enjoy DYNAUDIO state-of-the-art sound reproduction.

DYNAUDIO dealers include:

A & S Speakers
3170 23rd St.
San Francisco CA 94110
415/641-4573

Audio Concepts
1613 Caledonia St.
La Crosse WI 54602
608/781-2110

Madisound Speaker Components
8608 University Green
Box 4283
Madison WI 53711
608/831-3433

Meniscus
3275 Galdiola SW
Wyoming MI 49509
616/534-9121

Solen Electronique Inc.
4470 Ave. Thibault
St.-Hubert, QC
Canada, J3Y 7T9
514/656-2759

Speaker City
10605 Vanowen St.
Burbank CA 91505
818/508-1908

Continued from page 37
ing them through 1kΩ resistors, and the ground leads should be isolated with 100Ω series resistors, as shown in *Fig. 5*.

The amplifiers for the rear channels need not be the same quality as those for the front channels, according to my extended listening evaluation. I believe you can obtain the surround-sound effect without any degradation of the system's overall sound quality with this arrangement, even if the rear-channel amplifiers are not the best. Low-power amplifiers could even be used here, since the necessary rear-channel signals require much less power than the accompanying front-channel signals.

Of course, you could simply tie a Dynaquad adapter (if you can find one) to the outputs of the front-channel amplifiers, if the rear-channel speakers are efficient enough. Then you would use the Dynaquad level control to adjust the rear-channel levels.

AMBIENCE LEVEL. The rear-channel output level is a set-and-forget type adjustment for a high-quality audio system. It provides the correct level for ambience information playback, and that depends on the recording techniques that were employed.

If the rear-channel levels you have set are just right for good stereo recordings, any increase in the rear-channel level to compensate for dry-sounding recordings will simply add noise, and then you'd be no better off than if you had obtained the rear-channel signals the conventional, noisy, expensive way.

CAPABILITIES. With this type of ambience-recovery system, you give up the ability to select different kinds of hall acoustics and reverberation times. However, you gain a pronounced improvement in playback realism, with no attendant extra noise or hash, and at very little extra expense. The use of tiny rear-channel speakers is also an attractive feature of this arrangement, if listening room decor is important to you.

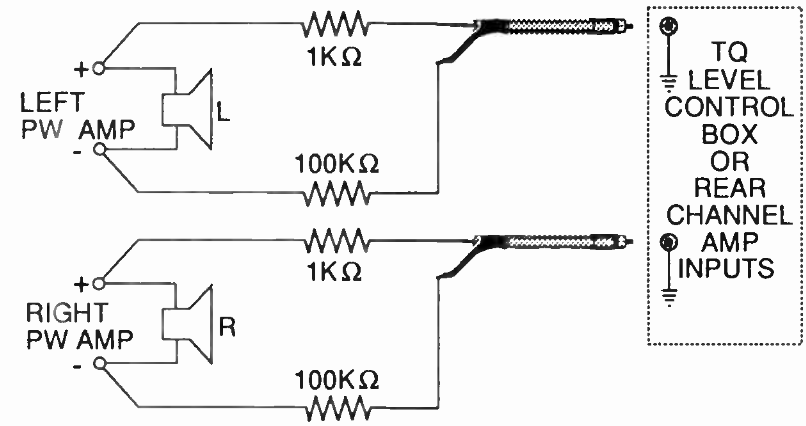


FIGURE 5: Isolating two, separate, front-channel amplifiers' signals.

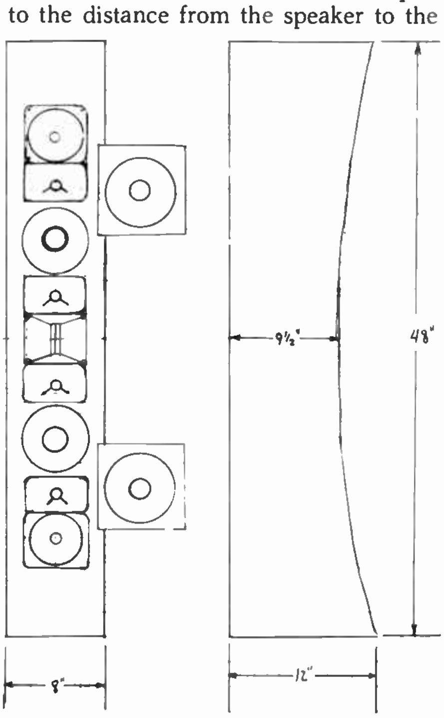
Craftsman's Corner

The Arc

My speakers have grown in a bacterial-like fashion since my first project. Innocently, I began with a three-way system—10-inch woofer, 2-inch dome midrange and a 1-inch tweeter. To that I added a 12-inch woofer, another tweeter and midrange, mid-bass drivers, subwoofers, super tweeters, active crossovers biamped, triamped and on and on.

The current incarnation is an attempt to add some finesse to my brute force approach to speaker design, using the curvilinear array from Scott Ellis's article (*SB* 2/85). The basic theory is that with a tall, flat line source the path length to the end is longer than to the center. This can result in interference, especially at higher frequencies. One way to alleviate this is to restrict the high frequency drivers to the center of the line.

Ellis's method is to bend the line source so the ends are the same distance from the listener as the center. This means the front baffle must be an arc, with a radius equal to the distance from the speaker to the listener. This is a bit like tilting the baffle to align the drivers' voice coils. I chose a 10-foot radius. A ¼-wavelength at 20kHz is 0.17 inch. The error from a 10-foot radius and 1-foot driver is 0.05 inch.



I chose to interleave the drivers—tweeter, midrange, tweeter, midrange—to avoid interference effects in the horizontal plane. By using a vertically symmetrical design, the super position of the drivers at the "focus" of the curved speaker means the signal from each arrives with the relative phase intact. I placed the drivers as close together as possible to minimize interaction in the crossover region. The mid-bass drivers are placed equidistant between each pair of midranges. This allows a nearly symmetrical arrangement and keeps the interdriver distance as small as possible.

To make the baffle, I used a giant compass—a piece of string with one end looped around the leg of my dining room table, and a pencil tied to the other end. I inscribed an arc on a piece of ¾-inch poplar veneer plywood and cut it with a jigsaw. I used ¼-inch poplar plywood for the front baffle. After I cut out the holes for the drivers I attached the baffle with drywall screws. The back is particle board, top and bottom are poplar plywood. I filled the enclosure with acoustic foam and mounted the drivers after assembling the panels.

This large, thin front panel does not seem to vibrate for several reasons. Because the length is flexed by the curvature of the side panels, it is stiffer. Also, since I use separate bass enclosures, the midrange and tweeter drivers do not excite much panel action. I mounted the drivers with thick rubber gaskets (⅛ by ¼ by ⅜-inch) and bolts, to damp vibrations and align the drivers' acoustic centers. The unsupported length is less than 6 inches.

For the mid-bass I used cylindrical sealed enclosures, from Bernhard Muller's design ("Sonotube Speaker Cabinet," *SB* 2/85, p. 43), using builders' cardboard tubing. Mine has a poplar plywood front and rear and is filled with polyester fiber.

The bass enclosure is a small conventional second-order type with two small woofers, which I braced, filled with scrap Sonex foam and used pyramid-shaped dense cardboard (packing material for toilets—we must be resourceful) to break up standing waves.

The separate subwoofer is actually two enclosures. The volume is shared by a 12-inch woofer in a fourth-order enclosure and a 10-inch woofer in a second-order enclosure. I use internal baffles and foam to break up standing waves. I made the enclosure from ¾-inch particle board, which I glued, sealed with silicone caulk and screwed along all joints.

I used active crossovers, a Heathkit at 80Hz, 18dB/octave with adjustable slope and bass level, and a Biamp SM-23 at 400Hz, 12dB/octave; and passive crossovers at 1.5kHz and 5kHz, 6dB/octave, with a capacitor to block frequencies below 14kHz.

I also built an inexpensive rack for the electronics using 2 by 4s and 1 by 2s, with slats running from front to rear. The top is poplar plywood trimmed with moulding. I added small braces along the bottom for rigidity.

You could choose better drivers with a little planning. As my system gradually evolved, I had no master plan for construction or selecting drivers. I would prefer ribbon drivers for the midrange and tweeter—perhaps Strathearns, or I would like to try to make my own. I would also use the curvilinear approach with these drivers.

Scott Henion
Bedford, MA 01730

ABOUT THE AUTHOR

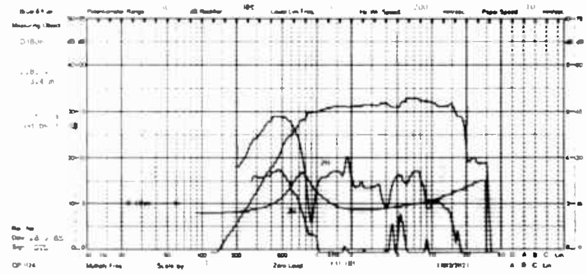
Scott Henion is employed at MIT's Lincoln Laboratory in Cambridge, MA. He recently obtained his master's degree in physics from the University of Lowell, where his thesis work primarily consisted of building lasers.

scan-speak

NEW PRODUCT INFORMATION

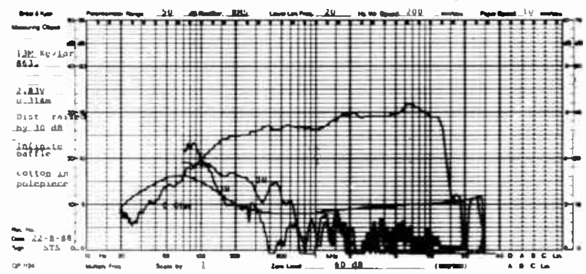
D3806 Midrange/Tweeter

Characteristic sensivity	91 db/1W/1m	Lin & max excursion	±0,4/±1 mm
Free air resonance Fo	730 Hz	Air gap flux density	1,3 T
DC resistance of V.C.	6,0 ohm	Force factor BL Product	3,5 Tm
V.C. inductance	0,13 mH	Moving mass incl. air	0,36 g
Rated power (cross over		Nett weight	1,2 kg
12 db/oct. at frequency)	100/1000 watt	V_{as}	No measurements
Effective cone area	15 cm²	Q_{ms}	
V.C. diameter	38 mm	Q_{es}	
V.C. height	3,2 mm	Q_{ts}	
Air gap height	2,5 mm	Recommended enclosure volume	
Data stated are measured after a 24 hours' test		Recommended frequency range	1.000-16.000 HZ



13M8636 Kevlar-Midrange

Characteristic sensivity	88 db/1W/1m	Lin & max excursion	±1,5/±6 mm
Free air resonance Fo	61 Hz	Air gap flux density	0,9 T
DC resistance of V.C.	5,8 ohm	Force factor BL Product	5,5 Tm
V.C. inductance	0,1 mH	Moving mass incl. air	4,4 g
Rated power (cross over		Nett weight	1,2 kg
12 db/oct. at frequency)	110/300 watt	V_{as}	6 l
Effective cone area	48 cm²	Q_{ms}	1,18
V.C. diameter	38 mm	Q_{es}	0,32
V.C. height	9 mm	Q_{ts}	0,26
Air gap height	6 mm	Recommended enclosure volume	2-8 L
Data stated are measured after a 24 hours' test		Recommended frequency range	250-9 kHz



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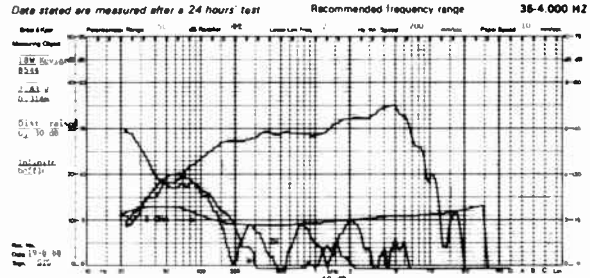
Characteristic sensivity	89 db/1W/1m	Lin & max excursion	±6,5/±10 mm
Free air resonance Fo	35 Hz	Air gap flux density	1,1 T
DC resistance of V.C.	5,5 ohm	Force factor BL Product	7,6 Tm
V.C. inductance	0,1 mH	Moving mass incl. air	17 g
Rated power (cross over		Nett weight	2,06 kg
12 db/oct. at frequency)	100 watt	V_{as}	40 l
Effective cone area	143 cm²	Q_{ms}	0,46
V.C. diameter	42,5 mm	Q_{es}	0,36
V.C. height	19 mm	Q_{ts}	0,20
Air gap height	6 mm	Recommended enclosure volume	10-30 L
Data stated are measured after a 24 hours' test		Recommended frequency range	35-4.000 HZ



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Fast Reply #FD572

HOME-BUILT SPIKES

BY PAUL DWIGHT

Having both read and heard the assertions that sonic improvements can be obtained by using spiked feet under loudspeakers, I decided to experiment with the practice myself. However, two obstacles drove me to try my hand at designing and building my own spikes.

First, commercially advertised spikes are seldom available for less than $20 per speaker. In the world of speaker building, a $40 expense is seldom regarded as prohibitive; however, I was reluctant to make the investment without being convinced of the audible benefits.

Second, most spikes available commercially are not suitable for use on existing cabinets without some modification, for instance, drilling holes in the bottom of cabinets I had worked long and hard to make airtight.

Whenever I embark on a design quest, I first attempt to understand the theoretical basis for what I'm attempting to put into practice. In this case, however, most "explanations" I found were limited to claims that the spikes "rigidly couple the speaker to the floor." To my somewhat methodical (my wife calls it "plodding") mind, this claim begs the question: Why should a rigid coupling between the floor and the speaker audibly improve the speaker's output?

First, we all know that when a speaker (without spikes) rests on a carpet, rug, or other similar surface, the addition of speaker spikes can greatly improve the speaker's mechanical stability (that is, it's not so easily moved or tipped over). When a dynamic driver responds to a signal, its forward motion is resisted by the air in front of it, resulting in a backward pressure on the cabinet. Any cabinet movement could audibly smear the soundstage. Spikes could alleviate this problem by making the cabinet stable enough to resist movement induced by the driver's motion.

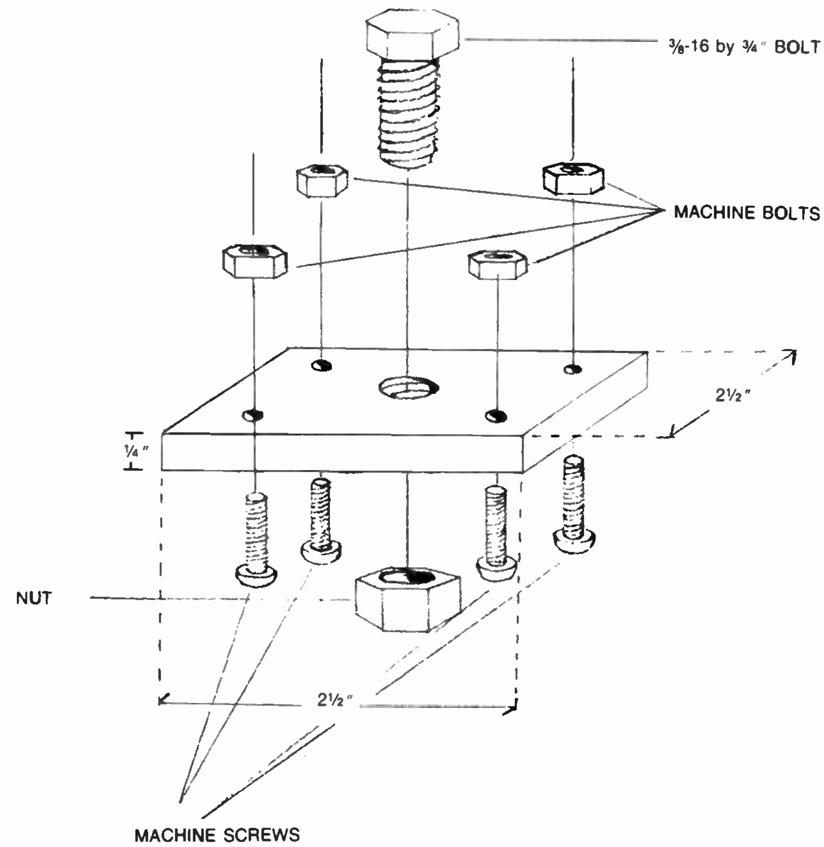

FIGURE 1: Constructing a spike.

Second, dynamic drivers often induce cabinet vibrations. The cabinets themselves become transducers at certain frequencies (in a manner essentially unrelated to the musical signal). I believe speaker spikes can alleviate this problem by providing a means for the energy stored in the cabinet to be transferred into a structure that can better store it without being forced into vibration—the floor.

To use these theoretical advantages, a spike design must make the speaker mechanically stable and provide a conduit for vibrational energy from the speaker to the floor. I reasoned that, since energy *transfer* is a significant goal, anything in the spike that absorbs energy (such as rubber or foam between the spike and the speaker) would impede the transfer of energy, and thereby impair the spikes' performance.

I believe my design (*Fig. 1*) satisfies these criteria, as well as abating my concerns regarding cost and suitability for use with existing cabinets. Mechanical stability is ensured, in spite of the carpet beneath the speakers, by supporting each cabinet on four spikes. The spikes penetrate the carpet and pad to rest directly on the subfloor; the speakers don't move unless the subfloor moves.

Sufficient energy transfer is attained by a noncompliant coupling between the spike and both the cabinet and the floor. The spikes can be used on virtually any set of speakers; all you do is lift the

Continued on page 46

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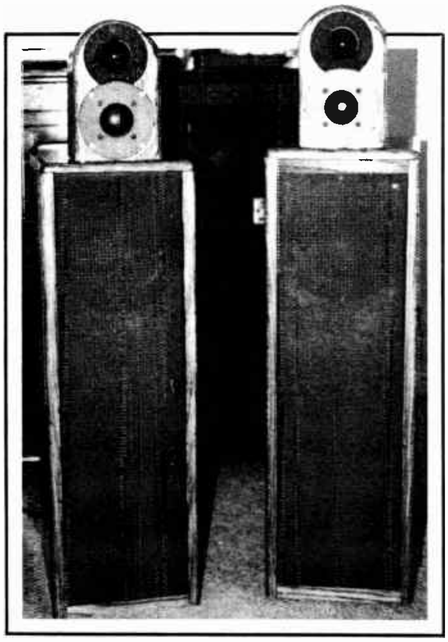

PHOTO 1: The cabinet on the right is resting on homemade spikes; the cabinet on the left is flat on the carpet. The spikes' appearance is subtle.

speaker and slip them underneath. A set of eight spikes can be built for about $20, even if you must buy all the necessary specialized tools. (If you already own suitable taps and drill bits, the cost is as low as $5.) And finally, the effect on the speakers' appearance is quite subtle, as shown in *Photo 1*.

DESIGN. The spike's "business end" is a ⅜-inch steel bolt, ¾-inch long, which I filed to form a spike. I mounted the bolt and the machine screws that couple the assembly to the speaker (facing in the opposite direction) in a ¼-inch steel plate. I drilled and threaded the plate to accept the bolt/spike and the machine screws; thus, the whole assembly screws together. The nuts, therefore, have no function in holding the assembly together but add support to the length of the bolt and screws.

You could construct a similar assembly without cutting threads (relying instead on the nuts to hold the bolt and screws rigidly in place), but I don't believe the result would be sufficiently mechanically rigid to support a moderate-size speaker. Screwing the components into threaded holes ensures close tolerances between the plate and the bolt and screws, thereby eliminating the possibility of lateral "slop" which could seriously weaken the assembly.

The only necessary part that could prove difficult to find is the plate steel. Around here, steel of this type is sold by home improvement centers in various shapes and sizes. The ¼-inch steel I used was 2½ by 36 inches, and cost about $3.50.

TAPS AND BITS. You'll need a tool called a tap to cut threads in a hole already drilled in the steel. The size of the hole is absolutely critical. For each size tap, only one size hole will give satisfactory results. The first step, therefore, is to choose a tap, which are sized in the identical manner as the bolt or screw whose threads they cut (for example, a tap for a ⅜-16 bolt is size ⅜-16). Therefore, you can simply select the bolt and machine screws and then pick up the corresponding tap. I used a ⅜-16 bolt (for the spike) and size 6-32 machine screws. After you've selected the right taps, find the appropriate drill bit sizes from the taps' packages—the manufacturer will specify what size to use. In my case, I needed 5/16 and 7/64 bits.

I'd like to offer a bit of advice regarding tools. Remember, you're going to be working in steel, which is hard on cutting tools. Ordinary high-speed steel drill bits will probably dull quickly; so unless you're able to sharpen dull bits, consider investing in bits made of a harder material. I had good luck with tungsten-coated bits marketed by Vermont American. Similarly, steel is hard on taps. In particular, it's easy to snap the smaller tap in two while forcing it to turn and cut. Look around for taps made by a reputable manufacturer that carry a guarantee of a refund. It will be a lot easier to get a replacement for the third tap you've broken in one day if you have a guarantee to back you up. Again, I had good results from Vermont American taps, which carry this sort of guarantee. You should be able to purchase all the tools—taps and bits—for about $15.

SPIKE ASSEMBLY. To begin construction, securely clamp the length of steel, backed by a piece of scrap wood, to your work surface. Mark the center of each hole for the first spike. Easily visible marks can be made with a scratching awl or colored pencil. I also use a center punch and a hammer to indent the center of each hole, so the bit does not "wander" as I start drilling.

Take your time when you drill. Line up the drill as nearly square to the steel as possible, and don't force the drill—let it work at its own speed. A little light oil in the hole keeps the bit cooler and helps expel cuttings from the hole. Also, be sure the steel is backed by wood. You'll find that the larger bit, especially, will tend to grab and bind at the very bottom of the hole as it cuts through the final bit of steel in one bite. Using wood behind the steel will lessen this problem considerably. When it still happens, as it will, reverse the direction of the drill to free the bit, bring the drill up to speed, and slowly lower it into the hole again. It sometimes takes several tries, but it will cut through eventually.

After you've drilled the five holes (four small and one large) for the first spike, reposition the steel so that it's securely clamped with a couple of inches of space between the workbench and the bottom of the plate. Now you must cut threads in the holes you've just drilled. If you've never used a tap before, I strongly recommend that you begin with the larger one. As I mentioned, the smaller tap is easily broken and, though you should be able to get a free replacement, it will take a lot longer to finish the project if you have to run to the hardware store every few minutes.

Lubricate the hole with light oil. Now place the cutting end of the tap in the hole, and grasp the other end with a wrench. Slowly turn the tap clockwise, while maintaining the tap's perpendicular attitude as steadily as possible. The first turns are the hardest because no threads are cut, and the tap will squirm around laterally as you rotate it. Eventually, you'll feel the difference as the first thread is cut and the tap becomes relatively stable in the hole. You must keep the tap perpendicular. If you make a mistake and the tap comes loose, you'll probably lose the length of thread which was just cut.

Continue turning the tap clockwise until you feel significant resistance (this will occur every 180°, or more often). When you feel the resistance, stop the tap and turn it slowly a quarter-turn in the opposite direction (counterclockwise). This process is "breaking the chip"—snapping the bit of metal just cut away from the side of the hole. If you don't do this, and instead try to force the tap to continue turning clockwise, you risk breaking the tap. Once the chip has been broken, resume turning the tap clockwise until the process must be repeated. Continue until the end of the tap extends at least ½-inch beyond the bottom of the steel plate. At this point, all the threads will be cut.

Before backing the tap out of the hole, take a second to wipe the end of the tap with a rag to remove the accumulated metal cuttings. Then remove the tap exactly as if it were a bolt or screw by turning counterclockwise. Spray the newly cut threads with WD-40 or a similar lubricant to wash out any oil remaining in the hole that could trap small metal cuttings. Repeat the process with the appropriate taps until you cut all the threads for the first spike.

Next, screw the bolt and machine screws in. It's convenient to do this before cutting the plate off the length of steel stock, since you can torque the whole thing together while it's securely clamped to your workbench. I also used a few drops of Loctite on the threads of each screw, as additional insurance that nothing will ever vibrate its way loose. Finally, use a hacksaw to cut the spike assembly free of the remaining steel.

All that's left now is to file the bolt and screws to points. I used a pair of vice grips to hold the assembly while I worked. Use a file on the machine screws. To couple firmly to the cabinet, their points need not be overly sharp.

You must also file the bolt to form a spike. I did several spikes using only a file, and this process, though time-consuming, is perfectly feasible. The work goes considerably faster, however, if you use a bench grinder to achieve roughly the shape desired, and then use the file to finish it off. *Photo 2* shows a spike assembly before and after being filed. Notice that my "spike" is actually an elongated four-sided pyramid. This shape works well and is also easy to achieve.

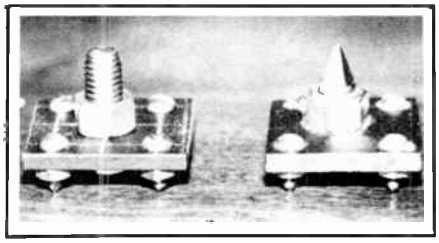

PHOTO 2: Spike assembly before and after filing. Note: shown upside down relative to cabinet position in use.

Begin by locating the center of the protruding end of the bolt, and mark it with a felt tip pen. Choose a convenient side of the bolt to begin work on, and carefully file or grind away the metal at an angle, creating a flat face, the top edge of which stops just short of the pen mark indicating the center of the bolt. Rotate the assembly 180° and repeat the process. Continue until the spike is the shape shown in the photo.

One note of caution: do not inadvertently shorten the overall length of the bolt by filing it too much. For the speaker to be mechanically stable, all the spikes must be the same length. The easiest way to ensure this is to make certain that the spike you end up with is the same length as the bolt you started out with.

CAT PROOF. Once you finish the spike assemblies, the only task remaining is getting them under the speaker. If your speakers are large, position the spikes in pairs. Tilt the speaker on its back edge and slip in the front spikes; then tilt the speaker forward to pivot on the front spikes, and position the back pair. Finally, check that each spike is securely seated against the speaker. If necessary, make adjustments until all four of each spike's machine screws make good contact with the cabinet.

After you position the speaker securely on the spikes, note the degree to which the mechanical stability of the cabinet is improved. In my case, the improvement is dramatic. My speakers are tall and slender, with a small "footprint" and most of their weight near the top. Before I added spikes, my wife's cat took great delight in jumping from the floor to the top of the speaker and seeing how far she could get it to sway before it toppled over. Now, my system is "cat proof." The speakers will still move with a determined effort, but accidental or unintended movement is eliminated.

ADDED BASS. The results? Unfortunately, I have no objective data to of-

Continued on page 71

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CD Reviews

CD Test Discs

Reviewed by Gary A. Galo
Contributing Editor

Digital Test: 106 Digital Tests and Demonstration Tracks, Pierre Verany Compact Disc PV.788031/32 (2 discs); Disques Pierre Verany, Celony, B.P. 206, 13607 Aix-En-Provence, France; $34. Available through Old Colony.

Pierre Verany's *Digital Test* is the finest collection of compact disc tests yet assembled. The two discs, the first is nearly 74 minutes and the second is 50 minutes, cost $11 less than CBS charges for CD-1, their single CD test disc (see my CBS review in *TAA* 3/88). I never considered the CBS disc a bargain, and in light of this new arrival, it appears greatly overpriced. While the CBS tests conform to the EIA standards (plus three of their own), the Verany discs are much more comprehensive.

Perhaps the least useful tracks on *Digital Test* are the first twelve musical demos. I was not particularly impressed with these recordings. All sound bright in tonal perspective and are extremely cold and analytical. I'm sure many *SB* readers own many CDs that sound far more natural than these.

Two full orchestral selections are included. The third movement from Jolivet's *Second Trumpet Concerto* is played by Bernhard Soustrot with the Orchestre Philharmonique des Pays de Loire conducted by Marc Soustrot (probably a relative). The same orchestra and conductor are featured in an excerpt from the last movement of Rimsky-Korsakov's *Scheherazade*. Both recordings feature wide separation and plenty of ambience, but are two-dimensional; the instrumental locations lack a sense of depth. The lackluster performances and Soustrot's raucous trumpet playing are completely out of character with Jolivet's music.

The first selection, although billed as a musical test, is a live recording of fireworks, complete with audience noises, and reminds me of those old Russian roulette records from the early days of stereo. Perhaps the best musical demo is *Jeux Interdits*, played by guitarist Arnaud Dumond. This recording is extremely natural, although the guitar seems a bit wider than normal.

The 94 (count 'em) bands of CD tests will be of greatest value to *SB* readers, beginning with a 1kHz, 0dB reference tone, recorded without pre-emphasis. This is most useful for making a quick check of a modified player to ensure everything is working properly. Next, the left/right separation tests use a 1kHz tone only, compared to the CBS disc tests at 125Hz, 1kHz, 4kHz, 10kHz and 16kHz. I'm not convinced that separation tests at frequencies other than 1kHz are really necessary anyway. Unlike analog playback systems, where mechanical characteristics of the cartridge can cause the separation to vary substantially with frequency, the CD does not suffer from such problems; although Dave Hadaway reminds me that while this is strictly true of CDs, in some players separation can be reduced by 40dB at 20kHz, due to crosstalk in the wiring of the analog sections. Since the digital end of the player maintains separation of over 90dB, the reduction still leaves you with over 50dB of separation, which is better than any phono cartridge and as good as most amplification devices. In my experience, CD player imaging has little to do with the static channel separation figure and much more to do with the dynamic characteristics of power supplies. I certainly wouldn't buy the CBS disc just to get the separation tests at the other frequencies.

Two swept frequency response tests are included at levels of 0dB and −20dB. For the harmonic distortion tests, CBS chose the tones relatively prime to the sampling frequency of 44.1kHz, so they are set at 61, 127, 251 and so on; up to a maximum of 19,997Hz. The Verany disc departs from this approach and sets the tones at the usual round numbers, starting at 20, 50, 100 and 200Hz, and working their way up to 15, 18 and 20kHz.

The signal-to-noise tests are at the digital zero level (infinite zero). CBS provides only one test, without pre-emphasis, but the Verany disc provides one with and one without pre-emphasis. Four de-emphasis tests are included on the Verany disc; CBS provides no such tests. For transient signal tests, Verany includes square waves at 400Hz and 1kHz, plus tone burst tests at 400Hz and 4kHz, which are a valuable inclusion. CBS provides only a 1kHz (actually 1,002Hz) tone.

The merits of 16-bit versus 14-bit quantization are controversial, and the Verany disc includes a musical test to allow you to judge for yourself. The opening of the last movement from Dvorak's *New World* Symphony is presented with 16-, 15-, 14- and 8-bit resolution. I'm sure that no one will argue in favor of an 8-bit system. As the liner notes point out, the background noise is reminiscent of a 78 rpm recording.

The Verany IM distortion tests include crosstalk between the left and right channels, with tones at 1kHz and 5kHz. Three mono IM tests are provided, with pairs of tones at 50Hz and 7kHz, 400Hz and 7kHz, and 19kHz and 20kHz. These, unlike those from CBS, do not follow the antiquated SMPTE standard of mixing the low and high frequencies in a 4:1 ratio; both tones are at the same level.

Disc 1 concludes with overload tests featuring test tones and music. Three levels of 1kHz tones are provided, at 0dB, +3dB and +6dB respectively. You won't need a distortion analyzer to confirm that something is amiss on the overloaded tracks. Finally, two tracks from Smetana's *Moldau* are recorded at maximum levels of 0dB and +9dB respectively. The overloaded track is quite unlistenable and I don't believe you'll need a double blind test to confirm this. Verany justifies the musical test, stating, "Furthermore, is not music the only and final aim of all audio technologies?" A laudable goal, I agree.

Disc 2 is designed to verify the mechanical, digital and processing performance of the player, and you can also test the compensating and servo systems. Test Series #1 varies the linear cutting velocity from 1.2 to 1.4m/sec, which are the outside tolerances of the CD standard. Many CDs are not cut at precisely the same speed, so your player must be able to handle discs within this range. Remember, unlike the LP setup, the CD's rotating speed varies across its playing surface. The disc starts at 500 rpm at its inner track and gradually slows to 200 rpm at the outside

(CDs play from inside to outside). The result is constant *surface speed* across the entire disc.

Test Series #2 contains variations in track pitch. A narrow track pitch is more difficult to read. The standard is 1.6μm, but the tolerances allow pitch to vary from 1.5–1.7μm. Two tracks are provided at each extreme, at a linear velocity of 1.4m/sec, followed by a test which varies the velocity while maintaining the maximum track pitch of 1.7μm. Finally, 1.5 and 1.7μm track pitches are provided at a linear velocity of 1.2m/sec. Test Series #3 simply varies the track pitch from 1.7μm down to 1.5μm in five successive tracks.

Test Series #4 tests the player's ability to detect the digital signal, simulating variations in cutting intensity, or asymmetry. The asymmetry is defined by the ratio of the pits to the flat zones. The greater the pits' depth, the easier the CD is to read. Very low asymmetry indicates a shallow pit depth. These tests vary the asymmetry from very low, to average, to very high; in five increments. I should mention my Magnavox CDB-650 (modified to beyond POOGE-4 standards) passes each of the Series #1–4 tests without difficulty.

The remainder of the disc is devoted to dropout tests. If you look at the playing surface of the disc, you'll actually see these tracks of varying length, where no data is recorded. Test Series #5 contains 14 tests with dropouts ranging from 0.05mm to 4mm in 0.05mm increments. A 0.2mm dropout is considered the CD standard limit. My CDB-650 played the 2.5mm dropout without problems and only the last two tracks produced audible difficulties—exceptional CD player performance.

Series #6 tests for dropouts at the minimum track pitch of 1.5μm. These vary from 1mm to 2.4mm in four successive tracks. My CDB-650 can play any of these bands if you access the tracks individually. They sometimes produce an error display if they are accessed with the NEXT, PREVIOUS or REPLAY controls. Finally, Test Series #7 checks the player's ability to correct successive dropouts. These seven tracks each contain two successive dropouts ranging from 0.1mm to 3mm. The CDB-650 made it to 1.5mm without problems, but above that it produced the error display. I've had many people try their "defective" CDs on my Magnavox players and these usually play without incident. This test disc verifies the excellent tracking capabilities of the Philips-made players.

An accompanying, well-written booklet explains all the tests in detail, although the English translation is occasionally awkward. Pierre Verany's *Digital Test* is an indispensable addition to your test bench (or listening room), and it puts the CBS test disc to shame. This two-disc set is the one to own, and the price is a bargain.

The Colossus CDs

Mozart: Piano Concerto No. 13 in C Major, K.415; Overture to *Lucia Silla*, K.135. Jeremy Menuhin, pianist; The Midsummer Mozart Festival Orchestra conducted by George Cleve. Bainbridge BCD6273.

Rachmaninoff: Sonata in G minor for Cello and Piano, Op.19. Stephen Kates, cellist; Carolyn Pope Kobler, pianist. Bainbridge BCD6272.

Louis Dorren has received a great deal of attention lately for his new digital recording system which, putting all modesty aside, he calls Colossus. Although available technical details of the Colossus recording system are few (Dorren will not release such information until his patents have been secured), we do know a few facts. Colossus is a four-channel, 16-bit recording system that is NTSC compatible and, as such, can be used with any video recorder. The system does not use any data compression or error concealment. Among its principal virtues is extreme accuracy in the analog-to-digital conversion process. Careful attention has also been given to the analog circuitry.

According to the liner notes, Colossus delivers "a three-dimensional and utterly transparent soundstage... . The stereo image is absolute and precise; instruments maintain their perspective without annoying phase shifts. The tone center of the ensemble as well as individual instruments contained in this recording is quite remarkable."

Thomas Edison stated when his Diamond Disk phonograph was introduced in October 1913, recording has reached its "final perfection." I am always a bit skeptical of products claimed to be free of all defects. What will Mr. Dorren's next improvement in digital recorders be called? Colossus II? Son of Colossus? How about Colossus Strikes Back?

Seriously, though, these liner notes (from the Mozart disc in particular, written by Pavel Moravan) do contain a great deal of hype about recording engineer Leo de Gar Kulka and the Colossus system. A rather silly "Oh wow!" style permeates the writing.

After hearing a modified PCM-1 playback, conductor George Cleve approved the Concerto and Overture for release, and we were told "Ahhh, but the best is yet to come!" Then the Colossus tape was played back and we were told that Kulka listened "with great concentration (before falling out of his chair)." I guess I'd rather have the facts, and let the listeners determine whether falling out of the chair is warranted.

Mozart's *Piano Concerto No. 13* might be called his "other" C Major Concerto, since it is not nearly as popular as K.467 or K.503. It is a fine work and is worthy of more than the half-dozen or so recordings currently listed in Schwann (an admittedly incomplete source of information). The Midsummer Mozart Festival is held in the San Francisco Bay area, and the Viennese-born conductor George Wolfgang Cleve has led the festival for the past twelve years. The Festival Orchestra is a respectable, but less than world class, ensemble. The strings, violins in particular, have a somewhat scrawny sound, and we can hear some minor discrepancies in intonation within the sections.

Cleve gives rather straightforward accounts of these two scores. Listeners who prefer a bit of tempo flexibility in their Mozart will find little of it under Cleve's direction. I do not advocate an overly romantic view of Mozart, but I would prefer a little less rigid direction.

Pianist Jeremy Menuhin, son of Yehudi, is a different matter. In fact, his approach seems somewhat at odds with the conductor's. Menuhin seems to wish more flexibility. This is particularly noticeable in passages where the soloist echos the orchestra, or vice versa. Menuhin employs tasteful rubato, whereas Cleve forges straight ahead. Menuhin's cadenzas are beautifully conceived and convincingly improvisatory, as they should be.

Lucia Silla, an opera from the pen of a fifteen-year-old Mozart, was not a success when first performed, and has remained obscure. The Overture might be called a symphony in miniature, since it is actually composed of three short, but separate, movements. As the work of a composer in his midteens it is impressive, but hardly memorable. The performance is routine, and I do not see the purpose for the work.

This recording was made with a total of six microphones. Two spaced omnidirectional Neumann TLM 170I microphones, ten feet apart, were used for main orchestra pickup. The piano was separately miked with an AKG C-24 stereo mike, and although the notes do not specify the directional pattern, it is obvious from the sound that a figure-8 type was used. Finally, two Countryman omnis were placed halfway back in the hall for ambience retrieval.

The most immediately noticeable feature of this recording is an unusually high level of low-frequency background rumble, which sounds like noise from the ventilation system; the ambience mikes are probably responsible for most of it. I have never understood the need to place separate ambience mikes back in the hall; I normally work with coincident pairs and I prefer to find the right spot for the mikes that will give a natural balance between direct sound and ambience. This often will pick up less extraneous noise from blowers, the audience, and so on. I sympathize with recording engineers who must cope with noisy ventilation systems. In both the halls in which I regularly record I must turn off the blowers before recording.

The string sound on the *Lucia Silla* recording is unusually smooth and natural, although as I mentioned before, the sound of the strings in this orchestra is not exactly ideal. Having heard a number of second-rate string orchestras, I am inclined to believe the Colossus recorder is quite accurate. The soundstage is wide, with excellent left-right separation. There appears to be a hole in the middle of the orchestra; the center fill, particularly left of center, is rather vague.

The piano does fill up the center, but it sounds as if it were recorded in a different hall altogether. The orchestral sound is rather close in perspective. The piano, on the other hand, is nearly swimming in reverberation and is placed behind the orchestra. Even the wind instruments sound as if they are in front of the piano. The recording philosophies at work here are contradictory. The use of a Blumlein pair with spaced omnis is an odd combination. I'm not saying that it can't work, but it most certainly doesn't in this case.

The lower register of the piano images right of center while the upper register appears in the center. In fact, the overall balance leans toward the right. The orchestral image and depth from center to right is convincing, but all the violins appear to be in the extreme left, with little fill toward the center. The overture exhibits a better image, particularly center to left. The seating, without a piano onstage, probably helps.

Rachmaninoff composed his *G Minor Cello Sonata*, Op. 19 in 1901, and it is worth remembering that the *Second Piano Concerto* is Op. 18.

Stylistically the two have much in common and I fail to understand why the *Cello Sonata* has received so little recognition. It is a fine work, worthy of a greater place in the chamber music repertoire than it now occupies.

Rachmaninoff was a superb musical architect, but his longer works can appear to wander when placed in the hands of a performer unable to grasp their sometimes elusive structures. It should be no surprise that Rachmaninoff, one of a handful of pianists who can be given "greatest of the century" status, did not compose a sonata for cello with piano accompaniment. This piano writing is as demanding as any he wrote, and it would be an insult to label the pianist a mere accompanist.

The instruments used on this recording receive nearly as high billing as the soloists themselves. Stephen Kates plays a Montagnana Cello built in 1739 and we are assured that "When caressed by Stephen Kates, the instrument sings, it commands, it speaks in stentorian tones."

Yes, the notes for this recording are filled with the same rather sickening combination of hype and snobbery that permeates those for the Mozart disc. As for the piano, Kulka reflects, "From my early music days, I dreamt to once again hear the sound that meant *Grand Piano* to me. And now again, I own a Bosendorfer Konzertfluegel."

The Bosendorfer pianos have enjoyed a kind of cult status over the years. They are indeed superb instruments. I had a chance to try one at an AES convention a few years ago; Kimball rented a demonstration room to show off its flagship German import. To me, the most notable characteristic of the Bosendorfer is the clarity and definition of pitch in the extreme low register.

The Bosendorfer contains four additional notes below the low A found on 88-key grand pianos. These notes were added early in this century at the request of Ferrucio Busoni, who made use of them for the pedal notes in his transcriptions of the Bach organ works. The four added notes move the normal bass strings closer to the center of the sound board, which improves the low-end definition

One piano technician I know, who is a hard-core Steinway advocate, acknowledges the virtues of the Bosendorfer in the low register, but considers the instrument inferior to a well-maintained Steinway. In his opinion, the Bosendorfer lacks power and projection compared to a Steinway, especially when it is used as a concerto instrument with a large symphony orchestra. I hear first-class Steinways every week, and to me, that sound means *Concert Grand Piano*. The Bosendorfer is a fine instrument, though, and can be most effective in chamber music, which is the medium on this recording.

Cellist Stephen Kates delivers a sensitive and impassioned performance. His pianist, Carolyn Pope Kobler, shows no technical shortcomings in this treacherous music, but the overall architecture of the music seems to elude her. I found the performance wandering from time to time, without a clear sense of direction and shape of the work. Kates appears to have a firm grip on the musical structure, but the performance remains hampered by the pianist.

Rachmaninoff was a master explorer of the piano's coloristic possibilities; his recordings of his own music are superb examples. (Rachmaninoff remains one of my favorite pianists, and I own his complete recorded output, much of it on original 78s.) Kobler's playing is monochromatic, although the dry recording (more on that in a moment) does not help her at all.

Recorded performances of this work are too rare. My favorite remains the old Nonesuch (H-7115, out of print) with Harvey Shapiro and Earl Wild. Their powerful and coherent reading should be re-issued on CD as soon as possible. They also received a rich, atmospheric recording from Marc Aubort. A new Decca/London CD with Lynn Harrell and Vladimir Ashkenazy is available, but I have not heard it as yet. Given Ashkenazy's affinity for Rachmaninoff, this one looks promising. Why Mstislav Rostropovitch and Sviatoslav Richter were never asked to record this work is difficult to understand.

This recording was not mastered on the Colossus system. It was originally recorded on a modified Sony PCM-1. The original PCM-1 recording was then transferred to the Colossus format for the remainder of the production stages. Finally, the Colossus copy was transferred to the CD format using the Harmonia Mundi Acoustica BW 102 digital standards converter.

I wonder why it was converted to Colossus in the first place. Since this recording was mixed "live" on two channels and no signal processing of any kind was used, how much "production" was really necessary? As good as Colossus might be, does it make any sense to add an extra step in the process just to put the Colossus name on the jacket? I believe original PCM-1 recordings can be transferred directly to CD, although I'm not absolutely sure of this, in which case *two* unnecessary steps were added.

Five microphones were used to record these two instruments: "My treasured Neumann M-49 for the cello; two AKG C-414s for the piano; and two Pressure Recording Process microphones from Ed Long and Associates for room ambience completed the recording setup."

The recording is extremely close in perspective, and extremely dry; as if recorded in a small room rather than a recital hall. I suspect the recording gear was moved to Kulka's living room (that is, his Bosendorfer). Virtually no ambience surrounds the piano; the low to middle register is bone dry. The low register of a Bosendorfer is perhaps its most distinguishing feature; on this recording it sounds like a thud recorded in a closet. The piano never has a chance to bloom.

The image on the piano is strange and seems to widen as the playing moves to the extremes of the keyboard. The piano image moves erratically. Kates' cello is also dryly recorded, although its sound is not as badly served by the recording as the piano. Kulka could have achieved a far more impressive piano recording by moving to a recital hall with good acoustics and using a less esoteric piano.

This premium priced CD contains less than 42 minutes of music. You'll find less than 36 minutes of music on the Mozart full-priced disc.

The Colossus digital recording system has earned the respect of several fine recording engineers and probably has far greater potential than these recordings reveal. The Rachmaninoff piece receives a far better performance than the Mozart works, but it is still less than ideal. Given the caliber of engineering, I can't recommend these two discs.

Telarc has issued a Colossus-processed Prokofiev recording, *Alexander Nevsky* conducted by Andre Previn, and I'm sure there will be many more, which will allow a more thorough assessment of this new recording system.

Tools, Tips & Techniques

ONE-LEG LUGS

One of the more creative applications of a spade lug at the end of a speaker cable may be substituting the lug, rather than a pin connector to fit in tight places. This can be accomplished by cutting off one of the tines, as shown in the sketch. This modified spade lug has, I believe, two distinct advantages over the pin connector:

- It is sturdier; you can really cinch down on a terminal without concern for breaking (pins often break at the notch)
- The open-ended sleeve on the end of the spade lug is easier to solder well to the cable than is the closed-end socket on the typical pin connector.

High-quality gold-plated copper spade lugs that accept up to 10-gauge cable are available for $1.50 per pair from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458 .

James T. Frane
Orinda, CA 94563

EASY PICKUP

I was recently searching for a contact pickup ("contact microphone," or "acoustic transducer") to obtain an acoustic sound from the body of my electric bass. A friend found an article describing an easy-to-construct pickup for only $2. This device could prove to be an inexpensive means for determining resonances and isolation effectiveness in turntables, tonearms, speaker cabinets, and so on.

The article, by James Chandler, Jr., (*Electronic Musician*, March 1987, p. 68) details placement of the transducer for amplifying and recording acoustic guitar, acoustic bass, and piano, and suggests several other applications.

Construction is extremely simple. Buy a Radio Shack piezoelectric buzzer #273-064. Cut away the plastic housing, leaving the metal disk. Discard the blue lead. The remaining wires may be used as the pickup outputs, or if you wish, carefully solder shielded cable in their place, with the shield replacing the black wire.

For an example, you could temporarily mount this pickup (with rubber cement, double-sided tape, or sticky putty) on a wall of your speaker enclosure. Attach the leads to an oscilloscope and play a sine wave sweep through the speaker. Try to adjust the bracing and damping materials in your enclosure to minimize resonances detected by the pickup.

If you don't have the necessary equipment, simply tape record the pickup's output while playing music through the speakers, and listen for resonances while replaying the tape. The pickup has a maximum output of about 100mV, so you may need to use a microphone input on your preamp.

Ralph Gonzalez
Wilmington, DE 19810

Technology Watch
Ribbon Loudspeakers

By Peter Muxlow

With the current interest in home built ribbon loudspeakers[1-3] I thought it would be interesting to look at the Apogee ribbon[4] which has recently received much favorable attention. The first unusual feature is the use of a trapezoidal shaped ribbon to reproduce the low frequency range (*Fig. 1*). The advantage of the trapezoidal ribbon's shape, compared to a rectangular one, is that the resonances of the diaphragm are spread over a wider frequency range.

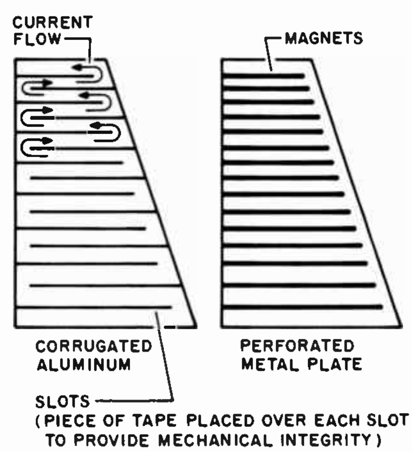


FIGURE 1: Apogee woofer.

This trapezoidal ribbon is made of 12μm, horizontally corrugated aluminum. The corrugations increase the flexibility. The trapezoid is supported on all sides and the tension is adjusted by a spring-loaded side piece that alters the diaphragm's compliance. The tension is adjusted to give a different compliance at the top than at the bottom of the trapezoid, which extends and smooths the bass response.

The aluminum diaphragm has slots cut into it to provide a conductor pattern. A large area is needed to give adequate sound level at low frequencies, since the diaphragm maximum excursion is only about $\pm$6mm. The crossover frequency is 800Hz for the Caliper model.

The magnetic field comes from the speaker's rear. Magnets are glued horizontally to a perforated metal plate and are aligned with the slots cut into the aluminum foil. This trapezoid is not a true ribbon and the magnetic field is not uniform. On a sine wave there is a greater displacement as the diaphragm approaches the magnets and less as it recedes from them.

All reviews I have read are favorable to this loudspeaker. The mid-treble array is an open-backed vertical line source. Depending on the model, the diaphragm is either a true ribbon or a ribbon conductor bonded onto a plastic film. The ribbon conductor returns are cemented to the face of the magnets. Quoting from the patent: "The current flow in the conductors up the side of the magnets provide an electromagnetic force to aid in the physical centering of the ribbon in a direction parallel to its plane and prevent the ribbon from touching the magnet faces."

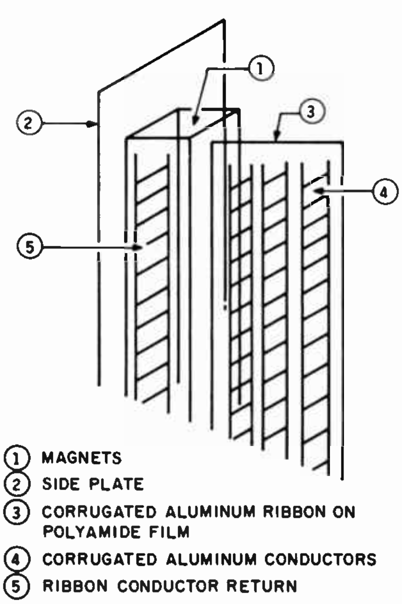


FIGURE 2: Mid/treble structure.

As shown in *Fig. 2*, the diaphragm is three aluminum conductors horizontally corrugated and bonded to a polyimide film located in the magnetic gap. The total length of the ribbon (for the Caliper model) gives a DC resistance of 3.9Ω, which is a reasonable match to the amplifier without requiring a transformer.

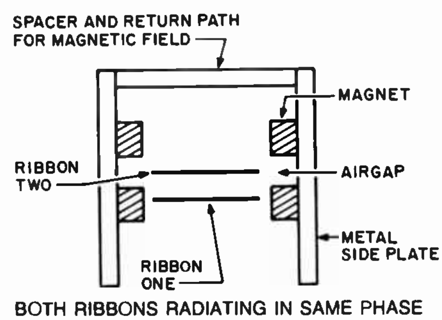


FIGURE 3: Alternative mid/treble structure. The airgap helps shape the magnetic field which helps keep the ribbons physically centered.

The patent also shows an alternative structure for a mid- and high- frequency unit (*Fig. 3*). The expensive magnet structure is shared by two ribbons parallel to each other and placed a small distance apart. It is connected so that the electrical current flows through the two ribbons in different directions, making the ribbons radiate an acoustical signal that has a unipolar radiation pattern. See references 5 and 6 for more good reading on ribbon speaker design.

REFERENCES

1. Lampton, Michael and J. Henry Primbsch, "Simple Ribbon Tweeters," *SB* 3/84, p. 7.
2. Thofte, Ole, "La Folia: A Kitchen Bench Ribbon Speaker," *SB* 1/88, p. 9.
3. Painter, Richard, "A Low-Cost Two-Way Ribbon System," *SB* 4/88, p. 38.
4. American patent #4550228.
5. Taylor, "Ribbon Loudspeaker," *Wireless World*, Jan. 1951.
6. Boleslav, Ales, "Loudspeakers for Model Investigations," *JAES*, Aug. 1970.

SB Mailbox

AMBER CORRECTION

Please note that *Fig. 6* of my Amber article (*SB* 6/88, p. 22) is not accurate. The conductive paint layer and the graphite-coated side of the diaphragm should be reversed, as shown, for easiest construction of the diaphragm and frame assembly.

David Lang
Wauwatosa, WI 53213

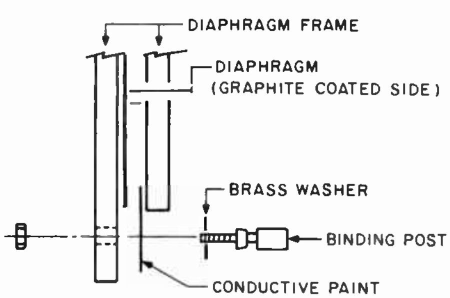


FIGURE 1: Corrected Amber construction details.

THE NOVAK FACTOR

Thanks to Jurgen Heinzerling ("Rhymes and Reasons," Mailbox, *SB* 3/86, p. 38) for giving some credit to James F. Novak. But Heinzerling's formula for Novak's optimum volume includes the 1.15 factor and that doesn't fit the Novak formulas that I remember. Novak suggested a compliance ratio of 1.44, obtained by putting the driver into a box that, before porting, would raise the resonance frequency 1.56 times that of the free-air condition.[1]

By approximation (isn't everything in acoustics?):

$$V_B = V_{as}/(f_c/f_s)^2 - 1$$

Or, by Novak's optimum volume (V_o):

$$V_o = V_{as}/(1.562)^2 - 1$$

$$V_o = V_{as}/2.44 - 1 \text{ or } V_{as}/1.44$$

For Q, Novak specified an optimum range from 0.3 to 0.4.[2] By 1973 he had modified his compliance ratio slightly, mentioning that Thiele's work contained an "exact solution for the optimum compliance ratio of 1.414.[3] This change just barely reduces the required resonance frequency ratio. Its main advantage is the esthetic satisfaction one may get by recognizing it as the ubiquitous square root of two.

Heinzerling suggests advantages in using Novak's ratio for low-Q drivers. About the time of Novak's original paper, many amplifiers offered "variable damping" that was designed to compensate for overdamped systems. Lacking that, one could add resistance to the speaker's electrical circuit. For underdamped speakers a pad of fiberglass stapled over the frame of the driver often made the speaker usable, but with reduced efficiency.

Novak stated that he developed his optimum volume concept strictly for best transient response.[4] He compared a reflex speaker system to an overcoupled IF transformer with its two tightly coupled tuned circuits. His criterion for optimum transient response was a system that would decay with one frequency, with a specific time constant, instead of with two frequencies.

Before Novak's optimum volume concept, most advice on enclosure volume was based on cone diameter alone, with no regard for compliance. The only fault we might find with his one-ratio concept is that it violates an old Ozark maxim: "You can't get all the possums up one tree."

Heinzerling mentioned the practical side of Novak's work and praised him as a designer who worked with real loudspeakers. I agree, but I also remember hearing people criticize him in the 1960s as "too theoretical."

David Weems
Newtonia, MO

Jurgen Heinzerling replies:

Seemingly I am not alone with my deep appreciation for Novak's work. As a matter of fact there was quite a rave about his approach shortly before Richard Small came up with his papers on woofer loading and one might wonder if Small was himself inspired by Novak's original work and, ahem, forgot to mention.... But before I go into detail on this subject I must state that most of my sources are single sheet copies of the original texts which I got hold of via dark ways and consequently, cannot be referenced correctly. If need be, I will supply copies to show the truth about an interesting conversation between a few authorities which must have taken place a few months before Small published his paper—deep waters, indeed. Here it goes.

The conversation took place in the letters pages of *JAES*—no year given (but p. 550). Mr. Albert W. Cobb cites a formula by Beranek:[1]

$$V_o = \frac{pc^2}{1.44}\left[1.15\left(\frac{f_1}{f_0}\right)^2 - 1\right]\frac{V_T}{pc^2}$$

$$= \left[0.798\left(\frac{f_1}{f_0}\right)^2 - 0.695\right] V_T$$

As you can see, the magic factor of 1.15 pops up. Another letter, same reference: Robert Ashley answers Cobb's letter and wonders about the 1.15 factor, hoping that Beranek himself might resolve this, since he brought it up. And now it comes: Ashley mentions a letter from a friend of his, Richard Small of the University of Sydney, Australia, who has equally uncertain feelings about the 1.15 factor. So, Small was not only aware of this, but working with Novak's approach, apparently.

Ashley states that he and Mark D. Swan ran their computer programs with and without this factor and found much better results using it. So you are right in stating the 1.15 factor is not included in Novak's original work. But it must be recognized as a valuable extension.

Anyway, at the end of his letter Ashley asks where Novak found his magic number of 1.44... so we are back into "Wisdom and Witchcraft about Old Wives Tales about Woofer Baffles."[2]

As I mentioned in my article, Novak's approach leads to relatively large enclosures which I consider advantageous with vented designs. More recent investigations into the subject have convinced me that, contrary to what our beloved analogies tell, woofer-Q at resonance is not a static parameter. It varies with signal level and thus, T/S alignment is valid only for one signal strength. But music, as

REFERENCES

1. Novak, James F., "Designing A Ducted-Port Bass-Reflex Enclosure," *Electronics World*, Jan. 1966.
2. Novak, James F., "Performance of Enclosures for Low-Resonance High-Compliance Loudspeakers," *IRE Transactions on Audio*, Vol. AU-7, No. 1, Jan.-Feb. 1959.
3. Novak, James F., Letter, May 8, 1973.
4. Novak, James F., Letter, Aug. 28, 1968.

it should be reproduced, consists of impulses of different levels and accordingly, each and every box involving any type of loading on a driver working after the principle of electrodynamic motion, will work in a momentary (dynamic) alignment. Further, one must account for amplifiers twice removed from the ideal and this results in a system totally uncontrollable and unpredictable.

So, once and all, I must confess that I cannot believe in exact alignment of speakers on a filter basis. This might work for radio-circuits, where we deal with relatively static signals.

But a way out of this involves relatively large speaker housings. Anybody using a computer program like White/Bullock's BOXRESPONSE can run large box (low alpha) alignments with F_b/F_s relationships around 1, and see that speaker response (theoretically) does not vary too much on different Qs with a large V_b. If you do the same with small vented systems, you will invariably find extreme dips and humps, leading to intolerable response.

Novak proposed a theoretical system that decays with one specific time constant only. Ted Jordan of Jordan/Watts fame took it one point further: he proposed a large box system where all resonances are of the same Q. Thus, since decay invariably takes place on the highest Q resonance, any impulse with a strong urge for decaying will find no way to do so and consequently, movement will stop. I do not believe such a system could work in reality, but it sounds (pun intended) good. And it shows that Novak was right again in approaching woofer behavior from the standpoint of impulse response and not frequency response.

After nearly two decades of Small alignment, Novak's original findings can only be interpreted as the old-fashioned, but effective way to plan vented boxes, for the tinkering speaker builder with limited methods and means.

David Weems replies to Mr. Heinzerling:

The use of the 1.15 factor appears to give V_{as} figures that are more consistent with those supplied by the manufacturers, perhaps because the manufacturers use the same factor. But yes, it is a valuable addition.

Anyone who has compared overvolume to undervolume in reflex systems will appreciate your attraction to large enclosures. Your skepticism on the alignment of speaker systems on the basis of filter theory may be a healthy attitude compared to our almost mystical belief in the theory.

In regard to Jordan, I don't quite follow your statement that all of his resonances were of the same Q. Jordan's theory of reflex design seems to be the most mystical of all, based on the 0.618:1:1.618 series, which will be recognized by classicists as the golden ratio of the ancients. It is a valid ratio for many purposes, such as enclosure dimensions that make a good visual, as well as auditory, impression. But Jordan used the ratio for just about every aspect of reflex design, such as: speaker compliance/box compliance ratio; upper resonance Q/speaker Q ratio, and so on. His ideal speaker Q of 0.618, the reciprocal of 1.618, seems high by either T/S theory or Novak's specification. However, I once built a system using Jordan's formulas and it sounded much better than I had expected. Perhaps there is yet something to be learned about the behavior of speakers in boxes.

REFERENCES

1. Beranek, L., *Acoustics*, McGraw-Hill, 1954.

2. The original title of Ashley's *JAES* article, Vol. 18, p. 524.

CERAMIC MODELS

David Weems' ceramic enclosure speaker (*SB* 6/88) looks very attractive. As usual, he has produced a well-thought-out design, and the SEAS and Audax drivers appear excellent.

Out of curiosity, I modeled the crossover and drivers with my LMP (*SB* 1–3/87) to see whether his choice of "book" crossover was appropriate. Of course, the conclusions obtained from LMP are only as good as the model used. I've listed the model in the Table, but I've not verified all the choices empirically.

Figure 1 shows the LMPG (graphics version) prediction for the case where the listener is on-axis with the slanted front baffle: for example, if the enclosure is floor-mounted at an 8-foot listening distance and the listener's ears are located about 38 inches above the floor (assuming the baffle angle is 20 inches). In this case a peak of about 4dB appears above

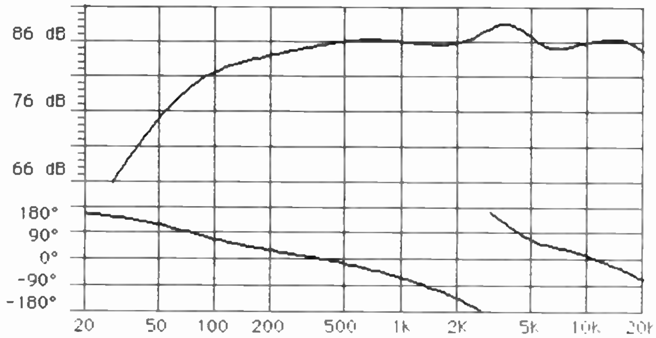


FIGURE 1: Floor-mounted speaker.

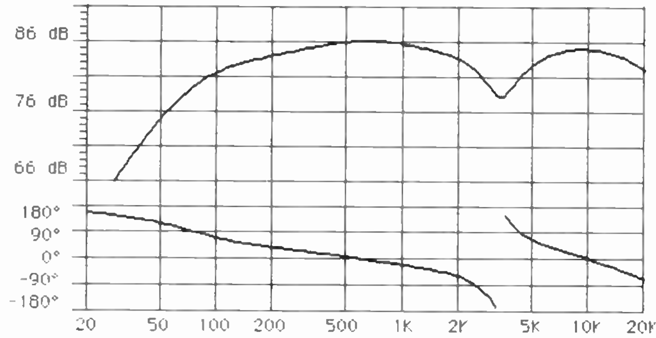


FIGURE 2: Speaker at ear level.

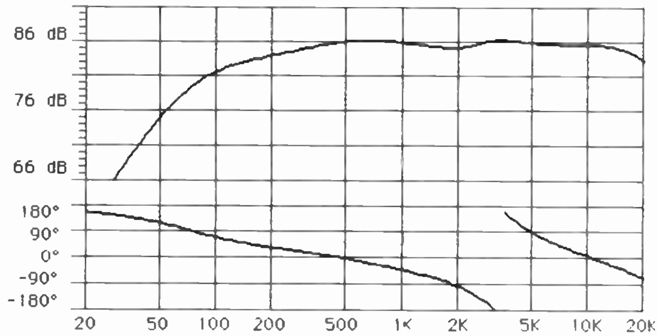


FIGURE 3: Speaker on a 15-inch stand.

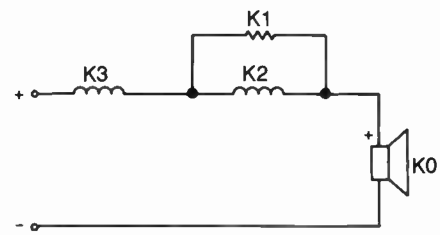
```
6410 G(X) = 1
6420 P(X) = 0
6430 N1(X) = 1
6440 A(X,0) = 1
6450 A(X,1) = K(X,2)/K(X,1)
6460 N2(X) = 2
6470 B(X,0) = 1
6480  B(X,1) =(K(X,1)*K(X,3)+K(X,0)*K(X,2)+K(X,1)*K(X,2))/K(X,1)/K(X,0)
6490 B(X,2) = K(X,2)*K(X,3)/K(X,1)/K(X,0)
6499 RETURN
```

FIGURE 4: An add-on LMP subroutine to model this crossover.

TABLE I—LMP MODEL*

PARAMETER	SEAS P17RC	AUDAX HO12X9D25
Low-frequency corner	74Hz	800Hz
Low-frequency damping ratio	0.76 (Qts = .66)	0.556 (Qts = .9)
High-frequency corner	5kHz	20kHz
High-frequency damping ratio	0.7	0.65
High-frequency rolloff order	4	2
Polarity inversion	N	N
Sensitivity (−6dB response step)	83dB	80dB (via L-pad)
Depth displacement	+0.2 inch (approx.)	0
Frequency of response step	1,500Hz (9-inch wide enclosure)	1,500Hz
Height of response step	6dB	6dB
Crossover #	14**	9
K0	6Ω (after impedance compensation)	8Ω
K1	4Ω	8.4μF
K2	0.32mH	0.3μF
K3	0.4mH	3.6μF

*For listener 10-inches below axis.
**See Figure



the crossover frequency. If the speaker is instead mounted on a stand at ear level, the relative distance between the listener and the two drivers is changed, and an 8dB dip appears above the crossover frequency (*Fig. 2*).

Finally, if the enclosure is mounted on a short stand, about 15 inches above the floor, the listener is about 10 inches below the the main axis (of the slanted front baffle) and a smooth response results (*Fig. 3*). By coincidence, in this case the listener is nearly equidistant from the tweeter and woofer diaphragms.

The last case suggests the design choices were very good indeed. In particular, the paralleled resistor/inductor combination in the woofer circuit effectively reduces the midrange prominence produced by the "response step" (mentioned in my article). LMP shows the correct setting for the tweeter L-pad should reduce sensitivity by about 2dB, for example, a 2Ω resistor in series with the tweeter, and a 40Ω resistor in parallel with this combination to maintain a net 8Ω load for the crossover.

Incidentally, I believe Mr. Weems is incorrect in his assertion that adding resistance "before the crossover elements" will reduce sensitivity without upsetting the crossover frequency. For example, with a first-order network, the crossover frequency is related to the total resistance in the circuit, regardless of whether the added resistor appears before or after the capacitor.

In conclusion, I think a simulation program such as LMP can offer insights even in cases where the correct design decisions have already been made. It appears that, like most systems, these ceramic speakers are sensitive to the vertical listening position, and in this case a 15-inch stand is optimum. I would like to know if Mr. Weems gets a chance to test this, since we all know there is no substitute for empirical measurements and listening tests.

Ralph Gonzalez
Wilmington, DE 19810

David Weems replies:

First, thanks for correcting my statement on resistor placement.

When I started reading about your LMP model, I was prepared to swear that the ceramic speakers were designed by the "other" David Weems, who tinkers with speaker building, not by me. As I read on it occurred to me that it is only fair to acknowledge your comments.

Someone once told me too many variables exist in speaker design to predict performance from theory. That statement was made long ago—longer in speaker development progression than in time.

I did not use your LMP, so I was interested in your analysis. After noticing your specified L-pad setting, I measured the actual values of R_s (series) and R_p (parallel) I had been using. They were approximately 2.5Ω and 25Ω, respectively, for a tweeter level within 1dB of your prediction. I made a quick check of Q and f_c on one speaker. The Q_{tc} was 0.67 and f_c was 70Hz. From that I estimate f_3 to be 74Hz.

I misled you on one point. The photo caption calls the tile 9 by 13 inches (the dealer's specs), but mine were just over 8 by 12½ inches, which puts the step response at about 1,675Hz. I tried to measure the output of my crossover network, but I'm not sure how valid the results are, measured with AC operated equipment. It appeared to be down about 4dB at the step-response frequency.

To test your LMP predictions on vertical placement, I made preliminary tests at three levels: on the floor, on a 15-inch base, and at ear level. My conclusion, based solely on listening tests, is that the speakers sound "right" placed on the 15-inch base. The floor position is the worst, probably because of altered bass loading as well as the rougher, higher frequency response that you show. In the floor position, the sound was less open, more like that of a "speaker in a box."

Frankly, I am amazed at the apparent accuracy of your LMP predictions.

LC1 BROTHERS

Ralph Gonzalez suggested I order the 1987–88 issues of *SB*. They really aren't what I expected, they are much better. In *SB* 6/88 (*Mailbox*, p. 74), Mr. Bennett comments about the LC1-C speakers, described by Harry Olson in *Acoustical Engineering*. I happen to be quite familiar with the closely related LC1-B design since I have heard a pair which belong to an older acquaintance who used to work in Dr. Olson's group many years ago.

They are able to produce an extraordinary live image that I've not encountered in many modern designs. According to my friend, Mr. Neves, Olson was trying to show that a speaker should be coaxial and coplanar to produce a good image. I think his LC1-Bs make the point.

The crossover of the speaker is a simple capacitor in series with the tweeter, coupled to adequate acoustical parameters. That is probably the reason the tweeter may be so easily overloaded.

As we see, "linear phase" speakers are at least 30 years old. Nowadays, KEF in England has resurrected the concept with its "Uni-Q" design. Perhaps some of the lucky readers who have the opportunity can give them a try.

Jorge O. Oliveira
13083 Campinas-Sp, Brazil

JORDAN ADJUSTS

EJ Jordan USA and its affiliate, Option Audio, announce major changes in their operations. EJ Jordan is no longer distributing Jordan drivers, related products and kits. All future warranty and sales for Jordan drivers is being handled by Ted Jordan, The Dak, Manorbier, South Pembrokeshire, SA70 8QR England, (011) 44-834-871-209.

Both our cabinet making and speaker design divisions offer *Speaker Builder* readers apologies for any unfulfilled responses for product information. Any further requests must be accompanied by a self-addressed stamped envelope; phone call requests will be returned only if we can provide service to the request or to Jordan Club members. The new address: PO Box 1411, Wall, NJ, 07719, (201) 280-9427.

Kirk Neal
EJ Jordan USA/Option Audio

R2 ATTENUATION

In reading *Part I* of the Swan IV article (*SB* 4/88), I noticed in the section on Crossover Design the authors discuss R3 in relation to attenuation of the D28—this is in the first paragraph on page 15. Shouldn't this be R2, not R3 as stated?

Robert Chambers
Corinth, VT 05039

Contributing Editor D'Appolito replies:

R2 is correct. Another correction: the satellite Q is 0.35 and not 0.5.

TWO-WAY TINKLE

I stole the 4μF/4Ω parallel circuit from the D-28 (Swan IV) and the improvement is enormous in my two-way system with a Vifa 8-inch 121WO-12 woofer and a 2kHz Linkwitz-Riley crossover. I noticed that Joe D'Appolito changed the value to 4.7μF in the new schematic, but *Photo 1* (*SB* 4/88, p. 10) shows a 3.3μF capacitor. I was looking for a little extra tinkle, so the revised value was lovely, but which one is correct and what is the difference?

Also, I would like to try the Morel MDT-27 Dynaudio copy for the short circuit/lift circuit. The values and quality seem fairly close, perhaps 0.5Ω lower at 2kHz, but without quite the high-end droop of the D-28. Could Mr. D'Appolito provide me with the equations to attenuate the horn region similarly (4dB), while lifting the plus 15kHz response 2 or 3dB, so I can try it next time. I thank the authors for the article; my project is one hundred percent better with the improvement using the circuit.

M. Thompson
St. Catharine's, Ontario L2T-1R4
Canada

Contributing Editor D'Appolito replies:

The current value of C4 for the Swan IV is 4.7μF. The *Photo* shows an earlier version of the Treble Coupler. The difference between the two is subtle. With 4.7μF, tweeter lift begins around 4.5kHz, whereas with 3.3μF the onset of lift is pushed to 6.5kHz. I picked the final values through listening evaluations.

I have not experimented with the MDT-27. You should measure the average value of its impedance between 2–4kHz. Call this value R_t. Assuming R_t is largely resistive, the value of the series resistance R_s needed for 4dB is:

$$R_s = 0.585 \times R_t$$

For an attenuation of NdB the general equation is:

$$R_s = (10^{N/20} - 1)R_t$$

SWAN SWAP

Apparently, a large number of readers have built or are building the *Speaker Builder* version of the Swan IV. I invite them to call or write to provide their name, address, phone number and any comments. A list of Swan owners could form the basis for an organization to exchange ideas with each other and with us, and could route potential buyers to nearby owners for a listen and chat. Once started, I would try to pass the torch to a more organization-oriented person to build the idea into a group. Builders, please call me at (207) 526-4343 or write to PO Box 356, Swan's Island, ME 04685.

James W. Bock
Swan's Speaker Systems

SCOLDING SWAN

I read the Swan IV article (*SB* 4/88) with interest. The design has many features that I like and many design goals with which I am in sympathy. However, some features deserve comment.

The 6dB/octave dividing network is well-known to be too slow in the rolloff to avoid substantial dips caused by the nonflat behavior of the upper and lower frequency section speakers. Most practical realizations of the 6dB slope embody amplitude compensation parts—for instance, the Vandersteen speakers.

In the case of your electronic dividing network, it would be easy to compensate for the rolloff of the midrange/tweeter module, which has its fundamental resonance at 75Hz, only 1.41 octaves below the crossover frequency. You could derive the high-pass section at 200Hz by filtering and the low-pass by subtraction.

Next, the output of the high-pass filter should be directed through a circuit similar to the IC2B (*Fig. 13* of the article), but with its transfer function being high-pass with an f_o of 75Hz and a Q of 0.5, to model the midrange fundamental resonance. The output of this second high-pass should be what is subtracted from the input to create the woofer's low-pass signal. This would more nearly approach the desired perfect transient response of the complete system. Two 12-inch woofers would have no trouble with the extra 2 or 3dB of boost at 75Hz.

The stacked woofer concept has had long application in studio monitor design because the directivity factor (Q) of the pair approximates that of a horn midtreble unit at the crossover frequency. The directivity factor of the stacked pair increases with frequency. At the authors' 200Hz crossover frequency, I would expect a step-down in Q as the upper module takes over. For this reason, a most interesting graph would be the measured total acoustical power output (4π) versus frequency, in addition to the anechoic frequency response on-axis, from 22Hz up.

D'Appolito's midrange-tweeter-midrange modules permit the use of any desired dividing network without danger of asymmetrical lobing at the 2kHz crossover frequency. Therefore, I was greatly disappointed they did not use a Butterworth response at this crossover point to preserve the constancy of energy into the room. The 12dB/octave Butterworth response yields an all-pass sum when used with an idealized horn. The 24dB/octave Butterworth response also works at the crossover point, as we can verify with a phasor diagram, but I have not worked out the algebra to prove it for all frequencies. I strongly suspect it would yield an all-pass sum also.

The beauty of the fourth-order low-pass is that midrange drivers roll off at their upper limit with a fourth-order characteristic to a good approximation over a wide frequency range, as I have found by measurement and least-squares regression. Obviously, some drivers are better in this regard than others. One can include the driver response in the passive network design and end up with a near-textbook approximation to the desired amplitude response. A nonlinear optimization routine can do this, such as Peter Schuck's network optimization program on the Madisound computer BBS. I found my own compensating dividing network, realizing a fourth-order response with direct radiators, to be rewarding.

In addition to the box diffraction effect, corrected in the Swan design by means of a second voice coil, smaller peaks and dips occur higher in frequency. These, too, can be accommodated by a compensating dividing network, even though they lie above the crossover frequency. An anechoic swept-frequency response measurement of the mounted drivers is necessary to reveal this, or a discrete-frequency anechoic measurement at a large number of frequencies. I have little faith in ⅓-octave noise measurements or in any other averaging method for gathering the data required for design purposes because the resolution is just too low for a high-accuracy design.

I wish the authors luck in their endeavors, but I would hope to see a more detailed optimization of the Swan system. After all, as they say, you must run twice as fast as the competition, just to stay even.

Victor Staggs
San Diego, CA 92109

Contributing Editor D'Appolito replies:

This letter raises a number of interesting points, however, the tone of your letter is a bit scolding, which entitles me to scold you in response.

It would appear you have not read our article carefully, for the design changes you suggest contradict our clearly stated design philosophy. You are obviously a proponent of the "constant power" school of design. This approach may be fine for sound reenforcement in large auditoriums, but it has little relevance to the accurate reproduction of two-channel stereophony in the home environment. You are certainly free to disagree with our design philosophy, but you should clearly label your "public address" system design to home sound reproduction for what it is—not the ultimate design, but only another design philosophy. It is most ungentlemanly of you to suggest that we have not done a complete design job simply because our system implementation does not meet your expectations.

Having said that, I would like to comment on specific issues raised by your letter.

Our purpose in presenting ⅓-octave frequency response data was to convey graphically the sound quality of the Swan IV. Based on our listening experience we find that ⅓-octave data correlates better than does the raw, or unaveraged data. We did not, as your letter suggests, use stepped ⅓-octave noise. The curves presented are *swept* ⅓-octave warble tone responses. These curves are, in effect, continuous frequency response curves that are then continuously smoothed with a sliding ⅓-octave averager. The resulting plots have substantially more resolution than *stepped* ⅓-octave noise responses and clearly reveal those aspects of loudspeaker response that relate directly to musical sound quality.

We did not use ⅓-octave data exclusively in the design process. Our instrumentation is highly sophisticated and quite up to the job of designing high-end loudspeaker systems. We use a PC-based loudspeaker analysis system consisting of a calibrated laboratory microphone, a Microway AD-160 data acquisition board and Doug Rife's MLSSA software. The hardware comprises a dual-channel 12-bit A/D converter with software selectable sampling rates, an eight-pole anti-aliasing filter also under software control and a pseudo-random maximal length sequence generator to obtain highly precise time synchronous samples of white noise, which we used to obtain loudspeaker impulse response.

The MLSSA software allows us to compute fast Fourier transforms (FFTs) of response data up to 64k samples in length for a resolution of better than 1Hz over the entire audio band. MLSSA determines impulse response, frequency response, phase response, group delay and Heyser's energy-time curve. It also determines driver mean acoustic center and identifies nonminimum phase behavior. Of course, the ultimate testing has been the dedicated listening of many "golden-eared," trained auditioners, whose thoughts have contributed to the final design.

We have measured the anechoic pressure response of the Swan IV outdoors down to 12Hz. It is quite smooth through the satellite/bass module crossover region and follows the bass module response curve, as given in our article, below that point. I had considered adding the compensation you suggest, but the departure from ideal first-order response, which is not evident in the anechoic response curves, would occur below 100Hz where our ability to perceive it is swamped by typical room effects (more on this later). The situation is quite different for the Vandersteen speakers because the crossover frequency between woofer and midrange drivers is much higher. The correction of this theoretical nicety would have greatly complicated the Pedal Coupler construction for *SB* home builders, with no sonic benefit.

Let me add at this point that both the Pedal Coupler design and some cabinet construction details have been simplified in the article, compared to the Swan IV production version. Both versions of the cabinet design are shown in the various *Photos*. These simplifications facilitate home construction of the Swan IV with no significant sonic compromise.

Acoustical power output is not an interesting quantity when talking about the low-frequency responses of home sound systems. If you are concerned with sound reenforcement in large auditoriums, there is a direct relationship through statistical acoustics, between radiated power and the sound pressure to which our ears respond. Within the boundaries of typical home listening rooms the assumptions of statistical acoustics break down at frequencies below 300Hz or so and we enter the so-called "discrete standing wave" mode. In this region sound power flow and sound pressure distribution are not simply related and the former is a poor predictor of the latter. We all know how difficult it is to get smooth low-frequency response in our living rooms.

It was certainly not my intent to "greatly disappoint" you, but the reasons for not using a constant-power crossover in the Treble Coupler were well documented in the article. For one, the 90° inter-driver phase shift associated with constant-power designs produces too broad a vertical response pattern. Furthermore, the even-order constant-power crossovers have been thoroughly discredited because they do not sum to flat frequency response. Beyond that, however, all the theoretical and experimental work I am familiar with, plus my own experience, leads me to believe the constant radiated power criterion has little to do with good imaging in a home listening environment. The majority of our clearly stated design criteria and goals cannot be met with constant-power designs. Our compensation for low-frequency diffraction loss (spreading loss) clearly produces a system which radiates more power at low frequencies. If the on-axis and near on-axis response is to be flat in the face of varying driver directivity, the constant power criterion must be abandoned.

Crossovers such as the Butterworth and Linkwitz-Riley have been analyzed *ad nauseum* in the professional and amateur literature. Many tend to believe that these networks represent the only valid crossover designs. Of course this isn't true. The critical lobing error problem is solved completely with my driver geometry. This freed me to examine the nontraditional crossover solutions.

My goals were simple—flat frequency response with drivers in-phase. The starting point for my design was the real world. First, the selected drivers were mounted in the desired geometry. Then, I made careful measurements of individual driver impedance, frequency and phase response, and mean acoustic center offset. The data constituted the starting point for a numerical and experimental design and optimization process, which is further described in my *Mailbox* reply to David Meraner (*SB* 1/89, pp. 57–59).

Incidentally, in its present form, the Schuck program was not useful for my purposes since it optimizes only individual driver responses against target functions. The proper approach is to optimize the summed response of all drivers against a target system response. It is my understanding that Dr. Schuck is modifying his program to do this. *[Vance Dickason reviewed Schuck's XOPT program in* Voice Coil, *Oct. and Nov. 1988—Ed.]*

The Focal drivers do indeed roll off to approximate a fourth-order response, but this occurs around 6kHz and has little effect on the Treble Coupler design. The excess phase shift caused by this rolloff is accounted for. Diffraction ripples above crossover were measured and carefully minimized by proper cabinet and grille design in the production system. Unlike low-frequency diffraction loss that is a spreading phenomenon, high-frequency diffraction is caused by reradiation from cabinet edges. As such, the resulting response irregularities are nonminimum phase and angle dependent. Compensating for these ripples produces a more nearly flat on-axis frequency response at the expense of greater group delay variation. Between 400Hz and 15kHz the group delay of the Swan IV is less than 0.2msec. It increases to 2msec at 100Hz and 5msec at 50Hz. These levels are well below the established limits of audibility and therefore the Swan IV is, for all intents and purposes, a linear phase transducer.

We chose to omit a summary of these elements of our design optimization process in our already amply technical article. Beyond these details, however, and at the risk of sounding self-serving, I point out that since publication of my AES paper and the original article (*SB* 4/84), my satellite design has been built by several hundred *SB* readers and copied by several manufacturers. Feedback from readers has been generous and constructive. Before our article was published, five systems were built and placed in the homes of respected auditioners for many months. We analyzed their comments and many were incorporated into the final design. The Swan IV has been in limited production for about eight months and has been auditioned favorably by professional musicians, recording engineers, high-end audio manufacturers and sales and marketing persons. Contrary to your statement, I believe we have done our homework well.

NAME THAT BOX

Reading *SB* issues from 1987–88, I've seen references to the use of two woofers in a single box that can be restated as follows: "If it is an Isobarik, V_{as} is halved, it is doubled otherwise."

This is correct, but only for tuned enclosures. I think we must be careful to clearly state to which type of box an equation applies. For closed-box systems, such as the Swan IV satellites, I use the following equation, deduced *ad hoc* from Small's passive radiator model ("Passive Radiator Loudspeaker Systems," *JAES*, Oct. 1974, Vol. 22, No. 8, Fig. 3, p. 593).

compound F_s = speaker $F_s/\sqrt{2}$
compound V_{as} = $(4 \times \text{speaker } V_{as})^2/$ speaker $V_{as} + V_b$

From this we can calculate F_b, Q_b and so on, using the usual T/S closed-box equations.

Jorge O. Oliveira
13083 Campinas-Sp, Brazil

SHORT SHORTLINE

I am in the process of building a pair of John Cockroft's Shortlines (*SB* 1/88), but have run into size and dimension problems and would appreciate some clarification.

Upon cutting all wood pieces per the parts list, I started assembly and found the rear panel is an inch or so short. Also, I assume this will make the front panel short and enlarge the port height.

With the top and bottom pieces as listed (9¼ by 9¼ inches), the only way to assemble the enclosure is to overlap the sides, and front and rear panels. However, I must recut the front and rear panels to fit. Then the front panel will depend on the exact height of the port, in relation to the bottom piece. How should I measure the 6½-inch port? Should the overall inside measurements be 36 inches high, and 8 by 8 inches?

Also, I could only find a Radio Shack 40-1021B woofer. Will it work just as well? The local Audax supplier sold me a DTW-100T25A Polydax tweeter and told me it was the same as the Audax HD100D8. Is this correct?

I have all the pieces scattered about my basement and would appreciate any help.

Foster Spain
Atlanta, GA 30327

Contributing Editor Cockroft replies:

I am sorry that, due to an error on my part, you have been unable to assemble your Shortline enclosure. The top and bottom pieces should be listed as 9¼ by 8 inches. The rear panel isn't short; it is sandwiched between the top and bottom panels, their ends being flush with the top face of the top panel and bottom face of the bottom panel.

The port is measured from the top of the bottom panel to the bottom edge of the front panel. The overall external (not internal) measurement is 36 inches; the length of the side pieces. This, of course, does not include the base. The inside measurement is 8 by 8 inches.

I have no experience with the Radio Shack 40-1021B. However a reader reported that he used a pair and was quite pleased with the results. He stated the Shortlines were "painfully accurate—especially on the Greatest Hits! But whatever is well recorded sounds really fine."

With that recommendation I assume the 1021B would be fine. To my knowledge, the two Audax tweeters are the same except the size and shape of the mounting plate differ. Incidentally, I am now recommending that you *do not* reverse the tweeter polarity, as shown in the crossover schematic (that is, connect the tweeter's positive terminal to the positive input connection, instead of the negative input connection).

I hope this answers all your questions and you will be able to clean up your basement and begin to enjoy some music.

OCTALINE/ RADIO SHACK TESTS

I have a few comments in response to Mr. John Cockroft's letter reply in *SB* 4/88 concerning the Radio Shack 40-1022 drivers.

When I began constructing a pair of Octaline speakers in March, I was lucky to locate the high frequency drivers still on the shelf at the nearest Radio Shack, but only one of the 40-1022 low frequency drivers. After scouring the town, I had to settle for one of the Korean wonders, the 40-1022A.

It is unfortunate that the Radio Shack mentality quickly relieves consumers of the opportunity to use an exceptional product as soon as its merits are discovered!

To see how bad the mismatch would be, I measured as I went along and was fortunate to come much closer than the unit Mr. Cockroft sampled. Just the same, there's a noticeable difference between the original and the newer A model.

Our measurements seem to indicate considerable tolerance variations in the A models. However, with a little understanding from the local Radio Shack manager and some hand selecting with F_o, you might find a pair resembling the originals close enough to yield satisfactory results.

The accompanying chart compares the numbers of the 40-1022 and 40-1022A models I used: first, out of the box; then, after a break-in (60Hz @ 2W for 30 hours mounted in a 0.5-ft.^{3} test box); and finally, after all of Mr. Cockroft's modifications.

RADIO SHACK DRIVERS

Specs	F_0(55Hz)	Q_{TS}(0.35)	V_{AS}(0.23 ft.^{3})
OUT OF BOX			
1022	49	.319	.311
1002A	52	.371	.223
AFTER BREAK-IN			
1022	52	.346	.225
1022A	52	.371	.264
AFTER MODIFICATIONS			
1022	40	.420	.228
1022A	39	.428	.164

A couple of years ago, *SB* introduced me to the excellent EJ Jordan System 5, which I admit has spoiled me rotten, but I did find the Octalines quite nice and was surprised at their extended low-frequency response.

In review, I would say they have a very slight boominess on the male voice but a very acceptable musical sound with nice stereo results—far outweighing anything on the market for the small expense involved. They are, however, quite inefficient—the obvious trade-off for the extended response—but a very good sound for a very small budget. Mine work well in the office, driven by a very old amplifier of probably 3W per channel. Very nice, but loud? No, disco they ain't!

Thanks to Mr. Cockroft for the design and his continuing experiments with TL cabinets.

Rodney E. Cavin
Altamonte Springs, FL 32715

Contributing Editor Cockroft replies:

I thank Mr. Cavin for his efforts in documenting his two Radio Shack speakers. He seemed rather lucky in his purchases. The two speakers come out close to two percent in their important parameters. I have not found the need to consider V_{as} in transmission line design.

I also appreciate his review of the sound of the Octaline. When listening to some radio stations, I too have detected a bit of heaviness in the male voice, but when listening to well-recorded opera recordings and certain live broadcasts from Berkeley station KPFA, which frequently uses exceptional technique in such cases, I am not aware of this condition. Many radio stations use recording studios the size of telephone booths to record voice, which results in this unnatural condition. However, possibly the Octaline, in locations other than I have heard it, could be responsible.

The Octalines are inefficient. They were designed to be used in rather small rooms, such as in apartments. I have received complaints from neighbors on occasion that I was playing my music too loud, although I seldom play them at such levels. I prefer naturalness of reproduction to the production of sonic booms. While 3W per channel might be satisfactory for low level sound in an office, the tweeter could be damaged at higher levels when the amplifier would be driven into clipping distortion. I would think that 50W per channel would be a reasonable minimum. I use an 80W per channel Parasound HCA 800, Class A amplifier and find it quite satisfactory. On the other hand when using a 35W Sony receiver at relatively high volume, amplifier distortion was readily apparent in high level passages.

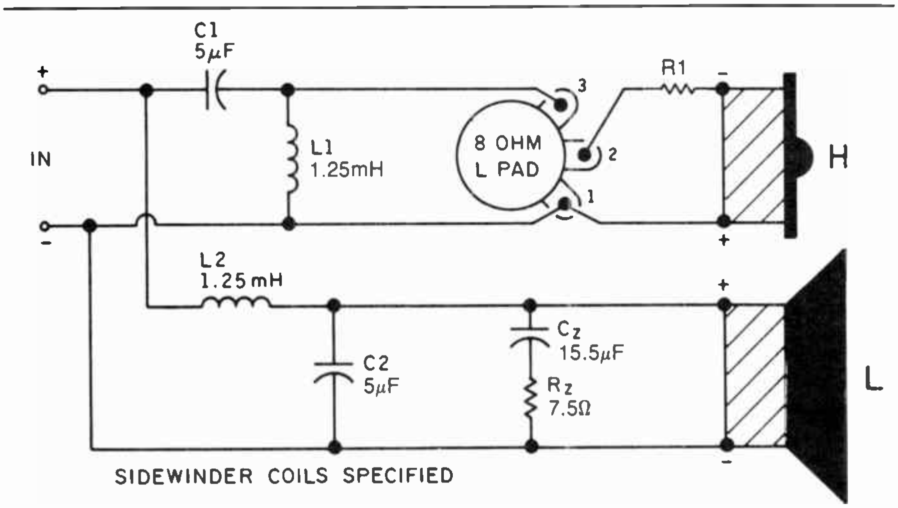


FIGURE 1: Shortline crossover modified for the Peerless KP825WFXPP woofer. For R1, use 1Ω with the Audax tweeter; for the Peerless tweeter, use 0.5Ω or just eliminate it.

SHORTLINE SUBSTITUTIONS

I just subscribed to *SB* for two years and loved issue 1/88. I liked the article "Shortline: A Hybrid Transmission Line." I want to build it, but I have run into the same difficulty I experienced when I bought David Weems' book, Radio Shack changed the drivers he uses.

The list of materials states, "do not substitute" about the Radio Shack 40-1021 8-inch polypropylene woofer. The current catalog has substituted a "new for 1989" 8-inch polypropylene 40-1024 with different specs.

Worse, I went to my local RS and they have the 40-1021B. What is the answer? Should I build the Shortline with the "B" model?

Another question: the cross-sectional diagram suggests by relative dimensions that the "stiffeners" are about 1.5 inches thick; I wonder what the thickness should be?

Larry Davis
Bedford, TX 76021

Contributing Editor Cockroft replies:

I'm afraid I can't advise you on the suitability of the Radio Shack 40-1021B. The spec sheet gives the same parameters the original spec sheet contained. These differed from the original driver I used and they may be different than the actual 1021B. It may work, since transmission lines are rather tolerant of drivers, but then again it may not. I'm not ready to start another Radio Shack merry-go-round chase.

Shortly after the Shortline article appeared I redesigned the crossover for a friend to use the Danish Peerless 8-inch KP825WFXPP. It seems satisfactory to me when installed in the original Shortline. My friend built several pairs using the Peerless woofers and claims they sound better than the first ones he made using the Radio Shack woofer. I am enclosing a copy of the crossover schematic I sent him.

Later I replaced the Audax tweeter with the MB Quart MB MCD-25 titanium dome 1-inch tweeter. This resulted in a smoother top end (at almost double the tweeter cost, however). The Peerless 010DT8 is also a good choice. When I installed the MB tweeter, I used the Zobel impedance compensating circuit recommended by Dick Olsher for that tweeter in his "Dahlia Debra" article in *Stereophile*. This consisted of a 7Ω resistor (a 10Ω resistor paralleled to a 25Ω one comes close, about 7.1Ω) and a .56μF capacitor. This is shunted across the tweeter leads just like the one for the woofer in the schematic (*Fig. 1*). It doesn't seem to make any difference as far as I know which way you wire it.

The stiffeners are 1¾ inches wide by ⅝-inches thick as specified in the Parts List on page 22 of the article. The thickness isn't critical. I merely used the same material as the rest of the enclosure for simplicity.

DREAM STUF

I thank John Cockroft for his review of Acousta-Stuf in *SB* 4/88. His comments are indeed most gracious. As a purveyor of "High Grade Fluff," we value his opinion. His articles have been a source of enjoyment and information over the past few years, and the equations in his recent article, "The Unline," prove that he is very qualified in his chosen area of expertise.

Some will say his "model" was not a working system, and his equations, therefore, must be flawed. But if all of his proven systems fall in line on his table of data, then the "model" would also work well. I decided to run mathematical relationships on the numbers in John's table, and came up with a very interesting point. Dividing (Cs1/d1), all the way down the list, came up with a constant relationship of 2.0. In other words, stuffing density was always half of the cross-sectional area figure. I'm not sure how important this factor is, but I thought it worth mentioning so it might be discussed, as I'm sure, the entire article will be scrutinized by your readers.

Let me point out one more thing about Acousta-Stuf. John mentioned that he prefers a more reactive material like fiberglass for vented systems. We think many builders use fiberglass, not because it works better in vented boxes, but because it is flat, which makes it easy to use for lining enclosure walls. Most builders tend to think of resistive fibers as being great for TLs and sealed boxes, but not very good for vented systems because you don't stuff a vented box, you line it. Since Acousta-Stuf is such a cohesive material, we recommend fastening wads of the fiber to enclosure walls with 3M spray adhesive to cover the desired areas and leave plenty of open air for reflex action to occur. The results are quite ear pleasing, and once you try this method, we don't think you'll want to go back to itchy fiberglass. Acousta-Stuf is also great for spot absorption panels to control room acoustic problems. It has vehicular uses also, as insulation in customized van walls, for example. In fact, its uses are so numerous that we can't begin to imagine them all.

Larry D. Sharp
Mahogany Sound
Mobile, AL 36695

DISPERSION DISCUSSION

The questions I must ask Mr. D'Appolito have arisen after reading an article in Harris Publications' *Audio Video Speakers Buyer's Guide for 1986*, "Specs for Judging Speakers." In this article, the author emphasizes the size of a driver in question as being the key element in determining dispersion characteristics for that driver. Put as succinctly as possible, the theory is: "Sound will radiate straight forward if the radiated wavelengths are shorter than the radiating area [of the cone or dome] and sound will radiate at a wide angle if the radiated wavelengths are longer than the radiating area."

By "area," the author is referring to the diameter of the radiating area specifically, or at least the area represented by its diameter (I mention this because of the discrepancy between linear and square units as it appears in the above stated theory). In any case, given that the wavelength is equal to the speed of sound divided by the frequency, we can determine the dispersion characteristics of a driver at a given frequency by its diameter. Because of the inverse relationship between piston diameter and frequency, we can see (what has been supported by experience) that two-way combinations using large woofers tend to "beam"—do not image well.

Let's look at an example. At sea level the speed of sound is approximately 13,376 inches/second and the frequency under consideration is 2,200Hz. I have an eight-inch woofer (Peerless) in a second-order two-way system, at the aforementioned frequency. Ideally, the maximum piston diameter at this frequency should be 13,376 inches per second/2,200Hz = 6.08 inches.

Did I say ideally? Yes, for an ideal piston match, we need to look at the two octaves on either side of the crossover frequency. Thus we are brought to 4.4kHz as being the upper frequency limit my eight-inch woofer will have to deal with. Plugging this into our equation, we get the painfully small diameter of 3.13 inches.

Believe it or not, this beaming is audible around the 2–2.5kHz region. Will the "D'Appolito configuration" help me? That is, if I go out and purchase a second set of woofers, and the appropriate crossover, will I get better dispersion?

The second question I have concerns Polk Audio's Monitor 10, a speaker I once owned which imaged remarkably well. Looking at the driver configuration, though, you'll notice the two bass/midrange drivers are placed close together in a horizontal axis, with the tweeter above, forming a triangle. Shouldn't this create a vertical dispersion pattern, as opposed to a horizontal one?

My last and most important question: if

I use 2-inch dome midranges, such as the Peerless KJM-20, in the D'Appolito configuration, would their smaller size, (and thus better dispersion characteristics) cause them to interfere with each other? If so, should I use some sort of felt or foam to prevent this? I would direct you to the article about Duntech's Sovereign 2001 (*Audio*, November 1986), a speaker which employs two dome midranges in this configuration.

Stephen K.M. Wakeman
W. Lafayette, IN 47906

Contributing Editor D'Appolito replies:

There certainly is a dimensional discrepancy between cone area and wavelength. As you surmised, the proper relationship is between cone diameter and wavelength. For a circular piston on a large baffle, the polar response is down 3dB relative to its on-axis response at ±30° when the piston diameter is equal to the radiated wavelength. At higher frequencies the polar response becomes even narrower. Patterns narrower than 30° can lead to poor off-center imaging and should be avoided.

The average 8-inch speaker has an effective diameter of about 6 inches which, as your calculation shows, leads to a 30° half-power pattern at 2.2kHz. This is an acceptable crossover frequency for an 8-inch driver. I believe your criterion of an octave overlap in dispersion response is too severe. You must remember the tweeter will fill in the off-axis response in this region. Several two-way 8-inch systems on the market with crossovers as high as 3kHz image well. (I must add that I know of drivers with sophisticated cone contouring or progressive decoupling, which decreases the active driver area with increasing frequency, and thereby maintains a wide dispersion pattern.)

You must distinguish between horizontal and vertical dispersion. Good horizontal dispersion is needed for good stereo imaging. Vertical dispersion can be relatively narrow as long as it is stable. The purpose of my geometry is not to improve horizontal dispersion, but to stabilize the vertical radiation pattern. The horizontal patterns are not affected so the criteria for single drivers also apply to my geometry in the horizontal plane.

Concerning the Polk Monitor: on the surface, its geometry would seem to present imaging problems. It is my understanding, however, that only one of the two 6.5-inch drivers is active in the midrange, thereby eliminating the potential for horizontal lobing.

Finally, you certainly can use two midrange domes with a central tweeter in my geometry. For best results keep the drivers as close together as possible, preferably no farther apart than one wavelength at the mid/tweeter crossover.

SANDERS: CONSTANT CHARGE PROBLEMS

In *SB* 1/89 ("Mailbox," p. 54), Peter Muxlow states the Beveridge loudspeaker operates in a "constant charge" mode, despite low diaphragm impedance, because the stators are high impedance. Perhaps the problem is one of definition, but I disagree with his assessment.

A high impedance diaphragm solves the problem of charge migration on the diaphragm. The diaphragm's electrostatic charge tends to move toward that area of the stator that is closest to the diaphragm. Since it is impossible to build perfectly flat stators, this will always be a problem, but is worse at low frequencies that allow more time for the charge to move. High diaphragm impedance inhibits this.

If we accept this concept, then the term "constant charge" refers to using high impedance on the diaphragm to prevent charge migration. It has nothing whatsoever to do with the impedance of the stators.

As an interesting observation, aside from the discussion of constant charge mode operation, high impedance stators are clearly inferior to low impedance ones because we want the audio signal to be able to change rapidly on the stators for good high-frequency response over the entire radiating area. If the stator's impedance were high enough, we would find that the area of the stator farthest from the amplifier contact point would have a lower output at high frequencies, compared to areas near the contact point. The high stator impedance slows the necessary electron flow that builds the electrostatic charge on the stator. This problem will be reduced at lower frequencies because the charge would have time to travel to the farther stator edges. The net result would be a speaker with a falling high-frequency response in some areas while maintaining flat response in others—certainly an undesirable situation.

Roger R. Sanders
Halfway, OR 97834

MIDRANGE MISSION

George Li asked (*SB* 6/88) for a Gary Galo article testing midranges. I too, as many others are, am searching for that perfect sounding midrange. But no one has the time or money to test samples of everything out there, so I propose that everyone who would like to participate send me a postcard with your favorite midranges in three categories:

1. under $25
2. under $50
3. who cares what it cost (over $50).

You are welcome to add a little note explaining why it's your favorite. I'll collect and compile the data and send it to *SB* for a future article. You must admit, most of the articles you see are for woofers and tweeters. I think we could all use some help choosing the unit that gives our system its tonal quality.

John Yanek
PO Box 724
Vernon, CT 06066

A PAT ON THE BACK

I cannot express to you adequately how much I have enjoyed my first year reading your publication. Your Editor is to be particularly complimented for his "mix" of articles. Well done.

Sid Smith
W. Newbury, MA 01985

SANDERS' "BOAR GUN"

I agree with Bruce Edgar's comments on cordless drill/screwdrivers ("The Cordless Revolution," *SB* 1/89). I had similar experiences, but I explored the situation a bit further.

I have the same Skil ⅜-inch cordless "Boar Gun" that Mr. Edgar bought from

Sears. Although reasonably powerful, it was not completely adequate for driving long screws or for driving lag screws with a socket. I also wanted to do some serious drilling in steel with my cordless and I therefore rewound the motor for more power. You might want to consider doing the same.

Motor power is a function of "amp-turns." In other words, to increase power you must increase the current that passes through the rotor, increase the number of turns in the rotor winding, or both. This can be accomplished in a number of ways. Increasing the voltage will drive more current through the rotor, but at the risk of overheating it. Also, increasing the voltage of a commercial battery pack is difficult. Increasing the number of turns requires that you reduce the wire size, or increase the motor size. Making a larger motor is usually out of the question. Using smaller wire and more of it to fit on the original rotor increases the resistance and reduces current, which reduces power more than adding turns increases it.

The reverse is also true, fortunately. We can use larger wire to reduce the number of turns, however, the larger, shorter wire passes much more current and increases power far more than the loss of turns reduces it.

Rewinding a motor is not as difficult as you might think. The biggest problem is getting large magnet wire. Such wire is available from electric motor repair shops, but many may not wish to sell you just a few yards.

To rewind the motor, disassemble it and unwind the rotor windings. Be careful to identify which slots the wire goes into, to match the commutator section to which each lead attaches. Also observe the direction of the windings. I suggest making a simple drawing.

I did this project several years ago and do not recall what size wire I used, but it measures 0.038-inch in diameter and is about 50% larger than the stock windings.

Rewind the rotor and ensure the windings are tight to put on as many as you can possibly cram into the rotor. Solder the ends to the commutator. Temporarily assemble the unit to check that the new windings do not rub the magnets. The rewinding process will result in the motor running at much higher speed than stock. To keep the windings from flying out of the rotor, you must coat them with epoxy. Apply the epoxy to the rotor and then use a heat gun or hair dryer to obtain better penetration and quicker curing.

This modification nearly doubles the power of the unit, increasing both torque and rpm by about 50%. Of course, the price of this power is reduced battery pack endurance and owning two battery packs is essential. Not surprisingly, I eventually burned out the trigger assembly and replaced it with a genuine solid-state unit rather than the original mechanical vibrator type, and it has given me no further trouble. I have had no trouble with the motor. The improved performance has made this modification worthwhile.

As good as cordless screwdrivers are, they are vastly inferior to air-driven impact screwdrivers, which are less expensive, have far more power, drive screws nearly instantaneously, never break screws or strip heads, run cold, never burn up, are lighter, smaller and more compact, and have no inertia. Beyond these advantages, I haven't mentioned the most valuable characteristic of an air screwdriver: you need not force the bit against the screw. At first glance this may not seem like an issue, but I believe the biggest problem with driving a screw is keeping the bit in heavy contact with the screw head so that it doesn't slip or strip the head when torque is applied. If it does slip off while you are pushing hard, the bit damages your project.

An impact air screwdriver operates by hammering the screw in rotation at several thousand impacts per minute. Each impact is so quick that the bit cannot back out of the screw far enough to slip. You can drive or remove the toughest screws while just barely touching the screw with the air driver and you can handle jobs where you can just barely reach the screw—otherwise impossible with the force needed to hold a cordless against the screw.

Also, with a power drill/screwdriver that does not have a clutch, when you drive the screw home the power screwdriver often slips and damages both the screw head and the bit. You can adjust an air driver's torque to a specified amount and although the driver continues to run when the screw is home, it simply stops turning the screw and does not slip.

Also, air tools stop and start instantly, unlike an electric device that keeps turning after you release the trigger. This is a more valuable feature with drills than with screwdrivers. With a large and powerful electric drill, if the bit jams, the rotor inertia suddenly increases torque many times above maximum power and can literally break your arm even though you release the trigger. Air drills of similar power simply stop when jammed, while holding essentially constant torque even if you keep the trigger pulled. They also have all the previously mentioned advantages of light weight, compactness, burn-out proof, and so on.

Air tools do have one disadvantage, obviously—they need a source of compressed air—but this is not as much of a limitation as you might think. Assuming you have an air compressor, there is no problem using air tools in your garage or shop. The problem is portability. As long as you do not intend to do extensive work, you can get an old "Freon" tank from an air conditioning/sheet metal shop, connect a short air hose to it and fill it from your compressor. This tank is light, portable, cheap, and you can take it to a remote work site to drive your air screwdriver. You can drive a lot of screws with a small tank because the air driver works quickly.

Roger R. Sanders
Halfway, OR 97834

Laboratory Loudspeaker

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sistance, but they are more expensive. *Do not* substitute components.

An alternate tweeter is the Philips ADO163/T8. The only difference between this and the one specified is that the case is round instead of square. An alternate midrange is the Philips AD50600/Sq8. This midrange does not give as good a frequency response, but it may be used if the Peerless unit is unavailable.

Muses and Music

Since the music moves you, the muse is almost surely able to do so as well—the writer's muse, that is. Put pen to paper or better yet, typewriter ribbon to paper with a clear, orderly account of your adventure in audio construction, or any related field of endeavor leading to good listening. Send it along with a stamped, return envelope. We pay modestly for articles, so write us about it and we'll answer promptly with suggestions and tell you whether or not we have such an article, whether one is already in preparation, or whether we are interested. Some of our best articles come from people who have never before written for periodicals. And if your muse is as silent as a tomb, don't let that stop you. Write anyway and let's see what develops. We have a nice sheet of suggestions for authors which we will send to nearly anybody who asks for it.

Passively Assisted

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positive lead to capacitors from banana input. Attach lead from caps to positive woofer terminals. Attach negative lead to negative woofer terminals.

7. Measure system F_{sb}, Q_{es} and Q_{ms}. If Q_{es} is less than 1.1 and F_{sb} is too low, place a small object in the box to reduce volume. If Q_{es} is greater than 1.1 and F_{sb} is too high, add polyester fiberfill to increase apparent box volume. If Q_{ms} is greater than 10, add more fiberfill until design goal is attained. If Q_{ms} is less than 10 throw up your hands in despair (but don't let it get your spirits down). Note: series caps must be shorted for these measurements.

8. Finish cabinets as desired.

9. Let 'em rip.

Author's Note: Bill Goodwin at Precision claims production of the TA305F will begin in April. The driver is currently unavailable. The Madisound 1054 10-inch woofer (f_s, 22Hz; Q_m, 4.3; Q_e, 0.29; V_{as}, 7 ft.^{3}) should be an acceptable alternative, giving nearly as good results (f_3, 37Hz and V_b, 1.1 ft.^{3}).

FOOTNOTE

1. Clark, D. and E. Geddes, "Passively Assisted Loudspeakers," SAE Technical Paper Series #860123, 400 Commonwealth Drive, Warrendale, PA 15096.

IC Power Amp

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In fact, the improved sound inspired me to add a subwoofer to my own (mono) television system, with very pleasing results. My main speaker is an Audax AM78GSM shielded 3½-inch, full-range driver, and the "subwoofer" is a Madisound 81524DVC 8-inch woofer in a vented enclosure. I inverted the subwoofer's polarity and the first-order active crossover occurs around 150Hz.

Home-Built Spikes

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fer; however, after many hours of critical listening, I believe I hear an improvement. My three-way system uses the Dynaudio 21W54 woofer in a closed-box design with aperiodic pressure relief. The bass and midrange have always been quite clear and "alive," with great speed; however, the system lacks much low-end extension. With the addition of the spikes, transients—especially in the midbass and lower midrange—are even more impressive. The sound throughout this region also seems clearer and more detailed. These are, to some degree, the gratifying effects I expected when I embarked on this adventure.

I believe, however, that I also have an added bonus: a noticeable improvement in low bass response. Using a purely subjective evaluation, I estimate that with the spikes I hear about a half-octave of low bass that was not apparent before. I'm not sure why this results. Perhaps other readers can offer an explanation.

This was an interesting project for me and I encourage you to try it. It should be particularly worthwhile for persons who, such as myself before this undertaking, are unsure of the potential benefits of speaker spikes. You can experiment with my design and verify or dismiss the claims without a major investment in time or money. Even if you doubt that spikes are better, isn't it worth $5 to find out?

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FOR SALE

Two JBL systems including N1200s, in use, excellent condition, best offer plus shipping; DeCoursey 1200Hz crossover, $50. (206) 927-2677, evenings.

Hafler DH500P (2) with pro features, $450 each; DH-110 preamp, $260; DH-120 (2) as new in boxes, $275; Morel PPC8 (pair) as new in boxes, $200. Dan (801) 225-8577.

Sony 6½-inch square woofer, 5Ω, 10W nom., 80W max., $35 pair; Sawafuji Dynapleat 6½-inch square midrange, 400Hz-4kHz, 5Ω, $35 pair; Focal 5N415S 5¼-inch woofer, long throw, 15Ω, as new, $35 each, $160 for 5. Joe Mcquade, (609) 397-8315 (NJ-EST).

McIntosh MC-30; Altec: 416-8As, 811Bs, 511As, 804-16A, 804-8Bs and 806s; some in green, some in Heath private label; JBL LE-14s. Sell or trade for Lionel train equipment. Mike Rockwell (616) 429-9271 after 6 p.m. EST.

Thirty-two cubic foot subwoofer cabinet: MFD fiber board, hardwood bracing, sealed cabinet, black with black grille, sand-filled front baffle, Tiffany binding posts, designed for two 18-inch Hartleys (not included), $500 or best offer. Charles (415) 833-0177.

Sony D-15 Discman, spare battery, case, mint, $275; CD pouch, $10; JBL 2234H 15-inch woofers, excellent condition, $350 pair, in studio enclosures, $450 pair; JBL 2221 10-inch midrange, excellent condition, $125 pair; Textronics tube scope, $250 (pick up). Steve (203) 397-3888, 7-10 p.m. EST.

SEAS 10-inch woofer from Dynaco A-25; two Community Light and Sound SRH-90 fiberglass radial horns cut for Altec Drivers; Dynaco FM-5 tuner. All items excellent, best offer. A. Lewis, 3929 S. Sherman, Englewood, CO 80110, (303) 781-5573.

National Semiconductor LM12CHK power op amps as featured in *SB* 4/88. Includes factory data sheets. Brand new, never used. Cost $30, sell for $20. 8 available, Harold (301) 661-1202.

Dynaudio drivers, used once for test only. Two D52AF, $45 each; two 24W75, $40 each; 30W54, $80. All mint condition. A. Nittoli, 4 Broadview Ave., Kings Park, NY 11754, (516) 269-9796.

New JBL 2245H 18-inch woofer, $275; Gold Line ASA30B 1/3-octave spectrum analyzer with pink noise generator and Gel cell recharger, $515. I will pay shipping. Jerry Saucier (601) 264-6971.

Hermeyer ESA-3 parts: oil capacitors, bridge rectifiers, plate/bias transformers, call for details; ESA-3 circuit board set, $15; 70,000μF 25V DC Mallory CGS, $20 each; DH-200 rack kit, $20 each; Soundcraftsman 20-12 equalizer, $75; stereo subwoofers, VMPS larger subwoofers, mint $700 pair. Nick Mastrobuono, 50 Copperfield, Sarnia, Ontario N7S 5K8, Canada, (519) 336-9160.

Dennison Polaris antenna complete, slightly used; Realistic 12-1982 underdash deck, search, metal, Dolby, headphone jack, slightly used; SAE 5000 impulse noise reduction system; Polydax tweeter. Sell or trade. (815) 657-8488.

Pair of Jordan 50mm module line arrays, four drivers per cabinet in Transcendental Audio "unboxes" (semi-tubular damped enclosures); two JVC ribbon tweeters with air core/poly crossovers, $425 for all. Dan Grogan, 11352 Gatewood, Dallas, TX 75218 (214) 328-8914.

Ampex 300-2 ¼-inch, ¼ track deck, new tubes, heads, like new, $1,200; Ampex AG-440 ¼-inch ½ track, good heads, in portable cases, $1,600. Both decks aligned and calibrated, in excellent condition. Jeff A., 871 Cottonwood St., Corona, CA 91719, (714) 735-7487 evenings.

Philips 12-inch bass drivers #12A251, $50 pair; subwoofer Becker 10-inch DVC and 10-inch pass rad in 2.5 cubic inch enclosure, $40; Ace Audio #5000 mono active crossover 80Hz at 18dB, $25; Sound Concepts IR-2000 image restoration control, $50; Advanced Audio Systems DNR unit #911, $50. Ed Weigand, 44 Inverness Rd., Holbrook, NY 11741.

Two pair gold ribbons, new, half price or best offer; pair 35 foot Monster Cable, $30; pair Morel Integra I 6-inch two-way auto speakers, $40; dual voice coil 15-inch subwoofer in walnut cabinet AC compact monitor style, $40; Levinson ML-9 amp, mint, $1,900; Jim Ciechanowski, 1046 Robin Way, Sunnyvale, CA 94087, (408) 738-0559.

Test equipment, scopes, electronic magazines. Pick up only. Consider Commodore C-64 in trade, etc. Nick, 203-589-6223 after 6 pm EST.

Altec, W.E., Emilar, Speaker Lab horns and drivers; Altec, RCA, W.E. literature on amps, speakers, transformers and tubes from 1930s-50s; hi-fi books and magazines from pre 1960s. V. Vogt, 2311 Pimmitt Dr. #114E, Falls Church, VA 22043, no calls.

Audio Concepts JCRS style subwoofer with twin Dynaudio 30W 54 drivers, professionally built and finished in ash and elm burl, very sharp, $450; 4 pair Jordan speakers, $80/pair; B & O 2400 turntable, $50; DBX 3BX range expander, $100. Allen Dixon, 1479 151st Ave., San Leandro, CA 94578, (415) 278-6422.

Rega Planar 2 with Adcom HC/E II cartridge, Mission sorbothane mat, Audioquest headshell, plus accessories, perfect; Luxman L-5 integrated amp; approximately 250 mint records; Pioneer TVX-9500 TV tuner. Offers or trade for audio, video, computer, or camera equipment. Kent Johnson, 2985 Quitman, Denver, CO 80212, (303) 455-5610.

dbx 4bx three-band expander with impact restoration, volume control, mute, expansion ratio all by wireless remote, includes box and manual, $400; Audio Pulse Model One digital time-delay system with White Labs update modifications (details available), excellent manual included, $300. Tom Meacham, 9500 Prospect Drive, Anchorage, AK, 99516, (907) 346-2981, evenings.

Breaking down tri-amp system. Heathkit crossover, $175; Crown VFX2 crossover, $175; both $300. Sansui 999 integrated amp, $175; Belles II basic amp, $200. I pay shipping. Jeff Grycz, 8559 17th Ave. N.W., Seattle, WA, 98117, (206) 783-5456, after 6 pm during week, anytime weekends.

WANTED

Philips (Amperex, Norelco) AD8066/W8 8-inch woofers, ADO160/T8 dome tweeters. Also, damping setup for SME III arm. Gary Judd, 3457 North 84th St., Milwaukee, WI 53222, (414) 224-2701 days, (414) 464-6447 evenings.

Source for 1 and 2-inch fiberglass other than Radio Shack. Source of veneer 12 inch width without seam. Focal 10N501s. M.J. Thompson, 194 St. David Rd. W., St. Catherines, Ontario, L2T 1R4, Canada, (416) 227-4586.

Dynaudio mid-bass drivers; Pyle mids; Bose 301 woofer; Peerless tweeters, or just about anything. (815) 657-8488.

Acoustical Engineering, by Harry F. Olson. H. Ray Mills, Jr., 2345-B Salem Court, Winston-Salem, NC 27103.

Loudspeaker testing equipment: B&K mikes, SA 3050 analyzer, pulse and tone burst generator, etc. Also Win Labs MC-10 jewel moving coil cartridges. Dave (213) 924-2666, in California.

Electrovoice "Temples of Tone" brochure, other EV brochures; 78 RPM record, "March of the Caucasian Chief" (Apolitoff Evanoff); Miller 565 Crystal AM tuner; old Klipsch catalogs and crossover networks/components; *Audio Engineering* volumes 1-5; diaphragm for International Projector Corporation LU-1000 hf driver; *The Throne of Merlin* by R.C. Schaller. D.R. Schaller, 6704 Schroeder Rd., Suite 6, Madison, WI 53711.

JBL 12-inch and 15-inch woofers. Preferably in good overall condition but at least reconable. David Adams, 2015 Paisley Dr., Arlington, TX 76015.

Any information on modifications/upgrades to the original Mission 770 loudspeaker will be greatly appreciated. Peter Lau, 305 Cherry St. #F3, NY, NY 10002.

Two JBL 077 high frequency drivers and two JBL LE85 compression drivers. George Derderian, 3743 N. Delaware Ave., Arlington, VA 22207, (703) 538-6061.

CLUBS

Space in this section is available to audio clubs and societies everywhere free of charge to aid the work of the organization. Copy must be provided by a designated officer of the club or society who will be responsible for keeping it current. Send notices to Audio Clubs in care of the magazine.

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

HI-FI CLUB OF CAPE TOWN, South Africa issues monthly newsletter for members and subscribers. Get a different approach to understanding audio, send two IRCs for next newsletter to PO Box 18262, Wynberg 7824 South Africa.

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

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

THE WESTERN NEW YORK Audio Society (WNY Audio Society) is an active and growing audio club located in the Buffalo area. We issue a quarterly newsletter and hold meetings the first Tuesday of every month. Our meetings have attracted many local and distant manufacturers of audio related equipment. We are involved ein all facets of audio—from building to purchasing at discount prices. For a copy of our current newsletter and information regarding our society, please write to M.A. Monaco, WNY Audio Society, PO Box 312, N. Tonawanca, NY 14120.

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

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

TUBE AUDIO ENTHUSIASTS. Northern California club meets every other month. For next meeting announcement send a self-addressed, stamped no. 10 envelope to Tim Eding, PO Box 611662, San Jose, CA 95161.

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

THE VANCOUVER AUDIO SOCIETY publishes a bimonthly newsletter with technical information, humor and items of interest to those who share our disease. We have 40 members and meet monthly. Six newsletters per year. Call (604) 251-7044 or write Dan Fraser, VAS, Box 4265, Vancouver, BC, Canada V6B 3Z7. We would like to be on your mailing list.

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

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.

AUDIOPHILES IN CENTRAL PENNSYLVANIA (also eastern Pennsylvania and Delaware): Interested in forming a serious audio organization? Contact Steve Gray, 625F Willow St., Highspire, PA 17034 or phone (717) 939-4815.

SOUTHEASTERN MICHIGAN WOOFER AND TWEETER MARCHING SOCIETY (SMWTMS). Detroit area audio construction club. Meetings every two months featuring serious lectures, design analyses, digital audio, AB listening tests, equipment clinics, recording studio visits, annual picnic and audio fun. The club journal is *LC, The SMWTMS Network*. Corresponding member's subscription available. Call (313) 477-6502 (days) or write David Carlstrom, SMWTMS, PO Box 721464, Berkley, MI 48072-0464.

THE BOSTON AUDIO SOCIETY INVITES YOU to join and receive the bimonthly *B.A.S. SPEAKER* with reviews, debates, scientific analyses, and summaries of lectures by major engineers. Read about Apogee, Nytal, Conrad-Johnson, dbx digital, Snell, music criticism and other topics. Rates on request. PO Box 211, Boston, MA 02126.

PACIFIC NORTHWEST AUDIO SOCIETY (PAS) consists of 50 audio enthusiasts meeting monthly, second Wednesdays, 7:30 to 9:30 p.m. at 4545 Island Crest Way, Mercer Island, Washington. Be our guest, write Box 435, Mercer Island, WA 98040 or call Bob McDonald, (206) 232-8130.

THE ATLANTA AUDIO SOCIETY, a club for men and women interested in high quality home audio/video systems and recordings. Monthly meetings are conducted where special programs and guest speakers are often featured. Guests have been: Jack Renner, Richard Vandersteen, William Conrad, William Johnson, Louis Lane, John Cooledge and others. Annual dues are $25 and includes a monthly newsletter. For information call Chuck Bruce, (404) 876-5659 or write Atlanta Audio Society, PO Box 361, Marietta, GA 30061.

SAN DIEGO AUDIO SOCIETY forming for hi-fi tinkerers and do-it-yourselfers. If you enjoy collecting, building, rebuilding and repairing classic audio equipment, especially tube-type, call Mike Zuccaro (619) 271-8294 (evenings & weekends). Old timers and engineers welcome.

NEW JERSEY AUDIO SOCIETY meets monthly. Emphasis is on construction and modification of electronics and speakers. Dues includes monthly newsletter with high-end news, construction articles, analysis of commercial circuits, etc. Meetings are devoted to listening to records and CDs, comparing and A-Bing equipment. New members welcome. Contact Bill Donnally, (201) 334-9412 or Bob Young, 116 Cleveland Ave., Colonia, NJ 07067, (201) 381-6269.

SIXTIES MUSIC FANS. The first club for fans of Sixties Era music wants to meet you. Strong audio and quality recording emphasis. How-to information shared. Worldwide membership. Informative, entertaining, and provocative newsletter published bimonthly. Help the world remember when rock had real artistic merit. Free brochure: send SASE or IRC to Classic Rockers Music Club, PO Box 1043C, Stevens Point, WI 54481.

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

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

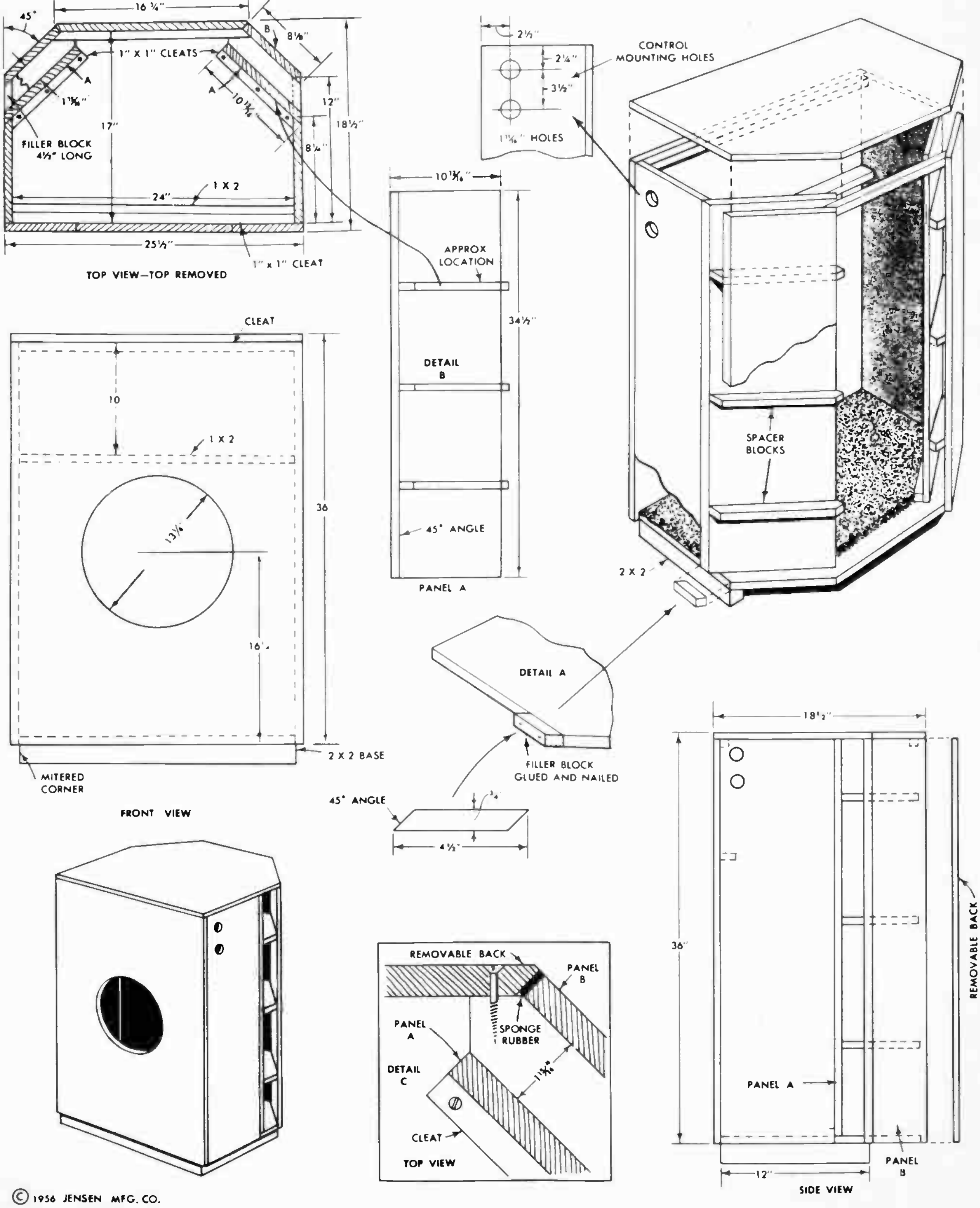
THE INLAND AUDIO SOCIETY IN THE SAN BERNADINO-RIVERSIDE AREAS, now in its third year of existance, is inviting audiophiles and music lovers in the San Bernardino, Riverside, Orange and Los Angeles counties to join us at our bi-monthly meetings and through our quarterly publication, in the pursuit for that elusive sonic truth. We provide a forum for auditioning equipment, sampling live music for educational purposes, guest presentations, discussing recordings, and the sharing of ideas, tips, theories, opinions, experience, and new product news relating to audio systems. Additionally we cater to the hobbyist who designs, builds and/or modifies electronic components and tranducing gear. Write for information concerning membership, dues and subscription. IEAS, PO Box 77, Bryn Mawr, CA 92318, (714) 793-9209.

Advertising Index

Vintage Designs

Jensen Bass Ultraflex-15

Courtesy of Charles Hickey, Charlotte, NC 28203



Pox Humana

Today's Episode: Truth

By Dick Fierce

Ok, gang, in honor of 1989, I have decided to do a special issue. Over the years, I have formulated several laws of acoustics, based on empirical evidence gathered in the business. So, without further adieu, I give you "Dick Fierce's Truths" (formerly "Suffolk Audio's Laws of Acoustics").

Laws of Acoustics

Truth #1
Any idiot can design a loudspeaker and, unfortunately, many do.

Truth #2:
You can say anything you want, who's to prove you wrong?

Truth #3:
The right amount of magnet is the right amount of magnet.

Truth #4:
The only transient of significance in the audio business is tranquility. It is also the briefest.

Truth #5:
Accuracy of reproduction is determined by how well a sound system models someone's warped set of preconceived notions.

Truth #6:
In audio, as elsewhere, foolproof systems prove the existence of fools.

Truth #7:
The size of a woofer is determined not by desired low-frequency response, but by perceived sexual dysfunction. After all, it's not the mass, it's the motion.

Truth #8:
Price buys not performance but paranoia.

Truth #9:
The most outspoken experts on concert hall sonic reality have seldom, if ever, been to a concert.

Truth #10:
The more money spent on an audiophile system, the less time spent on listening to music.

Truth #11:
In a minimum-phase system there is an inextricable link between frequency response, phase response and transient response, as they are all merely transforms of one another. This, combined with minimalization of open-loop errors in output amplifiers and correct compensation for nonlinear passive crossover network loading, can lead to a significant decrease in system resolution lost. However, this all means nothing when you listen to Pink Floyd.

Truth #12:
All small state-of-the-art audio manufacturers are really manifestations of Phineas T. Barnum.

Final Truth:
The audio business is no place for reasonable people to make a living.

morel®

high fidelity range

The MDM 85 is a mid range 75mm soft dome unit of extremely high standard, both from a design and technical viewpoint.

It incorporates the renowned Morel double magnet and Hexatech voice coil techniques, and results in a unit of above average sensitivity with extremely low distortion and high power handling capability.

With an output level of 96dB distortion in the area of 400-800Hz is slightly over 1% falling to 0.015% from 1Khz.

There are two different types available, one with a rear enclosure and one without (MDM 85NE). The type with the rear enclosure can be fitted into a cabinet as an integral unit.

The MDM 85NE without the rear enclosure can only be fitted into a system having a separate housing to enclose the unit. A volume of 0.7 litre is recommended for this housing, which is essential to prevent inter-reaction with the bass unit compressions and expansions. This housing must be filled full with damping material, such as fibreglass or rock wool.

The Thiele small parameters are given for both types under specifications. The contribution of this unit to a suitably designed system will be evident in the clarity and detail given in the 500-5000Hz region.

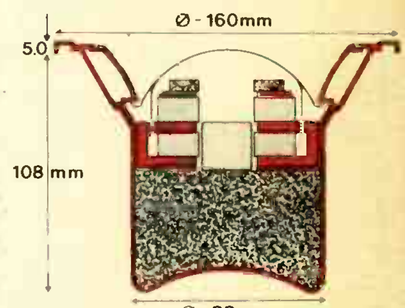
Specification

MDM 85 (with enclosure)

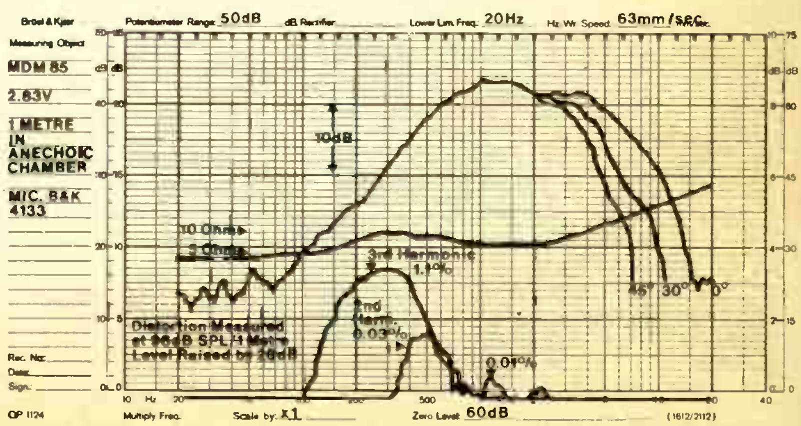
Overall Dimensions	Ø - 160mm x 113mm
Nominal Power Handling Din	300W
Transient Power 10ms	1500W
Voice Coil Diameter	75mm (3") Hexatech Aluminium
Voice Coil Former	Aluminium
Frequency Response	300-5000 Hz
Resonant Frequency	250 Hz
Sensitivity	92 dB (1W/1M)
Nominal Impedance	8 ohms
Harmonic Distortion for 96 dB SPL	<1%
Intermodulation Distortion for 96 dB SPL	<0.25%
Voice Coil Inductance @ 1 Khz	0.2mh
Air Gap Width	1.05mm
Air Gap Height	3.0mm
Voice Coil Height	6.0mm
Flux Density	1.0T
Force Factor (BXL)	4.6 W8/M
Rdc	5.2 ohms
Rmec	37.90
Qms	0.29
Qes	2.66
Q/T	0.20
Vas	0.33 litre
Moving Mass including Air Load	7.0 grams
Effective Dome Area	63.50 cm²
Dome Material	Chemically Treated Fabric
Nett Weight	1.25 kg

Variations to specification for MDM 85NE (without enclosure)

Overall Dimensions	Ø - 160mm x 60mm
Frequency Response	250-5000 Hz
Resonant Frequency	170 Hz
Rmec	39.33
Qms	0.19
Qes	1.81
Q/T	0.17
Vas	0.7 litre
Nett Weight	1.05 kg



Specifications given are as after 24 hours of running.



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

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Fast Reply #FD14