FOUR/88

U.S. $6.00
CANADA $7.00

Speaker Builder

THE LOUDSPEAKER JOURNAL

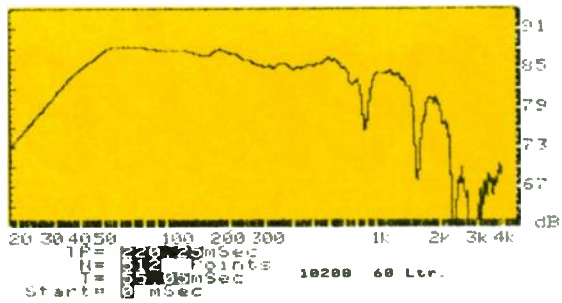
WOOFERS WOOFERS WOOFERS

MADISOUND 10208—10" Polypropylene Woofer 8Ω

Fs	22 Hz +/-2Hz
Mmd	50 Grams
Cms	1.06 mm/Newton
Vas	185 Litres
Rscc	5.5 Ohms
Z min	6.2 Ohms
Z max	42 Ohms
vcL	.7 mh
Qms	2.9
Qes	.56
Qts	.47
Xmax	7 mm pk
VD	248 cm^3
SD	.035 m2
Surround:	Foam
Voice Coil:	2" Kapton
Magnet:	30 Oz. Ceramic
Cone:	Black Polypropylene
Power handling:	100 Watts
Frequency response:	30-2K Hz
Efficiency:	87.5 db 1W/1M

Uses: home or autosound subwoofer

PRICE: $40.00



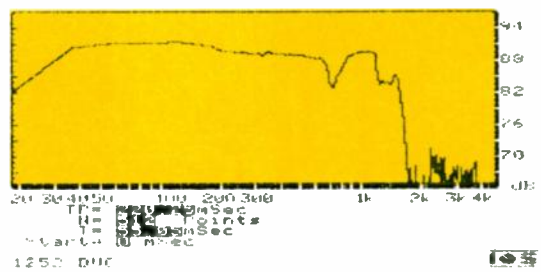
MADISOUND 10208 SEALED BOX ALIGNMENTS					
BOX VOLUME VB LITERS		35	56	70	85
BASS 1/2 POWER: F3	42	38	37	36	35
PEAK AT Res. R db	+1.5	+.5	+.2	0	-.2
QTC	1.05	.9	.82	.77	74
FILLING IN BOX	N	N	N	N	N
QL	5	5	5	5	5

MADISOUND 1252DVC 12" Polypropylene Woofer 8Ω/8Ω

Fs	19 Hz +/- 2 Hz
Mmd	98 Grams
Cms	.75 X 10^{-6} Cm/D
Vas	318 Liters
Rscc	5.7 ohms
Z min	7.0 ohms
Z max	69 ohms
vcL	1.7 mh
Qms	4.4
Qes	.39
Qts	.36
Xmax	6 mm pk
VD	300 cm^3
Surround:	Foam
Magnet:	30 OZ. Ceramic
Voice Coil	1.5" Kapton
Power handling:	100 Watts 50/50
Frequency response:	20-1.8K Hz
Efficiency:	90 db 1W/1M

USES: home or autosound subwoofer

Price $34.00



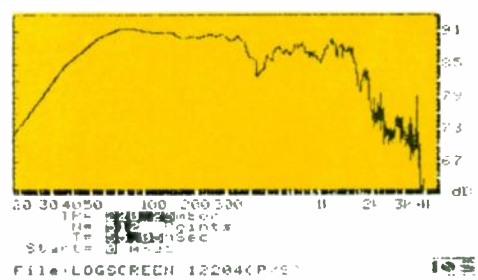
MADISOUND 1252 DVC BASS REFLEX ALIGNMENTS					
BOXVOLUME:VB	30 LITERS SEALED	56 LITERS SEALED	70 LITERS SEALED	120 LITERS VENTED	170 LITERS VENTED
BASS 1/2 PWR: F3	48.7 Hz	43.8 Hz	40.8 Hz	28.7 Hz	25.5 HZ
FILLING IN BOX	Y	Y	N		
QTC	.92	.73	.78		
PEAK AT RES: R db	+.8	.0	-.3	+1.2	+.3
BOX-VENT RES FREQ: FB				26 Hz	23 Hz
PORT: DIAMETER Inches				3.0	3.0
LENGTH Inches				4.5	3.8

MADISOUND 12204 DVC 12" Polypropylene Woofer 4Ω/4Ω

Fs	22 Hz +/-2 Hz
Mmd	106 Grams
Cms	.48 X 10^{-6} CM/D
Vas	206 Liters
Rscc	2.9 Ohms
Z min	3.36 Ohms
Z max	24.9 Ohms
vcL	.75 mh
Qms	2.116
Qes	.36
Qts	.316
Surround:	foam
Magnet:	40 oz. ceramic
Voice Coil	2" Kapton
Power handling:	200 Watts 100/100
Frequency Response	25-1.5K Hz
Efficiency:	90 db 1W/1M

Uses: Home or Autosound subwoofer

Price: $42.00

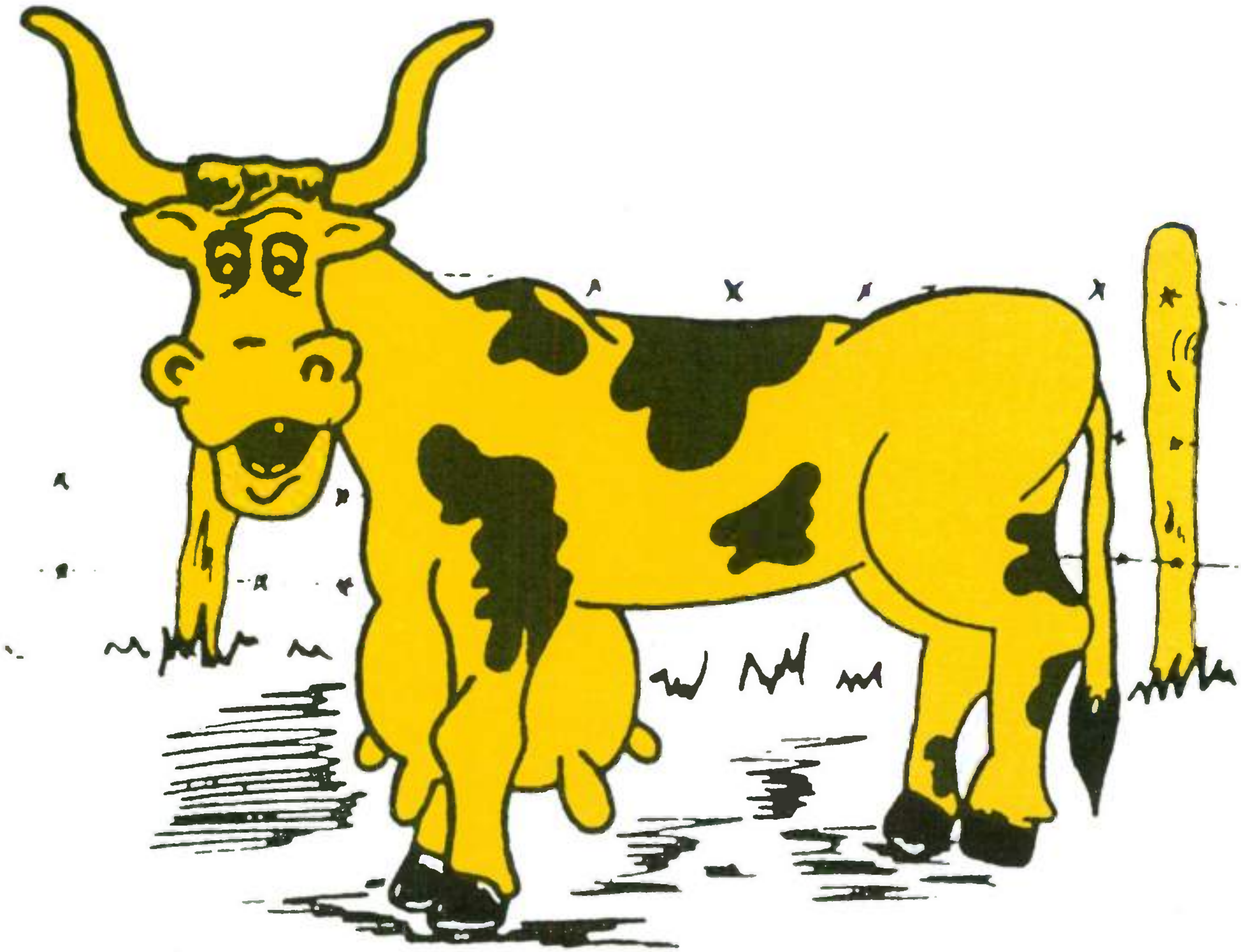


MADISOUND 12204 SUGGESTED ALIGNMENTS					
BOX VOLUME: VB	28 LITERS SEALED	42 LITERS SEALED	42 LITERS VENTED	70 LITERS VENTED	92 LITERS VENTED
BASS 1/2 POWER.F3	55 Hz	53 Hz	43 Hz	36 Hz	31 Hzz
FILLING IN BOX	N	N			
QTC	.83	.71			
PEAK AT RES: R db	+.3	-.5	+2.1	+.2	0
QL			5	5	5
BOX-VENT FREQ: FB			36 Hz	31 Hz	28 Hz
PORT: Diameter: Inches			2.5	3	3
Length: Inches			5.0	5.8	5.3

Ordering Information: All speaker orders will be shipped promptly, if possible by UPS. COD requires a 25% prepayment, and personal checks must clear before shipment. Adding 10% for shipping charges facilitates shipping procedure (Residents of Alaska, Canada and Hawaii, and those who require Blue Lable air service, please add 25%).

MADISOUND SPEAKER COMPONENTS
8608 UNIVERSITY GREEN
BOX 4283
MADISON, WISCONSIN 53711
PHONE (608) 831-3433

Fast Reply #HC20



Good News

The **ALTEC LANSING** acoustic suspension Model 55 mini speakers are now available in a matte white finish, in addition to the original black version. The mini speakers are ideal for use outdoors—on patios and swimming pool areas, and indoors—in kitchens, bathrooms, and as bookshelf or wall-mounted speaker systems.

The small size (7″W x 9⅝″H x 5⁵⁄₁₆″D) and keyhole receptables on the back enables them to be hung on a wall, be swivel-mounted using an optional bracket, or for flush mount installation in wallboard.

Each loudspeaker features a 4″ woofer cone constructed of carbon fiber cloth reinforced with epoxy and a 20mm polyimide dome tweeter. Strontium magnets are used for the drivers.

Specifications: frequency response per speaker is 85Hz-20kHz, ±3dB, power handling is 40W nominal, 80W maximum, impedance is 4Ω.

The Model 55 retails for $250 per pair, in white or black. The optional swivel mount bracket has a suggested retail price of $39.95.

For additonal information, contact Altec Lansing Consumer Products, Milford, PA 18337.

Fast Reply #HC328

ELECTRO-VOICE introduces a compression driver which is the first production use of neodymium in a loudspeaker product. EV, who first used neodymium in their N/DYM series of microphones, now uses the high-magnetic-strength material to down-scale the magnet structure and overall driver size. The N/DYM 1 weighs 7½ pounds and has a 5½-inch diameter. The comparable, non-neodymium model, the DH1A, weighs about 28 pounds and has an 8½-inch diameter. In addition, EV increased the flux density in the gap to 2.25 tesla.

The smaller, lighter N/DYM drivers, used in overhead cabinets, for example, 50 or 100 cabinets in a concert system, is significant in terms of overall weight, safety and transportation. The smaller diameter also allows greater flexibility in horn designs.

Two versions of the driver are available, the 8Ω N/DYM1 and the 16Ω N/DYM1-16, each priced at $750.

Contact Electro-Voice, Inc., 600 Cecil St., Buchanan, MI 49107, (616) 695-6831.

Fast Reply #HC43

RAPID SYSTEMS announces the R720 digital I/O and counter card for the PC, XT, AT or compatible computer. This card offers 32 digital input/output channels, and a three-channel programmable counter/timer. All of the I/Os are TTL compatible, and buffered with 74LS244s to increase output driving capacity and reduce input loading current needs beyond standard TTL. An Intel 8253 programmable timer provides three independent 16-bit presettable down counters as timing sources. Breadboard area on the card allows further flexibility for user development. Sample programs come with the user manual.

In stock now for $495, from Rapid Systems, 433 N. 34th St., Seattle, WA 98103, (206) 547-8311.

Fast Reply #HC948

ULTIMATE SOUND offers component loudspeakers for car stereo, including 20 separate polycarbon models of woofers and midranges, along with eight tweeter models.

The top-of-the-line UWP-1540 is a 15-inch woofer with 40-ounce magnet and a Kapton voice coil. The woofer is capable of 97dB SPL (1W/1M), and 200W/RMS maximum power handling. Suggested retail price for the UWP-1540 is $89.95.

In addition to 12-, 10-, 8-, 6x9-, and 6½-inch models, three midranges are available. The UWP-8020A is an 8-inch polycarbon woofer with 100W maximum handling, and a sensitivity of 93dB SPL (1W/1M); suggested retail price is $29.95.

The UT-996, a 4-inch tweeter with a 1-inch Titanium dome, comes packaged in pairs and has a suggested retail price of $64.95.

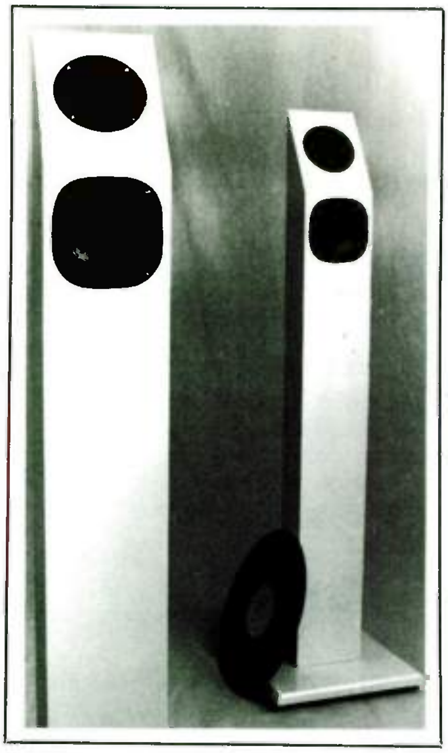
The UT-743 Titanium tweeter is 2⅞ inches round and has a ¾-inch Titanium voice coil with ferrofluid; suggested retail price is $9.95 each.

Additional tweeters include ribbon, polycarbon dome, textile dome and honeycomb models, and two piezo tweeters specificallly designed for high power handling capability.

Also, Ultimate Sound is changing the name of its "Ultimate Turbo" series of loudspeakers to "ULTIMATE PLUS." They are specifically designed for placement in particular types of vehicles although they can be used outside the vehicle.

For additional information and specifications, contact Ultimate Sound, 19330 E. San Jose Ave., City of Industry, CA 91748, (714) 594-2604.

Fast Reply #HC25



The newly formed **DELAC** (Delaware Acoustics) announces the S10 loudspeaker system, designed by Ralph Gonzalez. The integral sand-filled base eliminates the need for speaker stands, and a bass correction circuit allows two accurate 4½" drivers (one is rear-mounted) to provide a well-balanced low frequency response. The crossover to a ¾" tweeter was designed with LMP.

Other features include: biwirable/biampable design; very narrow (5"W x 3"D x 40"H) low-coloration enclosure with spikes; wide assortment of colors, genuine veneer finishes and reticulated foam grille; impedance, 4Ω; recommended power, 15-100W.

Availability is factory-direct with a free return period, $499/pair; $599/veneered.

For further information, send SASE to DELAC, PO Box 54, Newark, DE 19711.

AUDIOFEST '88—a major, complete audio show will be held September 24th and 25th, in Cherry Hill, NJ.

For the audiophile consumer: exhibits, demos, modifying, dealers, classics, up-and-coming manufacturers and big names, esoterica and accessories. The indoors show at the Garden State Park Racetrack Pavilion, just off the New Jersey Turnpike, will include over 225 exhibitors.

Advance admission: $9.50 day, $14 both days ($11.50, $16 at the door); tables: $37.50 per day; booths: $295 entire show.

Send your reservations to Audiomart Events Inc., PO Box 195, Crewe, VA 23930, (804) 645-8816 or 645-7838; FAX 645-8324. MasterCard and Visa welcome.

SPEAKER BUILDER

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About This Issue

Milestone is the only word that will do for the lead article in this issue. Contributing Editor **Joe D'Appolito**'s newly designed satellite system, whose first incarnation appeared in these pages four years ago (4/84), is now not only a splendid construction project, but will become a commercially available product as well. With **Jim Bock** as a partner, Swan Speaker Systems will be placing limited quantities of systems in selected showrooms across the country.

When **Art Newcomb** found his bass frequencies lacking, he turned to electronic means to remedy the situation with supplemental amplification. This compact electronic project (p. 22) may be a good way to begin exploring electronic construction for those of you whose soldering has consisted so far of assembling passive crossovers.

John Cockroft continues to push the transmission line thesis closer and closer to the ultimate minimum. The Unline design (p. 28) must be pretty close, but with John's record I wouldn't bet more than a dime that this is the end of the line(s).

Fernando Viesca is joined by **Marco Perez** to offer a simple program to do active filter design on personal computers (p. 34). Watch for much more on this topic upcoming from Editors **Galo** and **Koonce**.

If you have yearned for a ribbon tweeter but can't afford one of the beauties in the shops, **Richard Painter** has acted on **Mike Lampton**'s and **Henry Primbsch**'s neat tutorial article in *SB*'s 3/84 issue to produce a very respectable driver at a modest price (p. 38).

Vance Dickason reviews a major system design software package (p. 43) and **John Cockroft** a new stuffing material that may be better than wool (p. 48). And don't miss U.K. reader **A. M. Smith**'s pyramidal enclosure on page 50.

Congratulations to **Peter E. Sutheim**, whose two articles on Richard Heyser (*SB* 3 and 4/87) are mentioned in "Abstracts of Interest" in the *Journal* of the Audio Engineering Society, Vol 36, No. 6, June 1988, p. 521.

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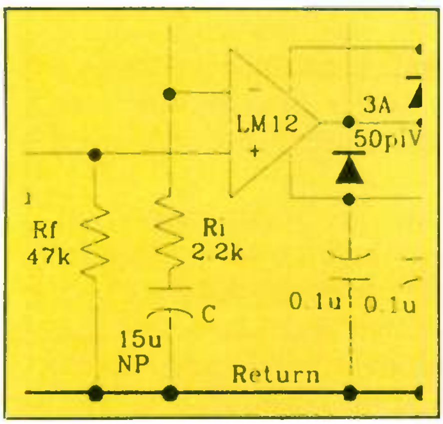

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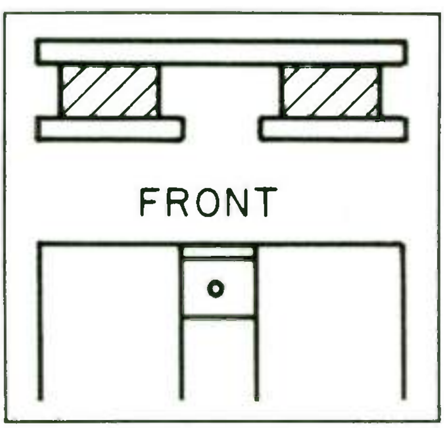

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Editorial

Rx For An Industry

Even a casual look at the US driver manufacturing landscape will reveal that some of the companies have apparently not been getting regular checkups. During the last six months, three companies, Becker, Oaktron and Precision have either ceased operation (Becker) or filed Chapter 11 petitions for protection against creditors.

In no instance is this due to a lack of orders, either from original equipment manufacturers or from the growing direct mail and retail trade in drivers. These failures, temporary or not for two of them, will have a serious effect on the US driver manufacturing community. Lead times in the industry are already overly long. This will doubtless make them even longer and may send OEMs overseas for answers to the product shortage.

The trouble can be traced to at least two sources: Undercapitalization and overage tooling.

One answer to the undercapitalization problem is already in place. In the "takeover atmosphere" presently prevailing, it may not seem a "good thing" to suggest some mergers, but in the world market we all now live in, some consolidation and financial muscle is obviously called for in the audio industry and especially in the driver manufacturing portion of it.

The financial condition of this country over the last seven years has not been favorable to industrial retooling. The steel industry is a prime example of this problem. And the problem besets anyone using the kind of machines used to make loudspeaker drivers. A large proportion of those companies who manufacture systems are, more and more, going overseas for their drivers. They report that they have not found the money to do the basic research and development necessary to retool for new materials and techniques. They then take the easy way out; let the overseas firms in Europe and the Pacific rim do the retooling and order drivers to spec from them.

I think it is more than a problem of money, however. It is really a problem of planning and strategy. Unless a company plans for steady, consistent R&D built into the overall growth of company, then profit taking and competitive pressures will spell trouble for a it amazingly quickly.

So what is the answer? Three consolidations have occurred in the industry in the same time period I am discussing. Harman-International has acquired Audax France and its subsidiaries. Here, a strong conglomerate has enough muscle to buy a company which, although it has had financial struggles recently, is very innovative in its research and product development.

Rockford Corporation has bought Carbonneau, a Michigan driver manufacturer and Hafler, Inc. who make components, and owned Acoustat.

These look to me like smart moves and should be good for the health of these expanded organizations and the companies that have been acquired. Both Harman and Rockford know something about the audio business and should be able to understand what to do with the companies they now own. We have had other acquisitions which have proved disastrous for the consumer, as in the sale of Dynaco to Tyco and the takeover of Advent by Peter Sprague.

In many ways, demand for good drivers has never been higher. And never have those using drivers in new designs been quite so clear about exactly what they want in the way of performance. In some ways the new and growing understanding of speaker technology has been a source of pressure on those making drivers in the US. The answer to relieve the pressure and the crisis is a new appreciation of the potential and a way to find capital and determination enough to put US companies producing drivers on the road to recovery.—E.T.D.

THE SWAN IV SPEAKER SYSTEM

BY JOSEPH A. D'APPOLITO and JAMES W. BOCK
Contributing Editor

Several years ago *Speaker Builder* presented a D'Appolito-designed, small, high-power satellite loudspeaker using a symmetrical dual mid-bass driver topology.[1,2] Judging by the correspondence this article generated, these speakers are highly successful and widely copied.

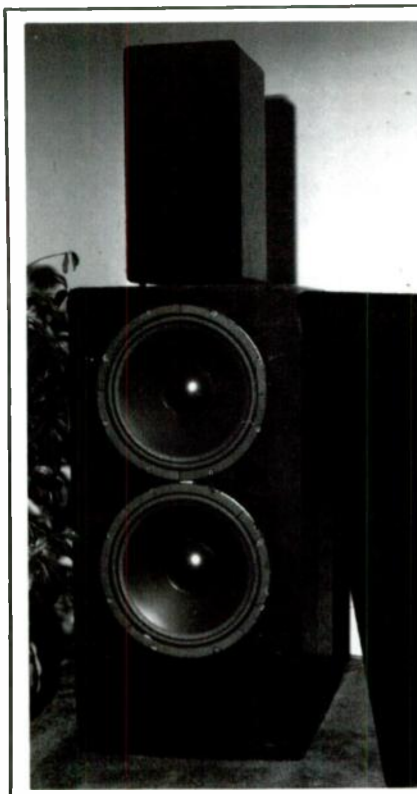
We have now revised and refined this concept, to evolve the SWAN IV™ speaker system. In addition to significant evolutionary improvements in the satellites, we have developed an entirely new bass speaker which matches the sonic quality established by the satellites. We have given special attention to enclosure design and finish so the resulting system is a visual pleasure as *Speaker Builder*'s 4/87 cover photo attests.[3]

Development Background

DESIGN PHILOSOPHY. Our design goal for the SWAN IV was nothing short of a state-of-the-art dynamic loudspeaker system for home use. The qualifier "home use" did place some limits on us. Generally, we believed listeners should be comparing the SWAN IV favorably against systems selling for $3,500 to $5,000.

Our first goal for the SWAN IV was to develop a loudspeaker system to reproduce the full dynamic and frequency ranges of digital recordings. Of course, this means the capability of playing loudly, and the final system will produce sound pressure levels (SPLs) of 110dB and more throughout the entire audio spectrum, with amplifiers of 200W/channel or more.

Beyond adequate volume and frequency response, however, imaging is all important. We wanted a system which would reproduce accurately the spatial and ambience information contained in coincident or minimally miked stereo and Ambisonic recordings. Mathematical imaging theory for these recording techniques is well developed and the consequent requirements for loudspeaker frequency and polar response are well known.



The listener is intended to be within the direct field of the loudspeakers. Near-term reflections (less than 5msecs) from nearby walls, floor, and ceiling must be minimized, as they cause frequency-response irregularities and confuse imaging. This in turn means polar response, especially at higher frequencies, must be focused forward and not be too broad, either horizontally or vertically. On the other hand, it should not be so narrow to impose a limited region or "sweet spot" as the only suitable listening environment. Frequency-response differences between left and right channels must be minimized to prevent frequency-dependent lateral image shifting. Frequency-dependent shifts in polar response must also be stabilized to prevent both vertical and horizontal image wander. We will explain how these general considerations influenced the SWAN IV system design.

DESIGN APPROACH. We designed the SWAN IV along two lines; refining the existing satellite concept, and developing a new complementary bass speaker. The imaging quality of the satellites depends in large part upon the use of two small mid-bass drivers symmetrically placed above and below the vertical

ABOUT THE AUTHORS

Joseph A. D'Appolito, contributing editor and author of several papers dealing with loudspeaker acoustics, holds several degrees in the field of electrical and systems engineering. For many years he has worked with an analytical science firm in Massachusetts.

James W. Bock holds degrees in mechanical engineering and law, and lives on a small island off the coast of Maine where he practices patent law and builds elegant wooden boats, harpsichords, and speaker cabinetry in a well equipped shop.

For over thirty years the authors have designed and built countless high fidelity systems together. They have embarked upon a joint venture called Swan's Speaker Systems to build and market quality systems such as the "SWAN IV."

FOOTNOTES

1. D'Appolito, J.A., "A Geometric Approach to Eliminating Lobing Error in Multiway Loudspeakers," presented at the 74th Convention of the Audio Engineering Society, New York, NY, October 1983, Preprint #2000.
2. D'Appolito, J.A., "A High-Power Satellite Speaker," *SB* 4/84, p. 7.
3. "Craftsman's Corner," *SB* 4/87, p. 39.

PHOTO 1: The satellite crossover is mounted on a circuit board.

axis of the tweeter to create a coincident sound source. Digital reproduction demands potential SPLs of 110dB. Reasonable cone excursion limits with two small mid-bass drivers effectively dictate a bass crossover frequency above 150Hz. Larger satellite mid-bass drivers allow for a lower crossover frequency, but would suffer a loss of imaging quality.

The old assumption that no useful stereo information exists at low frequencies is based in part on the lack of that information in earlier recordings. That lack was either a deliberate choice to reduce vertical stylus motion in the case of LPs, or the inability of the recording medium or the hardware to reproduce low frequencies accurately. Today digital recordings provide excellent response and great channel separation almost down to DC.

Through our listening experience over the years, we are convinced that combined or mixed bass systems do not image well, particularly when crossed over above 100Hz. A surprising amount of stereo information two or three octaves above the crossover frequency can be heard. This results in a smearing of detail and loss of coherence with most mixed bass systems. Further, image quality is degraded by physical separation of the so called "subwoofer" and high-frequency drivers.

Crossovers which are "transient perfect" or sum-to-one are desireable at low frequencies to preserve critical satellite/bass speaker coherence. They have 6dB/octave slopes which implies considerable information overlap between mid-bass and bass drivers. For our system, we chose a 6dB/octave crossover at 200Hz. For these reasons, we decided against using mixed bass schemes or bass enclosures located remote in space and time from the upper frequency drivers.

Since most listening environments serve other purposes as well, speaker placement should accommodate aesthetic considerations. Direct-field listening requires you to aim drivers, but angling large cabinets may not be acceptable for decorative reasons. Relatively small satellite enclosures can be aimed independently of the larger bass cabinets, upon which they are intended to sit.

The final result is the SWAN IV, a two box, full-range system, consisting of the Symmetrical Satellite™ and the Symmetrical Bass™ speakers. The complete system has a frequency response of +2/−3dB from below 25Hz to 21kHz and can produce average SPLs of 110dB or more, over this same frequency range. The satellite modules are 16½ by 8 by 11 inches and the bass modules are 32 by 16½ by 16 inches, which is reasonable in size and attractive to comply with our "home use" requirement.

The Symmetrical Satellite Speakers

DESIGN CRITERIA. Specific satellite design requirements that support our design philosophy are:

- Smooth, extended, on-axis response (±30°);
- Smooth, controlled, off-axis response (30+°);
- Controlled vertical response (±15°);
- Stable (i.e. frequency independent) polar response;
- Satellite pair frequency response matching within ±1dB.

We meet these goals with the free-standing satellites at least three feet from any vertical reflecting surface, to minimize near-term reflections.

The first requirement of smooth on-axis response is accepted universally. Concerning the second requirement, however, many small systems actually have a horizontal polar response which is too broad, causing excessive reflections from nearby rear and side walls. Horizontal response should be uniform within ±30° of the axis, but also should fall off smoothly at larger angles and higher frequencies to reduce reflection, and to maximize the direct-field energy (see *Fig. 8*).

Similarly, vertical response should be uniform within ±15° of horizontal, and fall off smoothly at larger angles to reduce floor and ceiling reflections, while permitting listening seated or standing.

Frequency-response mismatches between channels are a major cause of stereo image instability. Thus, the channels should match very closely. Last, but very important, the polar response pattern should remain stable and not shift with frequency. Frequency dependent polar axis wander causes image wander in otherwise well matched speaker pairs.

The vertically symmetrical driver arrangement of our satellites geometrically stabilizes polar response regardless of interdriver phase differences or crossover characteristics. This subject area is described in more detail in *SB* 4/84. This geometry places the effective acoustic center of the mid-bass driver pair coincident with the tweeter. The effect is analogous to the psychoacoustic phenomena in which a monaural virtual image ap-

pears to be exactly between balanced stereo speakers. This symmetrical geometry also produces a stable vertical polar response pattern, independent of interdriver phase and of magnitude response differences. The Treble Coupler™ crossover, described later, assures proper phasing and minimum time delay between the tweeter and the mid-bass driver pair, thereby avoiding the usual rationale for staggered driver placement or tilted front panels.

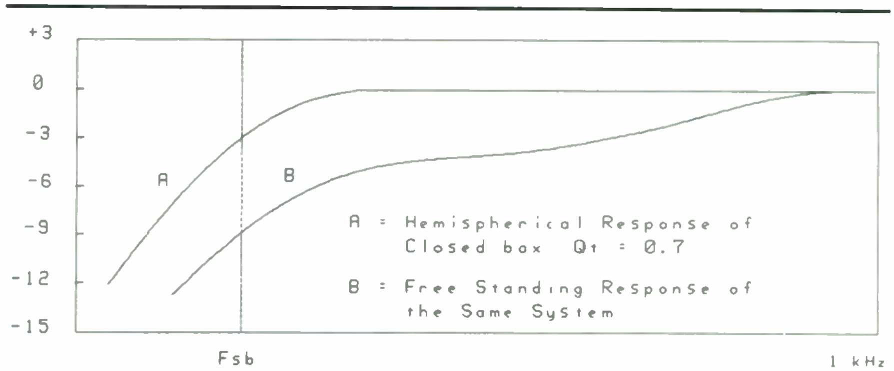


FIGURE 1: Diffraction loss with small boxes.

DRIVER SELECTION. Although similar to those described in *SB* 4/84, the present satellites are the result of many hundreds of hours of critical, comparative listening and measurement. Most of the musical burden is carried by the four 5¼″ mid-bass drivers. From the numerous choices available within our design parameters, we selected units made by Focal as being the least colored and those which consistently measure smoothly. The current design uses the Focal 5N412-DB for the mid-bass driver pairs. These are excellent units with cast frames, generous magnets, and dual voice coils. They are available in pairs and matched to within ±1dB. The second voice coil is used in our design to compensate for the falling on-axis low frequency response of small box speakers caused by diffraction loss.

Many "high end" systems with single mid-bass drivers use a sloping baffle or staggered driver geometry to compensate for interdriver time delay. In addition, a sloping front panel partially compensates for the poor on-axis response of many hard dome tweeters, and for frequency dependent polar axis tilt. Polar axis tilt refers to the swing of the vertical axis of symmetry of the polar response pattern above or below horizontal. The swing is caused by phase differences between non-coincident drivers in conventional geometries and causes on-axis frequency response variations throughout the crossover region.[4,5] Hard dome tweeters often display severe on-axis breakup modes above 15kHz which are narrow enough to be less apparent off-axis. With sloping baffle geometry, you listen to the drivers in an off-axis mode, thereby avoiding the thrust of the poor on-axis response.

Sloping panel systems, and single mid-bass driver systems in general, tend to have two acoustic disadvantages. First, they often have a very narrow "sweet spot" in the vertical direction. Usually, optimum listening is had only on a level with the tweeter. Listener movements of a few inches up or down produce noticeable changes in frequency balance.

Second because they are tilted upwardly, they bounce energy from the hard, reflective ceiling surface to produce undesirable early reflections. This second disadvantage may account for deliberate design of the typical narrow vertical response pattern which brings us back to the first disadvantage.

The Symmetrical Satellite avoids all of these sloping baffle or single mid-bass problems, but imposes requirements of its own on tweeter driver selection. the most important one is smooth on-axis response. As with the original design, the Dynaudio D-28 horn tweeter is used in the present version of our satellite. We have listened to and examined many other tweeters for this application, but we always come back to the D-28.

The combination of two 5N412-DBs and one D-28 measure extremely well together, and more importantly, provide the ultimate goal of critical listening quality without listener fatigue. We are aware of no other tweeter that has the combination of smooth, extended on-axis response, controlled off-axis response, power handling ability, and unit-to-unit consistency of the D-28. Consistency of driver parameters is crucial to the reproducibility of our design. Any project that must be hand-tuned to get the authors' results is not worth your time. Curiosity, not dissatisfaction, leads us to continue trying other drivers, but to date no other driver combination has achieved this goal as well.

Many readers have asked if the Dynaudio D-28AF flat-flange version can be used. Currently the answer is "no." The unique frequency- and phase-response of the D-28 horn driver lead to an elegantly simple satellite crossover design, as you will see. In addition, the D-28 horn assembly places the tweeter diaphragm behind the baffle and in closer alignment with the mid-bass drivers than the D-28AF.

DIFFRACTION LOSS. Let's examine diffraction loss in more detail to see how we use the second voice coil to compensate for it. We know a small closed-box loudspeaker with a Q_{ts} of 0.7 will have maximally flat frequency response down to its resonant frequency (F_{sb}), illustrated by *Fig. 1(a)*. This applies only if the loudspeaker has constant acoustic loading, which means it must radiate into a constant volume. Thiele/Small theory assumes a hemispherical space, approximated by a baffle with large dimensions, compared with the longest wavelength radiated. For small enclosures this usually is not the case. At higher frequencies, where wavelengths are short compared to the front baffle dimensions, acoustic energy radiates largely into the front hemisphere. At lower frequencies, where the smallest baffle dimension is comparable to one-half wavelength (about 850Hz for our satellites), some energy diffracts toward the rear, around the baffle. At still lower frequencies, radiation becomes omni-directional.

As the effective volume into which the speaker radiates increases, SPL falls off. At the lowest frequencies, on-axis output is down 6dB, as is illustrated by curve (b) of *Fig. 1*. Most small or narrow speaker enclosures display significant diffraction loss. The predominant effect of diffraction loss is a thinness of bass. Listeners often are unaware of this loss, for it is compensated somewhat by the rising reverberant field. The loss, however, is very apparent with human voices and we are convinced that it should be compensated.

4. Linkwitz, S.H., "Active Crossover Networks for Non-Coincident Drivers," *JAES*, Volume 24, January/February 1976, pp. 2- 8.
5. D'Appolito, J.A., "A Geometric Approach to Eliminating Lobing Error in Multiway Loudspeakers," presented at the 74th Convention of the Audio Engineering Society, New York, NY, October 1983, Preprint #2000.

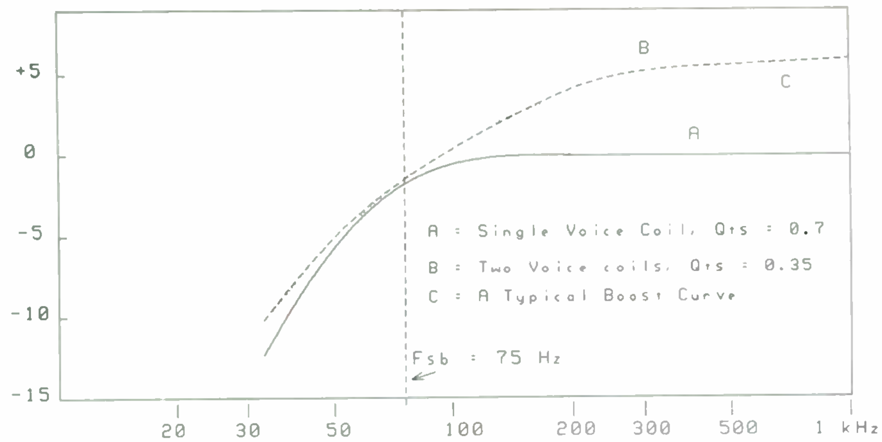


FIGURE 2: Hemispherical response of dual 5N412-DBs.

How does a dual voice-coil driver help? The mid-band sensitivity of a driver is directly proportional to B times L, where B is the magnetic flux density in the voice coil gap and L is the length of voice coil wire also in the gap. *Hemispherical* response for a single voice coil is shown by curve (a) of *Fig. 2.* When equal voltages are applied to both voice coils, the mid-band pressure response increases by 6dB because we have doubled L, and thus doubled the (B times L) product.

Not surprisingly, there's no free lunch here either. At resonance, driver response largely is controlled by electromagnetic damping which also is doubled, cutting Q_{es} in half. This reduces response an additional 6dB at resonance, relative to mid-band, and effectively returns us to the response of a single coil. The complete double coil response is illustrated in *Fig. 2(b).* The difference between the two curves is the boost available for diffraction loss compensation. Through proper design of the boost coil crossover, you can compensate for most of the diffraction loss. *Fig. 2(c)* shows a typical electrical boost profile.

CROSSOVER DESIGN. Our Treble Coupler passive crossover network is tailored to the specific electrical and acoustic characteristics of the selected drivers. It provides a transition between drivers, free of peaks or valleys and with a minimum of time delay. The dual voice coils of the mid-bass drivers are treated separately to compensate for the predictable low-frequency diffraction loss effects, discussed above.

In the original satellite design, we used an 18dB/octave acoustic crossover. The reasons for that choice are discussed at length in the *SB* 4/84 article and still are valid today. Two of these reasons are concerned with on-axis peaking of the mid-bass drivers and the natural mechanical roll-off of the D-28. A mild peak in the 5N412-DB response in the 4-5kHz region must be rolled off rapidly to prevent response ripple just above crossover. This dictates the use of either 18 or 24dB/octave filters.

On the tweeter side, the D-28 has a built-in response roll-off of 6dB/octave below 2kHz due to overdamping caused by ferrofluid in the voice coil gap. This first-order roll-off most easily is combined with a second-order high pass filter (12dB/octave) to obtain an overall acoustic roll-off of 18dB/octave.

We might consider a fourth-order Linkwitz-Riley crossover network, made up of two underdamped second-order pairs. However, to get an overall fourth-order response, the first-order response of the D-28 would first have to be cancelled, requiring a highly complex, fifth-order electrical filter for the tweeter crossover network.

An additional subtlety in the tweeter crossover design was not fully discussed in the original article. In a conventional 18dB/octave crossover application, the mid-bass and tweeter drivers electrically are 3dB down and 90° apart in-phase at the crossover frequency. Their acoustic outputs combine to produce flat frequency response over the crossover region. If the D-28 and the 5N412-DB pair are connected in a conventional 18dB/octave arrangement, however, their combined response is up 3dB at the crossover fre-

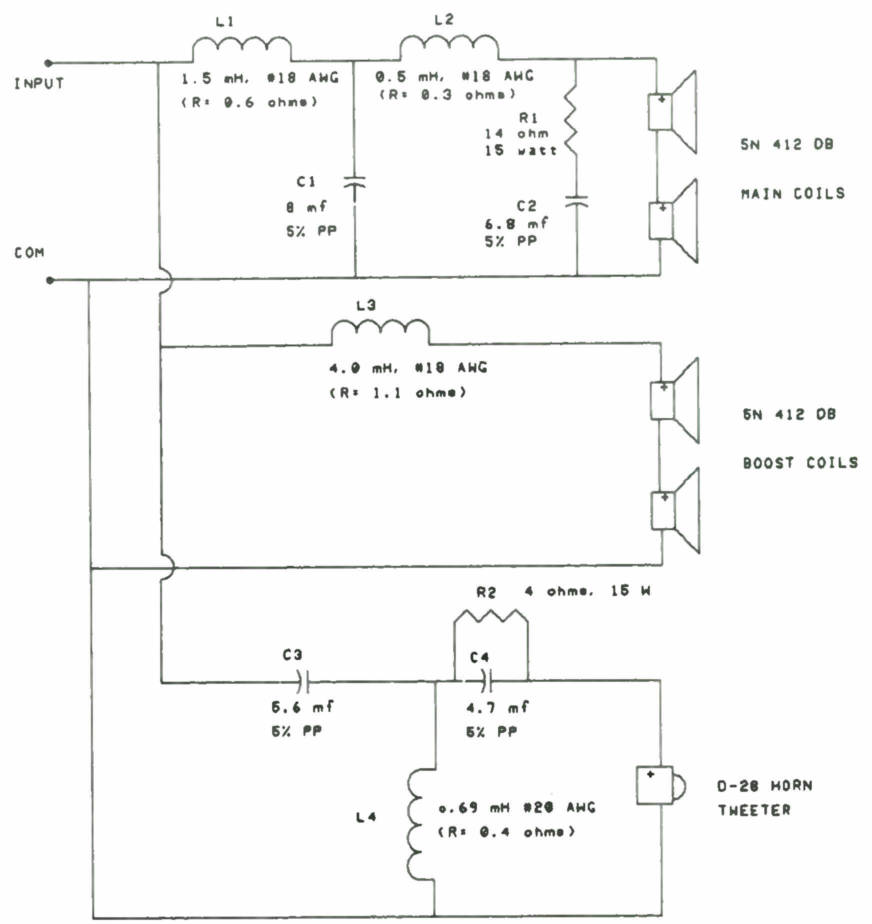


FIGURE 3: Treble Coupler schematic.

quency. We find the drivers actually are in-phase because the D-28 horn places a resistive load on the tweeter diaphragm. This resistive load produces an additional, almost constant, 90° tweeter phase lead, relative to the mass-controlled response of the mid-bass drivers.

If the crossover frequencies are spread by a factor of 1.4 and the crossover's Q is lowered slightly, the tweeter and mid-bass driver response curves will cross at a point where both drivers are down by 5.7dB and their acoustic responses then combines to produce an overall response which is flat to within ±0.3dB (*Fig. 7*).

Thus, the unique horn response of the D-28 can be used to produce an *in-phase* 18dB/octave crossover. The D-28AF flat flange version quite clearly will not work well with our crossover design.

The in-phase condition of the mid-bass and tweeter drivers has a very beneficial effect on our vertical polar response pattern, producing an off-axis null at 35°, above and below horizontal (see *Fig. 5* of Reference 2). This null, when combined with the natural high-frequency directivity of our drivers, limits the effective polar response to ±15° in the vertical plane. Thus, we have satisfied our vertical polar response goal.

Figure 3 is a schematic of the Treble Coupler satellite crossover. L1, L2, and C1 comprise an 18dB/octave low-pass filter. R1 and C2 are an impedance compensating Zobel, required to keep the network properly loaded in the presence of the rising voice coil impedance of the 5N412-DBs. It also controls the Q of the crossover/driver combination. We selected final values for flattest acoustical response. The two voice coils of each driver are identical. One is chosen as the main coil and one as the boost coil. Because of the transformer-like coupling between the dual voice coils of a single driver, a second Zobel across the series boost coil is not required.

Notice that the main coils of each driver are connected in series. The original design (*SB* 4/84) used parallel voice coils to maximize power sharing between unmatched drivers. Parallel connection of the dual voice coil Focals produced a questionably acceptable low minimum impedance of 3Ω. The Focal pairs now available match so well that series connection can be used safely with a concomitant 7Ω minimum

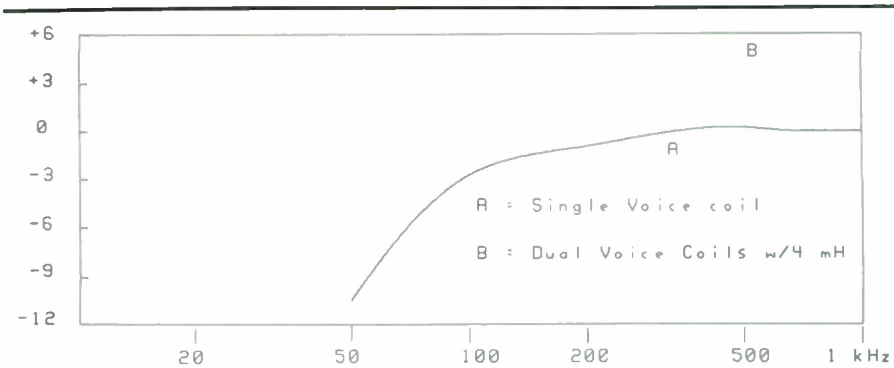


FIGURE 4: Measured hemispherical response of dual 5N412-DBs.

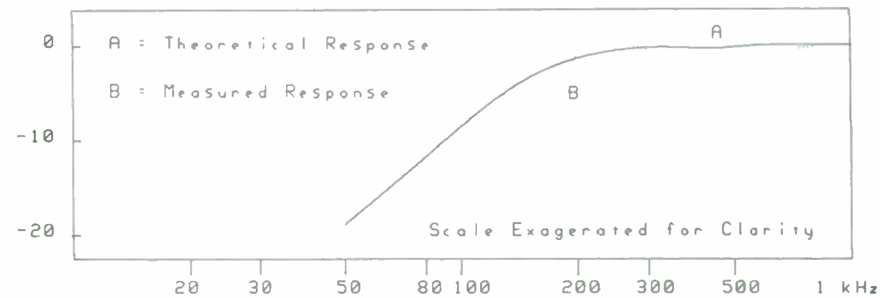


FIGURE 5: Theoretical vs. measured free-standing response.

impedance. The boost coils also are connected in series, fed from a 4mH coil to produce the low-frequency boost for diffraction loss compensation.

Figure 4 (*a* and *b*) show the measured hemispherical response of the Symmetrical Satellite mid-bass drivers, under single voice coil and active boost coil drive conditions. The single voice coil response agrees almost perfectly with Thiele/Small predictions. The 4mH first-order crossover produces a peak boost of 4.5dB at 320Hz. *Figure 5(a)* illustrates the theoretical free standing low-frequency response obtained by adding the curves of *Figs. 1(b)* and *2(c)*. Remember, diffraction loss occurs in the free standing condition. You should compare this against *Fig. 5(b)*, which depicts the measured response obtained by placing the satellite in the center of a large, mildly reverberant room with the tweeter axis 40″ off the floor. The theoretical curve is relatively flat to 200Hz and down 3dB at 150Hz. Departures of the actual response from the theoretical curve (which are caused by room reflections) are limited to frequencies below 200Hz and are no more than 2dB.

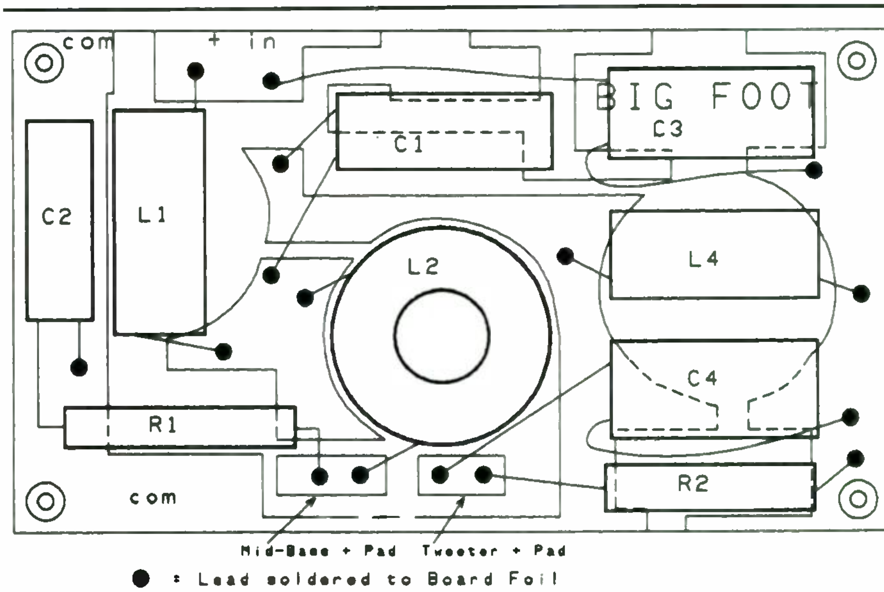


FIGURE 6: Treble Coupler circuit board layout.

The high-pass electrical crossover is second-order. L3 and C3 comprise the 12dB/octave high-pass filter. R3 attenuates the D-28 response by 4dB in the region below 8kHz to bring the sensitivity of the D-28 in line with that of the series-connected 5N412-DBs. Capacitor C4 shorts R3 progressively above 8kHz to lift the drooping high-end response of the D-28. Again, final values we selected give the flattest acoustical response.

TREBLE COUPLER. The Treble Coupler crossover and the 4mH coil are assembled on separate circuit boards, available from Madisound Speaker Components (see Sources at end of this article). The assembled satellite crossover is shown in *Photo 1*. Individual components are glued in place with "Goop" and leads are soldered directly to the PC board foil following the placement diagram of *Fig. 6*. L1 and L4 are mounted on edge to reduce mutual coupling between them and L2. A chord is cut across the sides of the coil forms for L1 and L4 to facilitate the on-edge placement.

Notice that wire size and approximate resistance are called out for each coil in *Fig. 3*. Tweeter and mid-bass driver sensitivities and mid-bass driver damping have been balanced carefully by proper selection of coil resistance. *Do not* substitute lower resistance, larger gauge coils in the hope of better performance. Lower resistance coils not only cost more, but degrade performance.

MEASUREMENTS. Individual driver responses and the overall satellite frequency response are shown in *Fig. 7* for the free-standing condition. We measured all frequency responses with swept one-third octave warble tones. We obtained these curves with the satellite at design height (tweeter 40" off the floor) in the center of a large room, and the microphone at 0.5m on the tweeter axis.

Notice the driver responses cross at roughly 2kHz, where they are down 5.7dB in accordance with our earlier discussion of the "in-phase" 18dB/octave crossover. Response is ±1.5dB from 200Hz to 18kHz with −3dB points of 150Hz and 21kHz. Although not shown, we measured interdriver time delay through the crossover region at less than 10μsecs, (typically 5μsecs), using shaped tone bursts. Satellite sensitivity is 90dB at 1m with 2.83V input.

Response at 15° and 30° off-axis is shown in *Fig. 8*. The responses essentially are identical with the on-axis response, but show gentle fall-off at high frequencies.

Figure 9 shows the satellite input impedance curve. System resonance is at 75Hz where the impedance rises to 51Ω; system Q is 0.5. The minimum impedance of 7Ω is reached at 250Hz and rises slowly thereafter to 10.5Ω at 1.2kHz, where it then begins to fall again as the tweeter cuts in.

ENCLOSURE DESIGN. Compared with the original *SB* 4/84 design, we reduced the enclosure volume to about 620 in.³, to suit the Focal 5N412-DB mid-bass drivers. We increased the front panel width slightly to 8", to reduce the frequency at which diffraction loss begins, and thereby to more closely match the capability of the boost coil to compensate for it. An acoustic foam septum divides the enclosure cavity to isolate the mid-bass drivers from each other at higher frequencies.

We have found high-frequency diffraction problems are aggravated by decorative grille frames and seemingly trivial anomalies in surfaces near the drivers. Consequently, we settled on a front panel having no interruptions. The tweeter and mid-bass drivers are rabbeted in to bring their outer rims flush with the baffle. We have tried various techniques to support the grille cloth, but each induces ripples in response. Our solution is a special foam grille. We made all measurements with our foam grille in place. We discuss

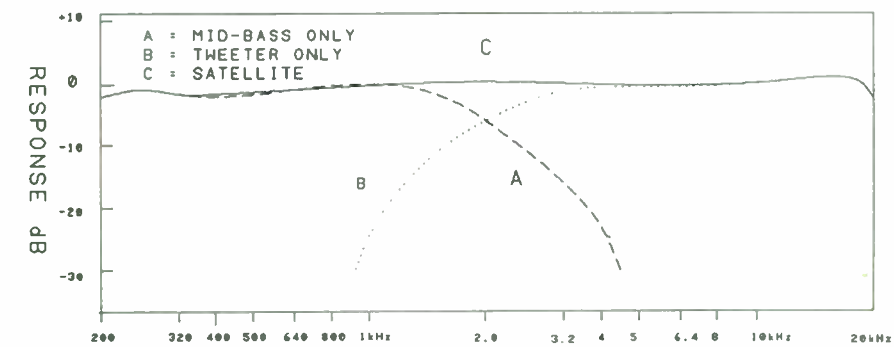


FIGURE 7: Satellite on-axis frequency response.

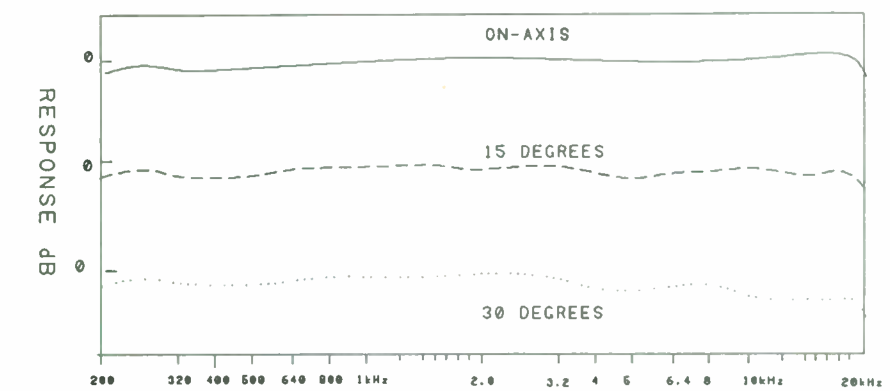


FIGURE 8: Off-axis responses.

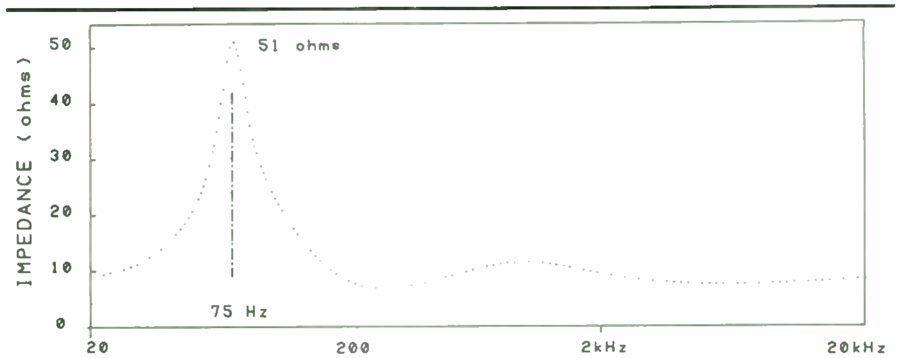


FIGURE 9: Satellite impedance curve.

grille covering options in the cabinet construction portion, in *Part II*.

The Symmetrical Bass Speakers

DESIGN CRITERIA. The bass module is not a subwoofer, but rather, the larger of two boxes which makes up a full-range system. Our specific design goals for the bass speaker are:

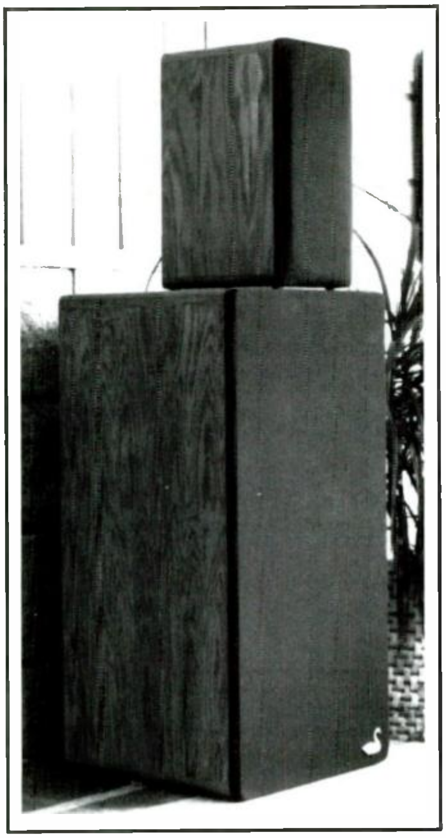
- Smooth frequency response from below 25Hz to 1kHz;
- A minimum power response of 110dB SPL above 25Hz;
- Exclusion of infrasonic signals below 20Hz;
- A maximum internal volume of 100 liters.

The extended frequency response of the bass drivers allows us to use slow slope crossovers between the satellite and the bass module. The SPL requirement will satisfy Telarc drum and organ fans while the limited physical size will help to satisfy home decorating aesthetics.

Our reasonable bass enclosure size makes it impractical, in view of our stated goals, to attempt reproduction of musical information much below 20Hz at high SPLs with the SWAN IV system. When operated without the infrasonic filter of the Pedal Coupler, the speaker cones pant visibly as LPs turn, and other non-musical artifacts may cause wild cone gyrations. That some CDs have significant recorded information below 20Hz and that proper reproduction of it requires high SPL is a subject well treated recently by Fielder and Benjamin.[6]

DRIVER SELECTION. We began our investigations with closed-box designs because of their purported superior low-frequency transient response (i.e., freedom from overhang or ringing), even though we knew it would be difficult to meet our low-frequency SPL goal. We tried a number of bass drivers in various enclosures, within our chosen physical dimension limits. None produced adequate deep bass (below 45Hz) and all lost cone control on certain recordings (particularly digital) at reasonably loud volume levels. All tended to thicken cellos and voices with unwanted groanings. We cast about for 12″ drivers with reasonably long throw under control of the voice coil, good magnetic circuit design, and cone materials unlikely to display strange vibratory modes. None seemed appropriate at any price.

A search for 15″ drivers proved equally fruitless. We decided to take a chance on 12″ drivers inexpensive enough to double up. We literally stuffed a pair into each of two previously constructed 90 liter sealed enclosures. The results were encouraging and the cost of four replacement drivers was half that of a pair of the expensive cast frame units. Still, our SPL requirements below 45Hz were not met and excessive cone motion occurred, causing occasional driver bottoming.



At this point we reassessed our bias against bass reflex systems. A few quick calculations (which in retrospect we should have made much earlier) showed that the SPL levels we sought at 25Hz would require a volume throw of 600 cm^3 even with bass reflex designs. Most good 15″ drivers have this much volume throw, but will not work well in 100 liter boxes. Most good 12″ drivers have a much smaller equivalent volume (V_{as}) than a 15″ of comparable resonant frequency, so a pair of 12″ drivers will work much better in a 100 liter box than a single 15″.

The final design uses a pair of Precision TA 305 F 12″ polypropylene cone drivers. These drivers have an F_{sa} of 25Hz and a Q_{ts} of 0.3, which make them ideal for a sixth-order Butterworth bass reflex alignment. A pair has a volume throw of 580cm^3 and smooth response to 1kHz.

Some of you may question using relatively inexpensive drivers in this application. The short answer is that they work very well. A great deal is written in the high-end press about the "speed" woofers require to meld properly with upper range units, but it makes no sense to talk about the "speed" of a driver which is fed by a signal limited to 200Hz or less. Woofers limited to this bandwidth may suffer when driven hard, but not because they have inadequate "speed" or poor high-frequency transient response which is a "small signal" property.

Their problems occur with large excursions because they have inadequate magnetic circuit or suspension designs and/or poor quality control of assembly tolerances, which affect "large signal" behavior. Under the large excursions required for high SPLs at low frequencies, drivers tend to become nonlinear, producing excessive distortion and dynamic compression. These effects easily are heard, but wrongly are labelled as transient response or "speed" deficiencies. The simplest and least expensive way to avoid this problem, is to reduce cone travel by doubling up drivers, thereby reducing cone travel for a given SPL by a factor of two. Since the onset of nonlinearity generally is very rapid, halving cone motion produces much more than a halving of distortion and essentially eliminates dynamic compression.

We have listened at great length to single and dual driver versions of our bass speaker. The difference in attack, punch, tightness, and so on, is dramatic and clearly shows two inexpensive, stamped-frame drivers are much better

TABLE I
PRECISION™ TA 305 PARAMETERS

F_{sa}	=	25Hz
Q_{ts}	=	0.3
V_{as}	=	180 liters
P_{avg}	=	200W

TABLE II
SIXTH-ORDER BASS-REFLEX SYSTEM PARAMETERS

System	V_b (liters)	F_b (Hz)	F_c (Hz)	G_c (dB)
Op. Box	145	25	25	6
Swan IV	100	22	25	9

6. Fielder, L.D. and E.M. Benjamin, "Subwoofer Performance for Accurate Reproduction of Music," presented at the 83rd Convention of the Audio Engineering Society, New York, NY, October 1987, Preprint #2537 (G-4).

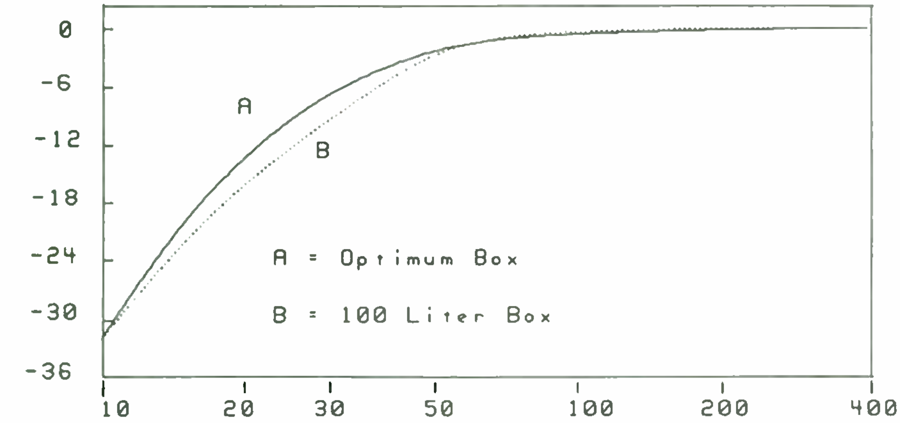


FIGURE 10: Computed optimum vs. 100-liter box.

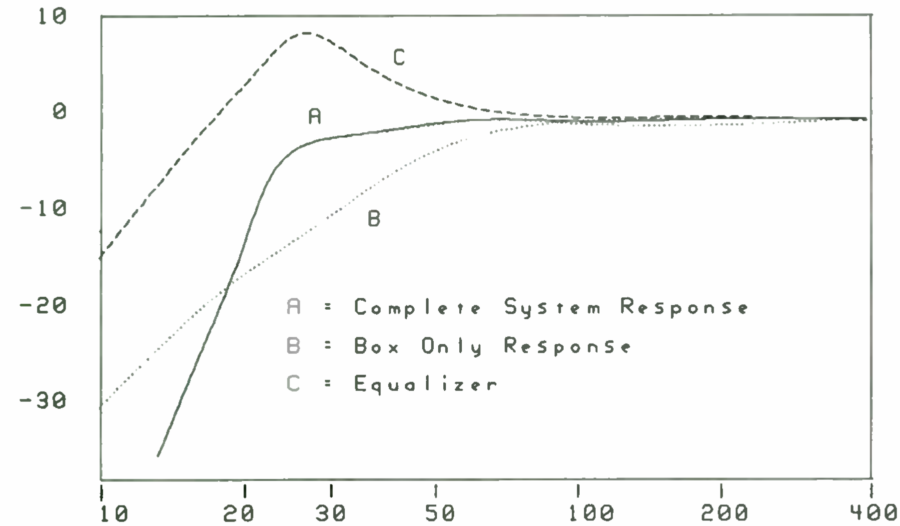


FIGURE 11: Bass speaker frequency response.

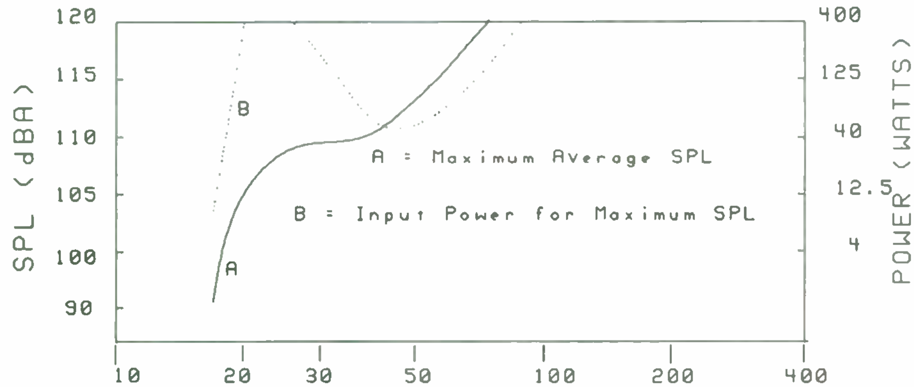


FIGURE 12: Power response, single bass module.

than one very expensive cast-frame unit. In one particular case, for the same distortion level, the inexpensive duals produced 12dB more SPL than the expensive single.

With the vertically symmetrical arrangement of the dual bass drivers, the apparent sound source appears half-way between the two drivers. We can distinguish this, with the satellites disconnected, because the gentle crossover slope at 200Hz provides enough higher frequency intelligence.

BASS DESIGN. Typical measured parameters for the TA 305 F are given in *Table 1. Table 2* lists box and equalizer parameters for a Class I sixth-order bass reflex system. Class I alignments produce the smallest box volume with the Precision TA 305 F pair.

Unfortunately, the optimum volume of 145 liters is 45% larger than we would like. Well, we're not slaves to alignment. Compare the computed response of the 145 liter and 100 liter boxes in *Fig. 10.* The response of the 100 liter box is down

PARTS LIST

Drivers: Four Precision TA 305 F 12″ bass drivers; Four Focal 5N 412 DB 5¼″ drivers (two matched pairs); Two Dynaudio D-28 horn tweeters.

Pedal Coupler: One Channel

Resistors

R2-7, 16, 17	24.9kΩ
R8	78.7kΩ or 75.0kΩ
R9, 10	10kΩ
R11	51.1kΩ or 53.6kΩ
R12, 18	100Ω
R14, 15	2.49kΩ

All resistors 1% metal film

Capacitors

C1	4.7μF polypropylene film*
C2, 5-9	0.1μF polypropylene film*
C3	47pF polystyrene
C4	0.033μF polypropylene film*
C10	22pF polystyrene

* Panasonic P-series

Op Amps

IC1	AD712
IC2	AD712 or TL072

Potentiometers (Dual Linear)

R1	100kΩ conductive film
R13	5kΩ conductive film

Pedal Coupler: Power Supply

T1	32VCT/0.075 A Stancor SE-332 or equivalent
B1	50V, 1A bridge
C1, 2	1,000μF, 25V DC electrolytic
C3, 4	10μF, 25V bead tantalum
IC3	78L15 100mA, +15V reg.
IC4	79L15 100mA, −15V reg.

Treble Coupler: One Channel

Inductors: Solen

L1	1.5mH, #18 AWG, R ≈ 0.6Ω
L2	0.5mH, #18 AWG, R ≈ 0.3Ω
L3	4.0mH, #18 AWG, R ≈ 1.1Ω
L4	0.68mH, #20 AWG, R ≈ 0.4Ω

Capacitors: 5% polypropylene

C1	8μF
C2	6.8μF
C3	5.6μF
C4	4.7μF

Resistors

R1	14Ω, 15W
R2	4Ω, 15W

Cabinet Materials

(1) 4 x 8: ¼″ veneer faced plywood
(1½) sheets 4 x 8: ¾″ medium density particle board
½ x 1½″ x 20′ hardwood to match veneer
4 x 4 x ⅛″ tempered hardboard for backs
¾ x ¾ x 10′ softwood for cleats
4′ x 2½″ ID schedule 40 plastic pipe (electrical conduit)
Epoxy materials (quart), screws, staples, silicone caulk, varnish, paint, grille materials, wire, terminals, etc.

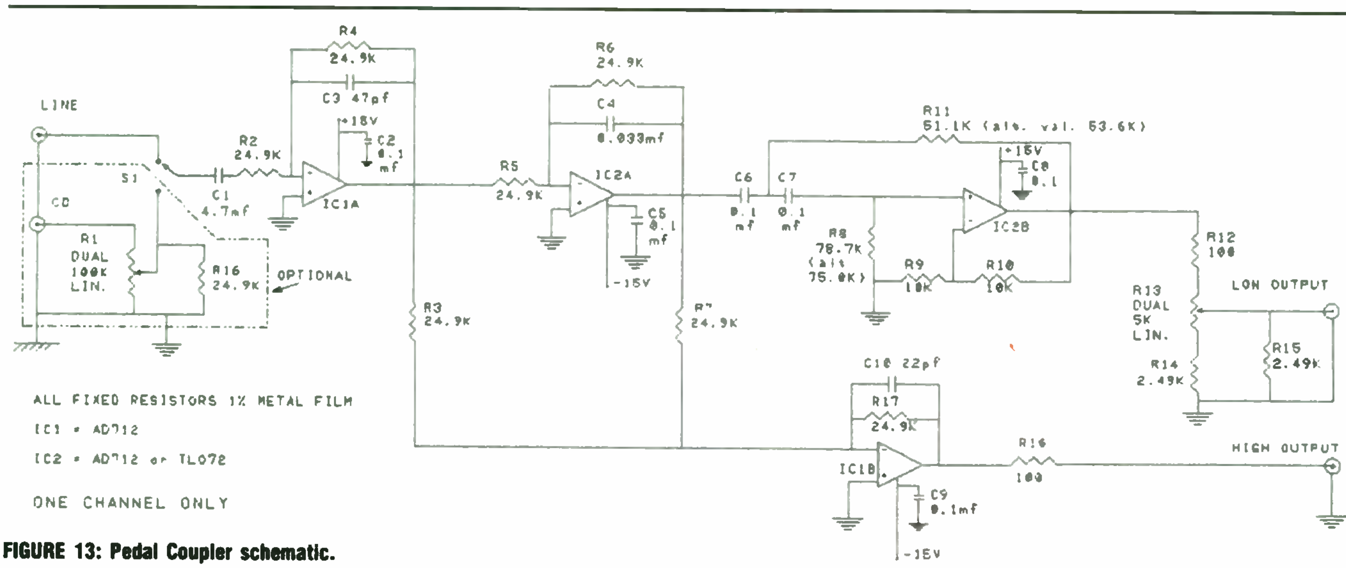


FIGURE 13: Pedal Coupler schematic.

3dB relative to the 145 liter box at 25Hz, which is the box resonance frequency (F_b). The 100 liter box also peaks slightly relative to the 145 liter box in the 60–100Hz range. We can cure the peaking by lowering the box frequency F_b to 22Hz. Increasing equalizer gain by 3dB, as shown in *Table 2*, compensates for the frequency response loss.

The only drawback associated with the 100 liter box is a 2-3dB loss in maximum SPL, in the 25–32Hz range. The complete 100 liter bass-reflex design is given in the second row of *Table 2*. The required second-order electronic filter is included in the Pedal Coupler active crossover which we discuss in the next section.

The frequency response of a single bass module and its electronic filter is shown in *Fig. 11*. Power response, in *Fig. 12*, shows a maximum average SPL capability of at least 110dB, all the way down to 25Hz when working into a hemispherical space. Most home listening rooms actually will provide additional boost, by appearing to the speaker more like a quarter space than a hemisphere.

The input power required at each frequency to reach the plotted SPL is also given in *Fig. 12*. Throughout most of its operating range, the bass speaker output is limited by the voice coil thermal power rating of 400W (200W per driver). The dip in the maximum allowable input power between 32 and 64Hz, however, indicates in that frequency range the output is throw-limited rather than power-limited. The system also is throw-limited below the box resonance of 22Hz where the cone unloads rapidly. Cone motion is limited in this region by the equalizer infrasonic electrical roll off of 12dB/octave. Two bass modules will produce 3-6dB more output; you can reach 110dB with an amplifier capable of delivering 200W per channel into 4Ω.

ACTIVE CROSSOVER. We use the Pedal Coupler to biamplify the satellite and bass speakers and provide the necessary equalizer filter for the SWAN IV system. A first-order crossover, it uses a subtractive circuit topology to guarantee "transient-perfect" melding of satellite and bass speaker acoustic responses without critical component matching. The crossover also includes the second-order equalizer required to complete the sixth-order bass reflex alignment chosen for the bass module. As shown in *Fig. 12*, this equalizer provides 9dB boost at 25Hz. Frequencies below this point are rolled off at 12dB/octave to prevent excessive cone motion at infrasonic frequencies.

We increased the crossover frequency between satellite and bass modules from 100Hz in the original design, to 200Hz, to reduce mid-bass driver cone excursion demands and increase overall system dynamic range by 6dB. The system can produce peak SPLs of 120dB at frequencies above 50Hz, if your amplifier can provide the power to drive it.

A schematic diagram for one channel of the Pedal Coupler is shown in *Fig. 13*. Most op amps are operated in the inverting mode to eliminate common-mode signals and produce lowest distortion. IC1A is an inverting unity gain buffer. Two inputs are provided; a line input, and an optional second input with level control for direct input from a CD player.

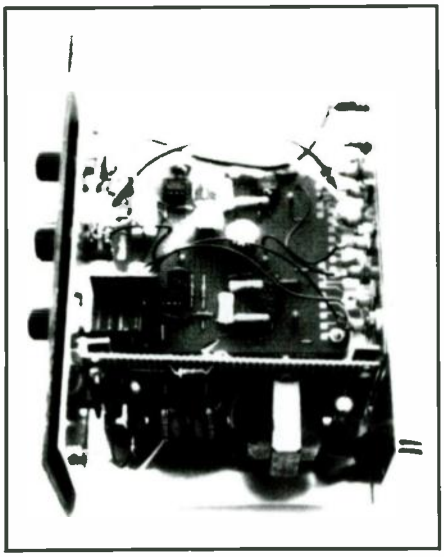


PHOTO 2: The Pedal Coupler active crossover.

IC2A is an inverting, first-order low-pass filter with a corner frequency of 200Hz, as determined by the time constant of R6 and C4. The output from this stage, which is in-phase with the input, feeds IC1B and IC2A. IC1B is an inverting summing amplifier, which produces the required high-pass output by adding the low-pass output of IC1A to the inverted input and inverting the result to produce the in-phase high-pass output. This is equivalent to subtracting the low-pass filter output from the input. Since the high-pass function is obtained by subtraction, it should be clear the high- and low-pass outputs must sum back to the original input signal.

IC2A forms a second-order high-pass filter with a corner frequency of 25Hz and a Q of 2.8, which produces 9dB of boost, shown in *Fig. 11*. IC2A has a mid-band gain of 2 (6dB) to permit adjustment of the bass module level relative to the satellite. R13 is a dual linear potentiometer which is padded by R14 and R15 to provide a total linear gain range of ±6dB relative to the satellite output level.

Similarly, R1 is a dual linear potentiometer padded by R2 and R16, to produce an audio taper volume control. We used padded linear controls because they track much better than audio taper pots. Because of the low gain levels throughout the circuit, exotic construction techniques are not needed. The whole circuit can be laid out on a Radio Shack #276-157 op amp circuit board.

The Pedal Coupler power supply, shown in *Fig. 14*, is quite conventional. It uses a 1A bridge, capacitive filter and TO-92 100mA regulators to supply ±15V DC to the op amps. You may use the power supply from Old Colony Sound (KE-5, $20) in place of everything ahead of the T0-92 regulators. A prototype of the Pedal Coupler is shown in *Photo 2*. For the interested reader, a completely packaged version of the Pedal Coupler (similar to that described, but without CD input) is available (see Sources).

This concludes Part 1 of the SWAN IV system. Continued in the next issue, Part 2 will show, in complete detail, cabinet design and construction, final assembly and grille considerations, and room placement and listening tests.

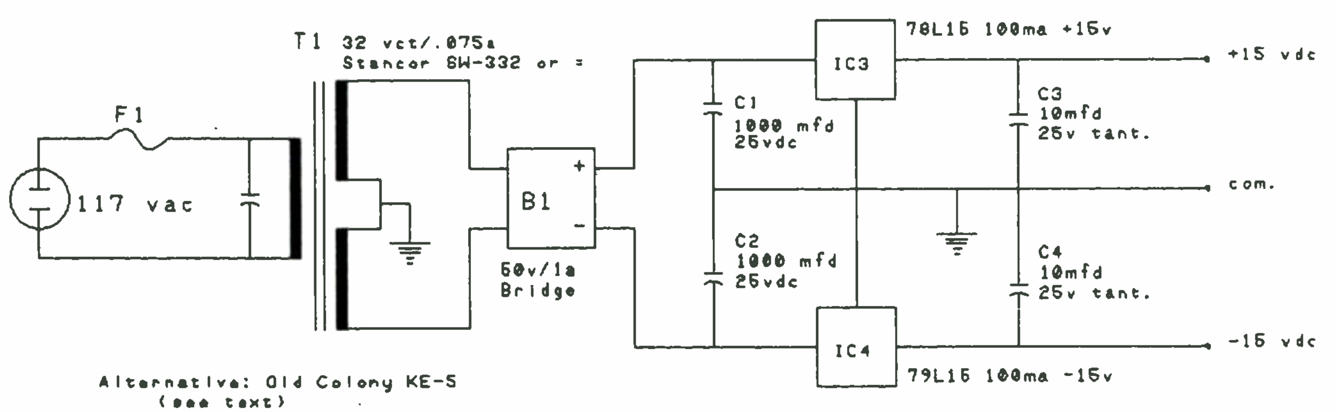


FIGURE 14: Pedal Coupler power supply.

AUTHORS' NOTE: The authors encourage *Speaker Builder* readers to use the information in this article to build their own systems. Toward that goal, the authors hereby extend a free license to readers, limited to non-commercial, personal use of the technology and knowhow. The authors have begun marketing finished loudspeaker systems, and licensing arrangements for kits and components using this proprietary information are contemplated. Consequently, use of this information for commercial purposes is not permitted.

SOURCES

Madisound Speaker Components
8608 University Green
Box 4283
Madison, WI 53711
(608) 831-3433
(kits of speaker drivers, the Treble Coupler passive crossover, and components for crossovers, terminals, wire, etc)

Swan's Speaker Systems
Box 356
Swan's Island, ME 04685
(207) 526-4343
(the Pedal Coupler electronic crossover and equalizer, $225 plus shipping, preformed foam grilles)

Old Colony Sound Lab
Box 243
Peterborough, NH 03458
(603) 924-6371
(Power supply, KE-5)

Gougeon Brothers
Box x908
Bay City, MI 48707
(517) 684-7286
(epoxy materials, supplies, and tools)

Radio Shack Stores
(Grille Cloth or Foam)

The Woodworkers' Store
21801 Industrial Boulevard
Rogers, MN 55374-9514
(612) 428-2199
(square-drive particle board screws)

Woodcraft
41 Atlantic Avenue
Box 4000
Woburn, MA 01888
(woodworking tools)

SUB-BASS POWER BOOSTING

AND OTHER APPLICATIONS USING THE LM12

BY ART NEWCOMB

With the availability of National's LM12 power operational amplifier, I was inspired to revisit a circuit I developed in 1981! I redesigned it to compensate for the somewhat disappointing bass performance of a large Altec speaker system I thought should do better. The circuit provides the additional low-frequency power boost of a third amplifier in the stereo system to augment the power output in the sub-bass range. Compensation of this type can be done with a graphic equalizer but boost slopes are usually limited to 6dB per octave or less, whereas speakers tend to roll off at 12dB or more. With some systems I find it the next best thing to a subwoofer.

The circuit of *Fig. 1* is designed to:

PHOTO 1: A completed bass booster.

PHOTO 2: LM12 circuit boards.

- Create a virtual ground for the stereo speaker pair with the third amplifier;
- Derive a composite low-frequency signal from the stereo amplifier;
- Use the inverted signal as a differential low-frequency drive via the third amplifier.

You could say, "It's done with mirrors," or an analog thereof. Here are the prerequisites:

- Your stereo amplifier must have a common return for both speakers, that is, it cannot be a bridged output (a frequent practice in automotive units).
- Your speakers must be capable of handling the additional power.

To accomplish this chicanery, the third amplifier must provide a suitable return for the stereo amplifier and have the drive to operate into very low imped-

1. Newcomb, A.L., "SubBass on a Budget" *Audio*, Vol. 66, No. 8, pp. 36-39, August 1982.

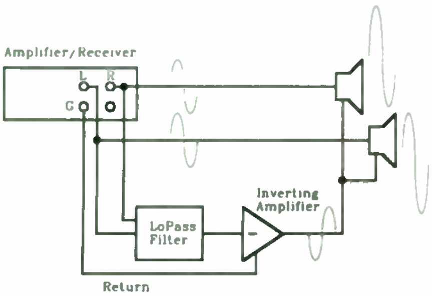


FIGURE 1: The sub-bass boost scheme.

PARTS LIST I

SUB-BASS BOOSTER

R1, 2	240*
R3	1k*
R4,	5k (linear pot)*
R5	3.3k
R6	50k
R7	5k**
C1, 2	50μ NP*
C3	0.05μ**
Cps(2)	0.1μ
C6, 7	2500μ*
D(2)	3A, 50V

* Not mounted on circuit board
** Mounted vertically, in series

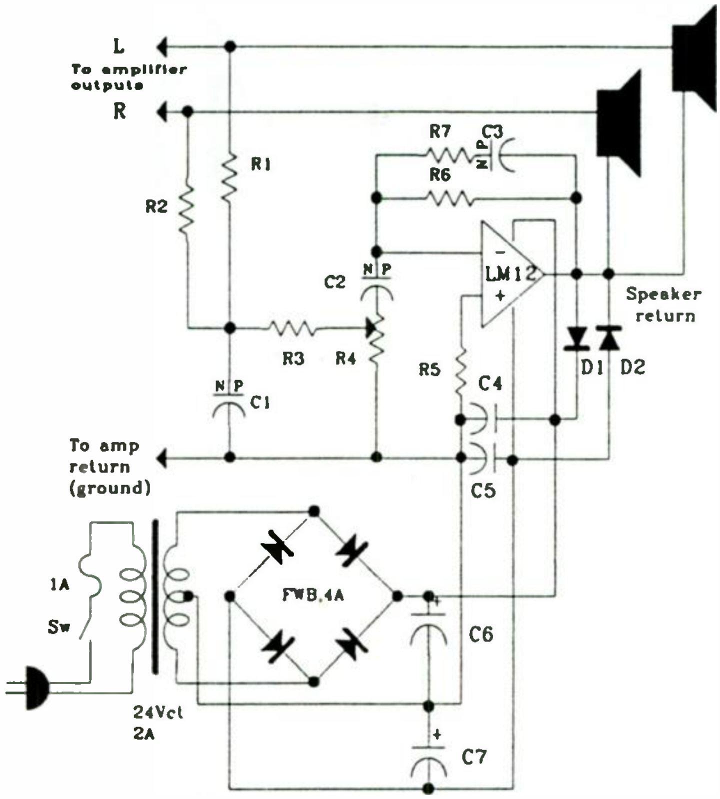


FIGURE 2: Sub-bass boost schematic.

ances (as low as 2Ω for a pair of 8Ω speakers). The new amplifier must be at least as good as the stereo pair it assists in frequency response, distortion, voltage/current capability and low output impedance. The new amplifier provides its benefits as an additive bass amplifier having minimal effect on normal operation of the stereo channels. The LM12 fills the bill for stereo amplifiers with power ratings up to about 60W per channel (this is a current limitation; the LM12 must sink the current from both amplifiers).

The schematic is shown in *Fig. 2*, with the parts list. Note there are no "bandage" components. The only parts required are those which set gain, determine response break points or balance input currents; the sure sign of a friendly operational amplifier.

The combined circuit response is shown at three positions of the control in *Fig. 3*.

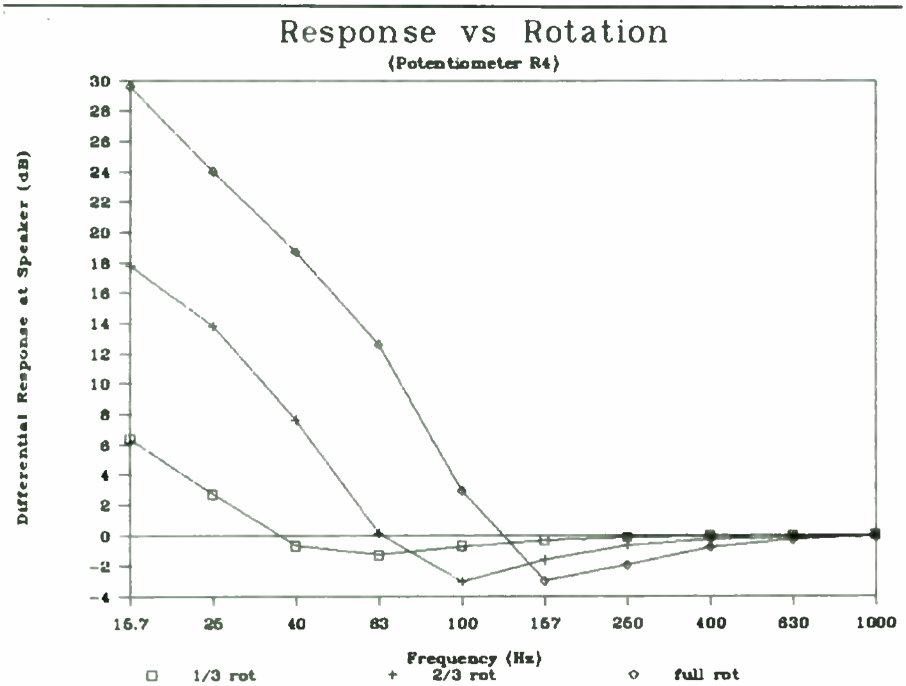


FIGURE 3: Amplifier net response.

The photographs show the completed demonstrator (*Photo 1*) as well as the amplifier circuit board (the smallest of the three) along with two dual units (*Photo 2*), explained later.

Three wires connect to the stereo amplifier and then I connect the speakers to a standard spring-loaded connector at the rear of the unit. I used a full-wave rectified 24VCT transformer (±17V rails) as a power supply for this demonstrator which belies the respectable contribution of about 70W peak RMS in full differential mode (continuous power contribution is about 32W in this mode).[2] A beefier power supply should raise the capability to the 70W continuous level.

At these voltages, by the way, little heat-sinking is required; I simply used the demonstrator's chassis. The device

2. Computed for two 8Ω speakers. When considered in parallel and driven differentially they are seen by the amplifier as a 2Ω load. Since the unit can deliver about 4A RMS before voltage clipping occurs, I^2R computes to about 32W.

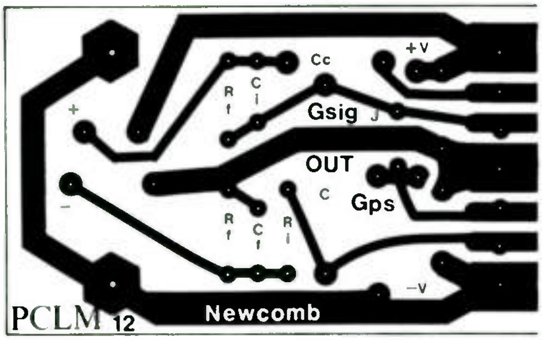


FIGURE 4: Circuit board pattern for the booster.

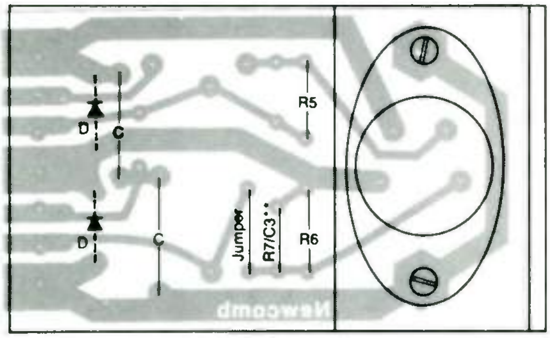


FIGURE 5: Parts placement.

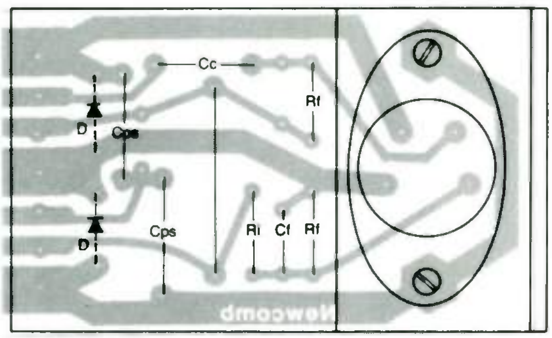


FIGURE 6a: Second parts placement (see *Fig. 6*).

PARTS LIST II

LM12 AMP*

C	15μ
Cc	1.0μ
Cf	22p
Rf (2)	47k
Ri	2.2k
Cps(2)	0.1μ
D(2)	3A, 5PIV
J1-4	Jumpers

*Also, two-channel version (*Fig. 7a*)

PARTS LIST III

BRIDGED AMP

Cc	1.0μ
Rf (4)	47k
Ri	2.2k
Cf (2)	2.22p
CB	15μ
Cps	0.1μ
D(2)	3A, 50PIV
J1-4	Jumpers

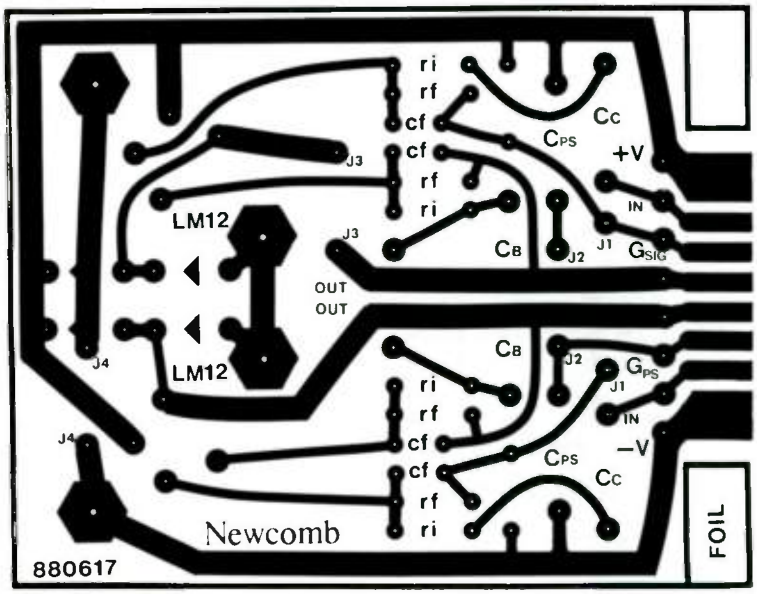


FIGURE 7: Amplifier circuit pattern.

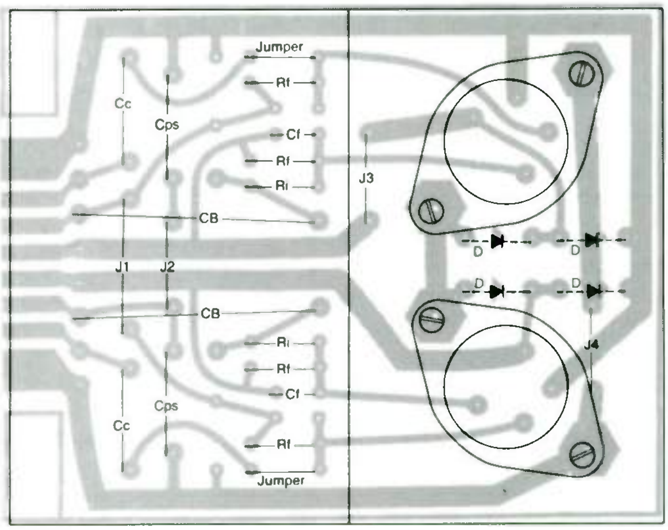


FIGURE 7a: Parts placement for a conventional amplifier: 2 single channels. (Parts List II)

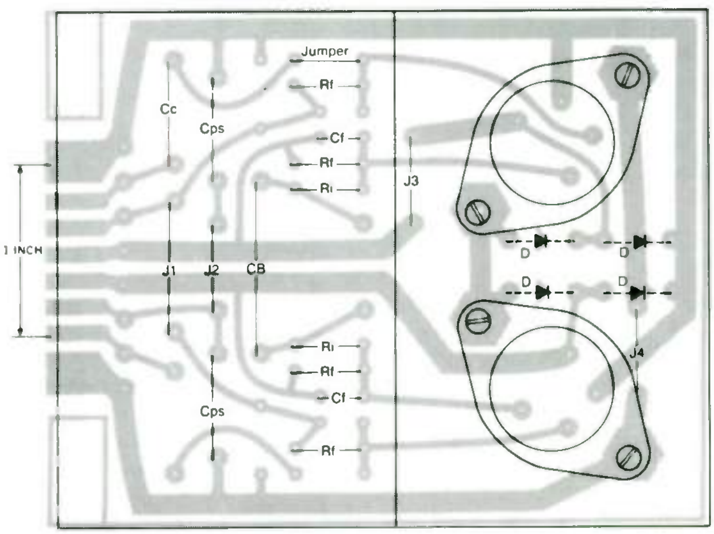


FIGURE 7b: Parts placement for a single-channel bridged unit.

HAFLER PRODUCT UPDATE: *First, the XL-280.* This Excelinear amp is being recognized as just what we claimed it to be—the first amplifier in which distortion of all types is inaudible. Since the XL-280 was introduced 1½ years ago, no one has successfully challenged the validity of the differential test for audible distortion, which the XL-280 passes head and shoulders above the competition. Some individuals may prefer the sound of other amps at times, but it has been our experience that when we were able to measure such a preferred amplifier, there were recognizable indicators which showed that it was, in fact, less accurate. The argument of preference thus seems to be one of complementing inaccuracies yielding a euphonius result.

A few early XL-280s could sometimes exhibit evidence of spurious oscillation, and a few changes were made in early 1987 production. If your XL-280 has a serial number below B1725000, or you have reason to question its high frequency accuracy, we invite you to contact us at (609) 662-6084 for information on possible updating. We will provide the parts, when needed, at no charge, if you wish to make the changes on each module yourself. Some disassembly and soldering is required.

Another update is available for present owners of DH-100 preamplifiers. The just released DH-100 Series 2 has several dB lower noise in both the phono and line level stages. A kit is available for $20 which provides all the parts, including 3 new plug-in integrated circuits, which brings the specs in line with current production. Some soldering is required. If you wish the factory to make these changes, they are included in the normal service charge.

The long awaited XL-600 Excelinear power amplifier is going into production. As a higher power version of the XL-280 circuit, it has the same ultra low phase shift design. The Excelinear topology offers the capability for nulling out audible distortion products while driving your own speakers. In actual use, it can be demonstrated that the Excelinear design has inaudible distortion of any and all types. For more details on the input-output differential test procedure, and how you can use it to compare amplifiers, please write us at 5910 Crescent Blvd., Pennsauken, N.J. 08109.

Other features of the XL-600 include multiple power supplies; an 80% increase in power supply capacitance over the DH-500; and additional output lateral MOSFETs for superior low impedance drive capability. There are added niceties like gold input jacks, a continuously variable DC fan for quieter operation, and built-in mono bridging capability. It's Hafler sound at its finest, and most powerful.

The new sculptured faceplate for the XL-600 has a baby brother which fits the XL-280. This XL-281 is 19″ rack width, and includes sculptured handles. A 17″ wide version, without handles, is the XL-282, serving only to enhance the amplifier's appearance. In typical Hafler value-conscious fashion, there's also a lower cost standard 19″ rack mount, the XL-285.

A brief glimpse of the next Hafler introduction: a totally new, full-control preamplifier with optional infrared remote operation for under $1000. What is really unique is the convenience of analog volume and balance knobs on the remote—the only way to get audiophile precision. Just wait until you try it for yourself! And that's only for starters. It's an all-FET design, with CMOSFET electronic input selection, a cyber-optic passive level control system with precision tracking, a built-in moving coil/moving magnet phono stage, and buffered tape outputs. A big plus is the employment of a building-block concept which can interface with other plug-in audio products supported by the same 12 button remote controller.

At Hafler the quest for sonic excellence and exceptional value continues.

Hafler™

A DIVISION OF
ROCKFORD CORPORATION
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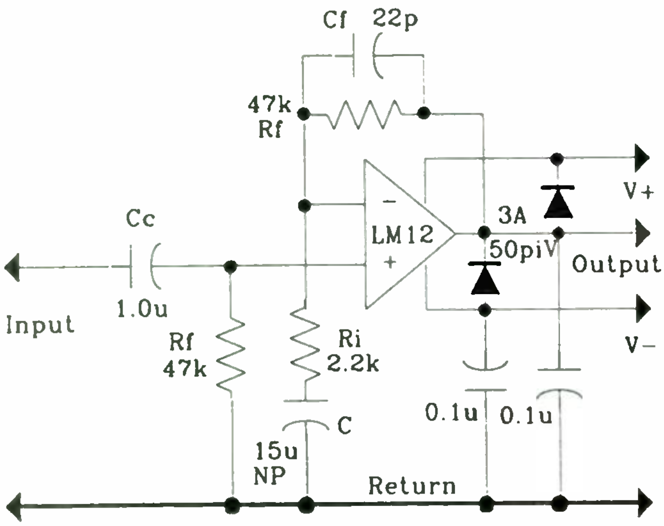

FIGURE 6: An LM12 general purpose audio amp.

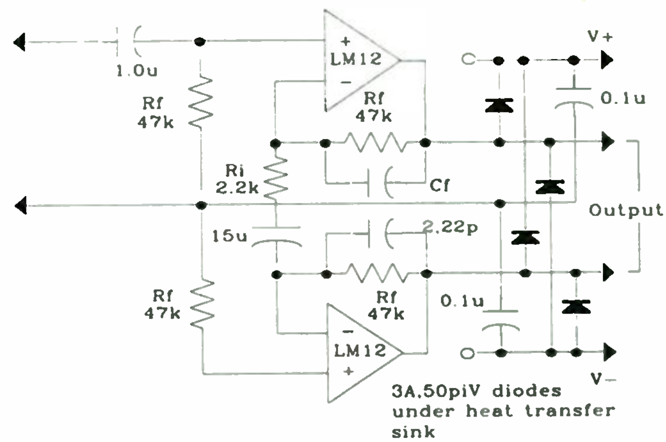

FIGURE 8: A bridged LM12 amplifier.

cannot be harmed by overheating (or little else when properly applied; see the Appendix, *National's LM12*).

Figures 4 and 5 show the circuit board pattern and parts placement for the small board (and the only one I recommend) used in the boosting scheme.

Please note that I made this board for the additional use of the general purpose amplifier, shown in *Fig 6*; the second parts placement is shown in *Fig. 6a*.

The larger board (*Figs. 7* and *7a*), can support a configuration for conventional amplifier applications: two single channels at 50W RMS each into 8Ω, or 100W into 4Ω with a suitable supply.

The same board can also be stuffed as a single-channel bridged unit (*Fig. 7b*) capable of driving about 200W into an 8Ω load (and little more for 4Ω due to the LM12's 10A limit). *Figure. 8* is a schematic of the bridged configuration I use.

I am currently using the LM12 in all the configurations discussed and have found few, if any, quirks.

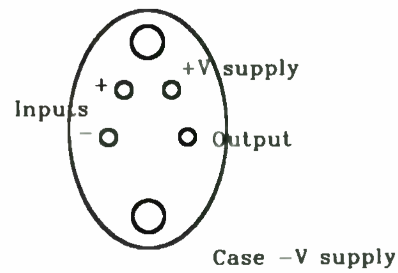

FIGURE 9: LM12 pinout.

3. Paraphrased here from National's preliminary data sheet, *LM12(L/C/CL) 150W Op Amp*, December 1986.

APPENDIX

National's LM12

We all look forward to new advances in technology, especially when they take the form of an elegant solution to an old problem. The National Semiconductor LM12 is a monolithic power operational amplifier with a host of impressive specifications[3] I greet this recently available power IC, however, with mixed emotions. It takes all the frustration, and adventure, out of designing and building power amplifiers. Features include:

- Input protection: Diodes provide reverse voltage protection by shorting excessive input voltages.
- Controlled Turn-On: No signal is passed to the output until the combined supply voltages reach 14V. If input circuits are designed to stabilize before this happens, turn-on/turn-off "thump" can be controlled in many applications.
- Dynamic Safe-Area Protection: Instantaneous peak-temperature limiting is built in. Allowing a short term dissipation of 800W permits the handling of reactive loads without derating.
- Thermal Limiting: Signal output is disabled when the case temperature reaches 150°C.
- Overvoltage Shutdown: When the supply voltages approach BV_{CEO}, the output is disabled and can withstand overvoltages of 100V, according to National. The voltage rating is not necessarily the maximum usable voltage however. I use the cheapest issue, the LM12CLK (±40V, 0 to 70°C operating range, costing about $30 in small quantities) and the safe operating area protection shuts down when the rail voltages reach between ±33 and ±35V.
- Output Current Limiting: Can deliver ±10A at rated supply voltages (±12A for the more expensive version).
- Total Harmonic Distortion: I don't have the facilities for distortion measurements but National claims less than 0.01% from 10Hz to 10kHz (less than 0.02% from 30Hz to 20kHz) at ±1V output and dropping as power increases.
- A Fast Sucker: With a small-signal bandwidth of 700kHz, the unity-gain compensated device has a power bandwidth of 60kHz and a very respectable slew rate of 9V/μsec. It can also handle capacitive loads with a 4μH choke wound around a 2.2Ω resistor in series with the load.

The unit is very well behaved when driven to voltage clipping, current limiting or in any one of its shutdown modes (no signal to the output). There are two versions of the device with ratings of ±40V and ±50V and both are available in commercial and military temperature ranges. It is packaged in a 4-pin modified TO-3 with the case at −V (you can ground the device when using a positive single ended supply). National says a socket is available from Augat (part #8112-AG7).

TABLE I—LM12

Parameter		Typical
Input offset voltage		2mV
Input bias current		0.15μA
Input offset current		0.03μA
CMRR		86dB
PSRR		90dB
Output saturation	1A	1.8V
Threshold	8A	4V
	10A	5V
Large signal voltage gain		>50,000
Power dissipation		>80W
Idling suppling current		60mA

THE UNLINE:
DESIGNING SHORTER TRANSMISSION LINES
BY JOHN COCKROFT

In the years since Dr. Bailey's straightforward presentation, the transmission line loudspeaker enclosure has become an object of myth and fancy. Useful lines have been produced by some which are among the finest music reproducing systems yet devised. Others' attempts have ended up more useful as doghouses.

The original concept's rather large enclosure, optimally designed for a large listening room, is capable of awesome performance. I think the principle of the TL is excellent and the benefits of the nonresonant performance should not be limited to such an occasionally ideal environment.

Large TLs have extended low-frequency performance well into the nether regions of the threshold of human hearing. Equally important, they do so in such a smooth and natural manner, avoiding most problems designers experience with other types of enclosures. These side benefits of transmission lines have held my interest over the past several years.

As vented and sealed systems are designed in a myriad of sizes and parameters for a multitude of uses, why shouldn't the TL have the freedom to live a multifaceted existence for our listening pleasure?

To this end I have empirically designed over a dozen lines intended for use in small listening rooms. In principle, most performed quite well. Generally, I encountered the usual troubles that crop up when designing any type speaker system, such as matching speakers, balance and so on. I believe transmission lines can be used in many applications and have the potential to be extremely useful.

I recently took a retrospective look at my more successful TL systems and formed some general relationships. After considering and evaluating their empirical qualities, I developed an arbitrary "standard" transmission line enclosure that relates very closely to several of my systems. By using my concocted "standard" TL and others as points along a line, I came up with some equations that describe my speakers rather accurately and allow for a continuum of other enclosures between and beyond the ones I have constructed.

A CONTINUUM. The "standard" speaker I chose for the examples is reasonably close to Bailey's original. I could have used Bailey's exact line as the standard (and so can you), but it doesn't describe my speaker designs quite as well and that is the purpose of this little project.

The standard system I chose has a length of 72"; is stuffed with polyester fiberfill at a density of 0.5 lb./ft.^{3}; the minimum cross section equals the cone area of the woofer; and speaker unit Q_{ts} is 0.4. I never built this enclosure, of course. I use it here merely as a model on which to base other systems.

My procedure presents equations that predict line length, stuffing density, minimum cross-sectional area and speaker Q_{ts}, for enclosures with lengths shorter than the model system. You may design an enclosure starting with line length, or if you prefer, on the basis of speaker Q_{ts}.

First, I'll consider developing enclosures choosing a line length shorter than the model's. Once the length is settled the question of stuffing density comes up. It is solved by Equation 1:

$$d_1 = d_s \times \frac{1}{(LL_1/LL_s)^{1/2}}$$

where

d_1 = the desired new density for the shorter line
LL_1 = The desired shorter line length
LL_s = The "standard" line length
d_s = The "standard" density of 0.5lb./ft.^{3}.

After the density of the stuffing is determined, the minimum line cross section is considered in Equation 2:

$$Cs_1 = (d_1/d_s)Cs_s \times Ca$$

where,

Cs_1 = the desired minimum cross section;
Cs_s = The "standard" minimum cross section, in this case 1 times the cone area;
Ca = the area of the woofer cone. The cone diameter is usually measured from the center of the surround to the center of the surround.

Finally, the Q_{ts} of the woofer is calculated in Equation 3:

$$Q_{t1} = Q_{ts}(d_1/d_s)^{1/2}$$

where,

Q_{t1} = the desired speaker Q_{ts};
Q_{ts} = the "standard" speaker Q_{ts} of 0.4.

As you can see from the equations, as the line shortens, the stuffing density increases. To compensate for this, the minimum cross section also increases to give the speaker a better opportunity to "breathe" in the denser environment. The increased density tends to overdamp the speaker, so a higher Q_{ts} is required.

In my previous article ("The Shortline," *SB* 1/88, p. 18) I mentioned shorter lines might begin to function as a hybrid design between the classic line and the

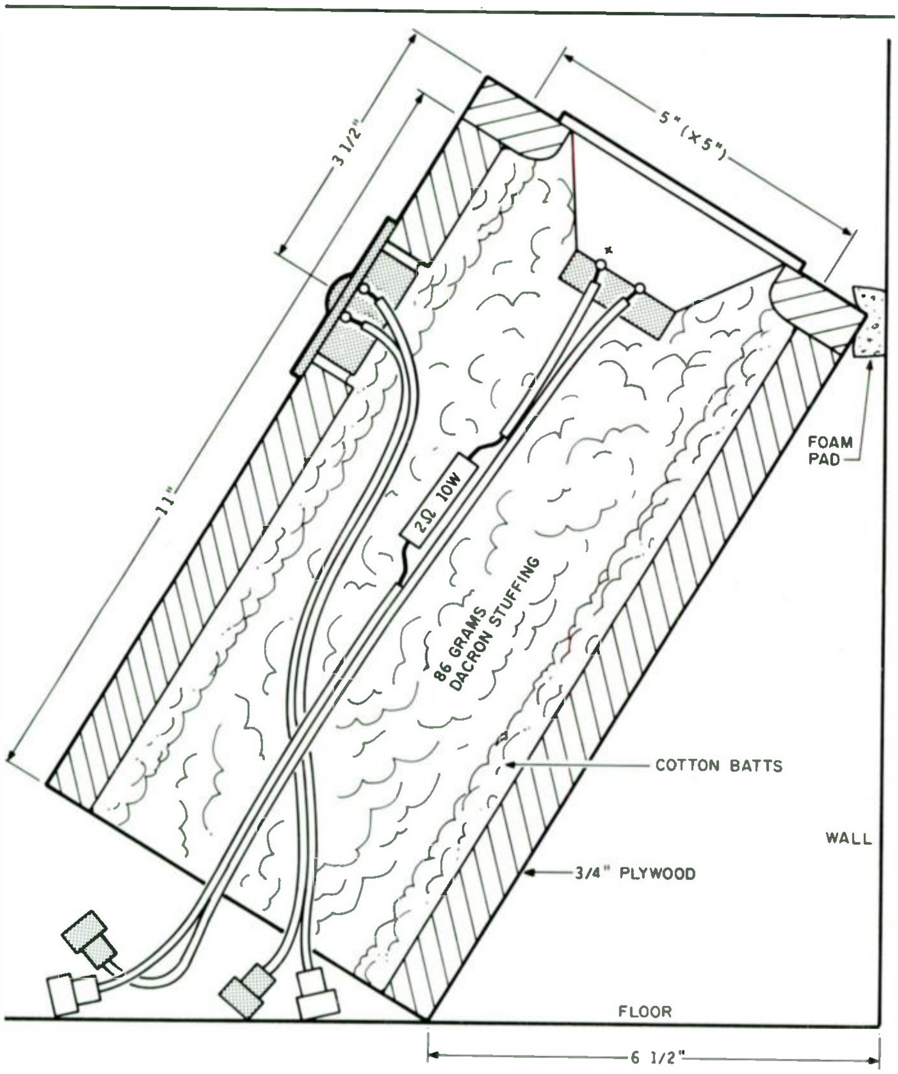

FIGURE 1: Diagram of the Unline showing the tweeter properly mounted. The original was biamped using a Marchand XM-1 fourth-order crossover at 3.5kHz.

aperiodic, pressure-relief type of enclosure, or as a "bottomless box." The sound remains very much "transmission line" in its general characteristics. The resonant frequencies remain low when compared to the aperiodic types.

OCTALINE FORMULAS. For an example, let's examine my Octaline speaker from *SB* 3/87, using the above equations. It is just about 36" long. Using Equation 1:

$$d_1 = 0.5 \frac{1}{(36/72)^{1/2}} = .7071 \text{ lb./ft.}^3$$

with Equation 2:

$$Cs_1 = (.7071/.5)1 = 1.414 \times (\text{cone area of } 8.9 \text{ in.}^2) = 12.6 \text{ in.}^2$$

Now Q_{ts}:

$$Q_{t1} = .4(.7071/.5)^{1/2} = .475$$

I constructed the Octaline a couple of years before I evolved these equations; I used a minimum cross section of 12.2 in.^{2}, and a Q_{ts} of about 0.505. Working empirically, the Octaline is within 6.3% of Q_{ts} and 3.2% of the minimum cross section of my "standard" line. Or, perhaps, my "standard" line is within those percentages of the Octaline. At the time I conceived the Octaline I wasn't aware of the role Q_{ts} might play in TL design.

To determine the parameters of a line for a speaker unit with a given Q_{ts}, use the following equations; begin by calculating the stuffing density (Equation 4).

$$d_1 = (Q_{t1}/Q_{ts})^2$$

Proceed with the cross section by using Equation 2. Finally determine the line length. (Equation 5):

$$LL_1 = LL_s \times \frac{1}{(d_1/d_s)^2}$$

As an example of the above, say you had a speaker with a Q_{ts} of 0.43. Then:

$$\text{stuffing density} = 0.5(0.43/0.4)^2 = 0.578 \text{ lb./ft.}^3$$

and:

$$\text{min. cross section} = (.578/.5) = 1.16 \times Ca$$

and:

$$\text{line length} = 72 \times \frac{1}{(.578/.5)^2} = 53.9''$$

Aside from having the correct Q_{ts}, to be successfully used with these equations a speaker should have a low F_{sa}. I haven't been able to work this parameter into my model, but I believe the lower the F_{sa}, the better. Using a speaker with an F_{sa} in the 30Hz area sounds much better than a speaker with an F_{sa} around 40Hz, and I'm sure those approaching 20Hz are even better.

Of course, these differences will only be noticed when music of ultra low frequency content is heard. For most music, speakers with 40Hz resonance will be quite adequate, particularly if Q_{ts} is at least .4 or higher. Lower Q_{ts} figures result in excessive roll-off of important musical content. Even with Q_{ts} .5, Q_{ts} will be down 6dB. Q_{ts} .3 will be 3dB down at 121Hz. It has been my experience that TL enclosures don't raise Q_{ts} greatly, as for instance a sealed box does.

If you plan to raise the Q_{ts} of a given speaker, I recommend you add mass to the driver cone, rather than using series resistance. In both cases you will lose efficiency, but with more mass you will lower F_{sa}, which is a much better bargain than a mere efficiency loss.

A speaker with a long voice coil overhang probably responds better when adding mass, since the loss of control with the higher Q_{ts} increases the tendency to overshoot, and the long coil gives more leeway for linear signals. Adding mass may make an infrasonic filter necessary, but I think one should be used in any case.

ACID TEST. Now, finally, let's look at the Unline. I built it because I wanted to give my equations a real test, to make sure they work. A small enclosure, that I had used on occasion as a transmission line loaded midrange, was 5" by 5" by 11," with a hole for a 4" speaker on top; the bottom was open. (What I'm calling

Continued on page 32

BOOKS from Old Colony Sound Lab

HOWARD W. SAMS

S-4 UNDERSTANDING IC OPERATIONAL AMPLIFIERS [3RD ED] *by Roger Melon* and *Harry Garland.* Basic course in semiconductor electronics. Covering integrated op amp circuitry, design factors, bias current, offset voltage, frequency compensation and slew rate. 212 pp., softbound. Each **$12.95**

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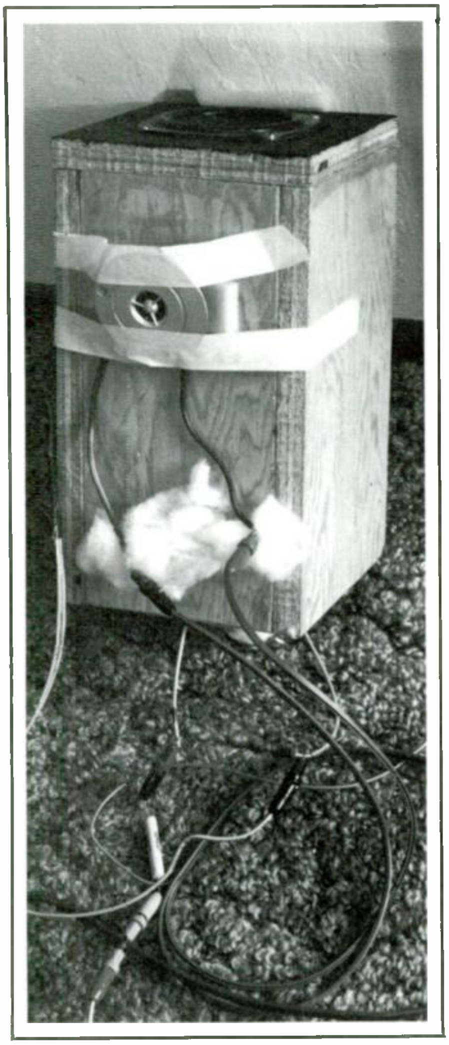

PHOTO 1: The Unline in all its unglory. The wad of cotton on the enclosure front keeps the tweeter leads from vibrating. Note the 2Ω series resistor used to raise Q_{ts}.

the bottom was actually the back when I used it as a midrange.)

I first thought I may be crazy to consider using this enclosure, a thought you may share, along with many others. Later, curiosity overcame my self-deprecation and I worked out the equations for the little box. The stuffing density came to 1.279 lb./ft.^{3}, hardly anything described by Dr. Bailey. The minimum cross section, at 22.8 in.^{2}, was feasible at least, as my enclosure's cross section is 25 in.^{2}. Q_{ts} came to 0.6398 and now the whole project looked like a loser and a waste of my time. I didn't have a 4" speaker with a Q_{ts} that high.

After a lot of stalling, I decided to proceed. It would be an acid test for the equations. I modified a Radio Shack 40-1022[1] which yielded a calculated Q_{ts} of 0.66, using 2Ω series resistance and 10 grams of ⅛" solid lead wire around the dustcap. 1.7Ω would have come to 0.638, but I didn't have the necessary values. Using series resistance is going against my own advice, but I couldn't see wasting a woofer on my hare-brained project.

I estimated the net internal volume at 0.149 ft.^{3} and used 86 grams (2.9 oz.) of polyester fiberfill. In addition, I lined the walls with a layer of surgical cotton, as it comes off the roll, which I spot-glued in place.

I connected 14-gauge wire to the woofer and terminated the ends in banana jacks. The wires merely extended out the bottom of the enclosure. As this was a quick-and-dirty breadboard type project, partly due to lack of faith in it and partly due to my natural laziness, I merely taped the Radio Shack 40-1376 tweeter to the enclosure (*Photo 1*).

I leaned the Unline against a wall, with the bottom rear edge 6 1/2" from the base. (I figured it needed all the rest it could get if it was going to do what I expected of it.) I biamplified the Unline through a Marchand XM-1 fourth-order crossover using a crossover frequency of about 3.5kHz.

CONCLUSIONS. As it turns out, the darned little thing works, and very well indeed. It sounds very much like my other lines, which isn't so surprising, since they use the same components. The sound is clean and open and very natural. On my Michael Murray organ CDs (Telarc), I am sure some fundamentals are missing, but not noticeably, or in any significantly disappointing way. It sounds "right."

Of course, the small driver, dense stuffing and 2Ω series resistor means it isn't as efficient as my other lines. The overall maximum SPL isn't as great either, but in my small apartment room of 1400 ft.^{3} it provides more volume than I want. I measured a single impedance peak of 22.6Ω at 34Hz (free air resonance was about 39Hz). I am quite pleased with the Unline's performance, which vindicates my equations.

As a further check, I have constructed 5, 8, and 10" driver versions of the Unline. All perform well. Should you be tempted to experiment a bit, I would be happy to know your results. For those interested, I have devised *Table 1*, listing various lines as a function of Q_{ts}, ranging from the "standard" to approximately the Unline. To arrive at values between those given, just interpolate, or work out the equations on any calculator with an 1/x and a square root key.

You can use this method with your favorite transmission line enclosure. For instance, if your "standard" model is a 10-foot line with a 0.5 lb./ft.^{3} stuffing density, a minimum cross section of 1.1 times the cone area and a Q_{ts} of 0.35, then a 4-foot line would require: density, 0.79; minimum cross section, 1.739 × cone area; and Q_{ts}, 0.44. I used a 0.5 stuffing density as it seems popular, but it could be 0.4, 0.55 or whatever you prefer. (I doubt if 0.4 would be a viable working density.)

A word about adding mass to loudspeaker cones: If the mass is increased by a factor of 1.5, Q_{ts} would be increased by $1.5^{1/2}$ and the resonant frequency is reduced by $1/1.5^{1/2}$. Of course all of this lowers the efficiency of the speaker; it drops to $1/1.5^2$, or about 6.3dB.

Thus, if you have a speaker with a cone mass of 20g, a Q_{ts} of 0.35, an F_{sa} of 40Hz and an efficiency of 90dB/W/M, you could add 10 grams to the cone to arrive at a speaker with a Q_{ts} of about 0.429, F_{sa} about 32.7Hz and an efficiency of about 83.7dB/W/M. I find the best

Continued on page 36

TABLE I

Transmission lines based upon shorter variations of an arbitrary "standard" transmission line with a length of 72;" a stuffing density of .5 lb./ft.^{3} of polyester pillow stuffing, a minimum crossection of 1x the speaker cone area and a speaker Q_{ts} of .4.

Q_{ts}	d_1	LL_1	Cs_1
.40	.500	72.0	1.00
.41	.525	65.3	1.05
.42	.551	59.3	1.10
.43	.578	53.9	1.16
.44	.605	49.2	1.21
.45	.633	44.9	1.27
.46	.661	41.2	1.32
.47	.690	37.8	1.38
.48	.720	34.7	1.44
.49	.750	32.0	1.50
.50	.781	29.5	1.56
.51	.813	27.2	1.63
.52	.845	25.2	1.69
.53	.878	23.3	1.76
.54	.911	21.7	1.82
.55	.945	20.2	1.89
.56	.980	18.7	1.96
.57	1.015	17.5	2.03
.58	1.051	16.3	2.10
.59	1.088	15.2	2.18
.60	1.125	14.2	2.25
.61	1.163	13.3	2.33
.62	1.201	12.5	2.40
.63	1.240	11.7	2.48
.64	1.280	11.0	2.56

d_1 = stuffing density
LL_1 = line length
Cs_1 = minimum crossection factor

1. Radio Shack's current 4-inch woofer version is the 40-1022A, which may not give the same results in this application. See Mr. Cockroft's reply to Mr. Lewellen in this issue's *Mailbox* section, for details.

ACTIVE FILTER DESIGN WITH CROSSOVER CALC

BY FERNANDO GARCIA VIESCA and MARCO A. PEREZ

Simulating parameters is one of the many applications of that useful device, the op amp. The active filter, which replaces bulky, expensive inductors and large value capacitors with only resistors and small value capacitors, captured many design engineers' fancy. Not only is the circuit more compact and less expensive, but other quickly realized benefits are better signal-to-noise ratio and elimination of required impedance matching. Nevertheless, active filter design remained outside the reach of the average electronics experimenter, mainly due to the complex mathematical calculations required for this type of circuit.

Fortunately, the need for this circuit is so great that many engineers have worked to find a simplified calculation procedure. We use one in the following program,[1] that allows you to calculate any high- or low-pass filter with the ease of just loading a BASIC program. The key to these simplified filter calculations is using *precalculated* constants for the different filter types.

We shall use the constants for a Butterworth filter, in four different configurations (high- and low-pass in second- and third-order). We chose the Butterworth response because it is a *maximally flat* filter, that is, the actual filter response very closely approaches an ideal one.

PROGRAM DESCRIPTION. The program is divided into four main sections and two subroutines, as follows:

Third-order low-pass filter: Line 140. For a given frequency, this program calculates the capacitor and resistor network for a low-pass response.

Second-order low pass-filter: Line 400. Same as above, but this network has only two capacitors and resistors.

Third-order high-pass filter: Line 640. For a given frequency, it calculates the capacitor and resistor network for a high-pass response.

Second-order high-pass filter: Line 890. Same as above but the network has only two capacitors and two resistors.

Frequency-select subroutine: Line 1110. Prompts the cutoff frequency (hertz) and converts it to radian-second.

Order-Selection subroutine: Line 1170. Prompts the filter-order and directs the program to the proper program segment.

The circuit configurations you use with this program are shown in *Fig. 1*. You can see that all the resistors for the low-pass filters and all the capacitors for the high-pass filters are equal-valued. Each respective program section, therefore, prints only a single value. For example, in a second-order high-pass filter there will be R1 and R2, but only one C, so substitute this value for all similar parts.

USING THE PROGRAM. Remember this useful trick before starting the program: It is much easier to find a precise value for a resistor than a capacitor. Set the capacitor's value to a standard, easily availiable part, and let the resistors be the odd valued parts. In the very worst case, odd resistor values may be achieved with a trimpot. Also, all resistor values should be entered in full; a 6800Ω resistor should not be entered as 6.8k, even though I show it that way in the text, for the sake of clarity.

The program starts with a prompt:

```
WHAT KIND OF FILTER WOULD YOU LIKE?
(1) HIGH-PASS    (2) LOW-PASS
```

You may choose either configuration. Then:

```
CHOOSE THE ORDER THAT YOU WISH
(2) 2ND-ORDER    (3) 3RD-ORDER
```

When you select the filter order, the program then prompts the frequency:

```
ENTER FREQUENCY=>
```

The frequency should not be zero. The frequency is converted to radians-second and then the program goes to the proper filter configuration and order.

LOW-PASS FILTER DESIGN. Both low-pass (second- and third-order) sections of the program are identical. The constants are different and the third-order filter has an additional resistor/capacitor pair. The program, after loading the constants, asks for the resistor value:

```
ENTER R=>
```

You may then enter any value of resistance different from zero. A good starting point is to use some standard resistor value, like 47k, 120k, 1M. The actual value at this point is not important. After the resistance value has been entered, the program calculates the capacitors' values and shows them on your screen. For example, for a second-order low-pass filter with $f_c = 60Hz$, and a resistor value of 1.2k, you should see the following values:

```
C1=3.125629µF
C2=1.562815µF
TRY ANOTHER RESISTOR VALUE? Y/N
```

1. The description of those procedures is beyond the scope of this short article. Those interested should read: *Handbook of Operational Amplifier Filter Design*, D. Stout and M. Kaufmann, Chapters 10 and 11, McGraw Hill Co., 1976.

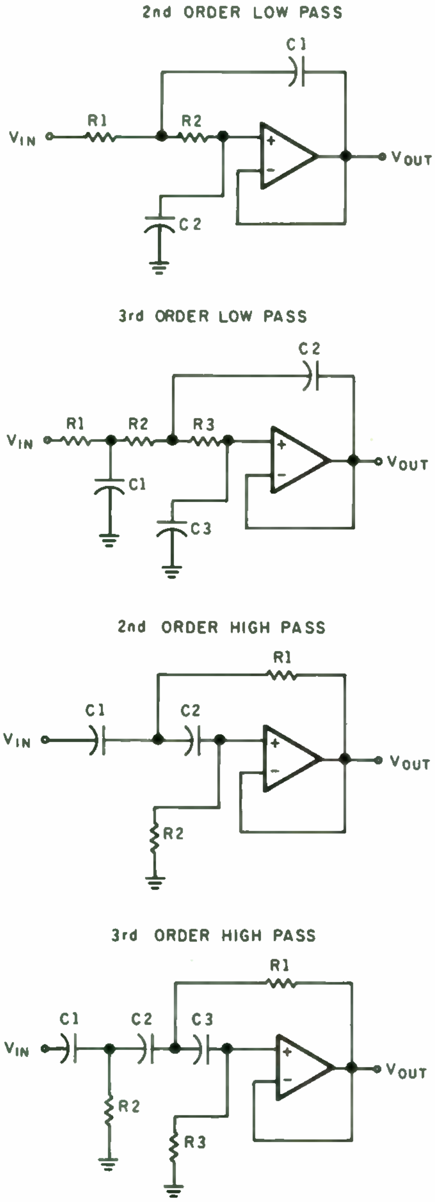

FIGURE 1: Crossover configurations.

Since a 12kΩ resistor yields a capacitor value that does not exist, type "Y" and the program will prompt you again for another resistor value. And what should this value be? Since an increase in the resistor value will proportionately decrease the capacitor value, then you should increase the resistor value an amount that will yield a practical capacitor value. You can see that a resistor 31.25 times larger will yield a capacitor of 0.1µF for C1 and 0.05µF for C2 (round off the decimals), which are very common values. Therefore your resistor should be 31.25 × 1.2k = 37.5k resistor (37.4k is the nearest 1% value). We enter 37.5k then, and the following display should appear:

C1 = .1000201µF
C2 = 5.001007E − 2µF
TRY ANOTHER RESISTOR VALUE? Y/N = >

Since those capacitors are now practical, we enter "N" and then the following display should appear:

2ND ORDER LOW PASS
C1 = .1000201µF
C2 = 5.001007E − 2µF
R = 37500Ω
FREQ. = 60Hz

The third-order low-pass filter follows exactly the same procedure, but it calculates three different capacitor values.

Press return, and the program prompts the following message:

DESIGN ANOTHER FILTER Y/N = >

If your answer is no, you are returned to BASIC; if yes, then you are asked the filter type, the order and the frequency again.

HIGH-PASS DESIGN Let's suppose we chose a third-order, high-pass filter with a corner frequency of f_c = 1kHz in the previous step. After preliminary calculations, the program prompts for an impedance constant:

ENTER IMPEDANCE SCALING CONSTANT =>

This constant will modify the capacitor and resistor values. Right now, give it a value of 1. The following display should appear:

R1 = .282Ω
R2 = .7184Ω
R3 = 4.941Ω
C = 159.1551μF
TRY ANOTHER CONSTANT? Y/N =>

For a capacitor value, 159.1551μF is too large, and 0.282Ω is too small for a resistor value. The impedance scaling constant allows you to decrease the capacitor's value and increase the resistor's value by the same factor. Since it is easier to find odd valued resistors than capacitors, let's use a constant that will give an available capacitor. If we choose a 0.01μF capacitor, then the constant should be: 159.1551/0.01 = 15915 (disregarding decimals).

So, answer yes to the question and use this constant. The computer will then display the recalculated values:

R1 = 4488.031Ω
R2 = 11433.34Ω
R3 = 78636.02Ω
C = 1.000032E-02μF
TRY ANOTHER CONSTANT Y/N =>

The standard 1% resistor values are 4.53kΩ, 11.5kΩ and 78.7kΩ, respectively, for R1, R2, and R3. These are practical value resistors, therefore answer no. The computer displays the final calculated values:

3RD ORDER HIGH PASS
R1 = 4488.031Ω
R2 = 11433.34Ω
R3 = 78636.02Ω
C = 1.000032E-02μF
FREQ = 1000Hz.

The procedure for the second-order high-pass filter is identical, but only two resistor values will be displayed, instead of three. After you press return, the program asks:

DESIGN ANOTHER FILTER? Y/N =>

Yes, allows you to calculate another filter configuration, and no returns you to BASIC.

TIPS AND TECHNIQUES. The best computer program may be rendered useless if you allow sloppiness to creep into the real world. Please use the following tips whenever building an active filter:

Choose the op amp carefully. You would be unwise to use a common 741 for a high-pass filter with a cutoff frequency of 25kHz. The slew rate and power bandwith of the op amp should be taken into account. Also choose low noise op amps if you will use these filters in high quality audio applications.

Choose the proper capacitor. Use the closest tolerance capacitor available (usually 5%), and with good temperature stability. Do not use electrolytics, or even tantalum types; NPO ceramics or polystyrene capacitors are best.

Choose the proper resistor value. You've realized while working with the program that a capacitor value may be substantially decreased by a resistor's increase in value. This is desirable, but a high impedance circuit is easily affected by noise and the op amp's bias currents. Try the smallest resistor value that will yield an availiable capacitor. But don't go too far in the opposite direction. A small impedance will overload the op amps.

The signal source impedance must be at least 10 times smaller than the smallest resistor value; if not, use an input buffer.

Note: The program was written in GW-BASIC—some statements, particularly the clear screen command (CLS), may be different in other versions of BASIC.

Copies of Mr. Viesca's program are available on 5¼," 360K DSDD IBM compatible disk by sending $5 to Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458. Letters or queries to Mr. Viesca should include a stamped, self-addressed envelope and one international Postal Coupon.

UNLINE

Continued from page 32

way to add mass is to glue a ring of lead or solid core solder, around the cone's dustcap junction. Make the ring curve equal to the dust cap's circumference and cut into 4 segments; it will lay flatter and fit into the junction better. I usually use white glue, which so far has held well, even on polypropylene. You may prefer RTV silicone rubber adhesive. If the cone has a viscous coating I don't know what works best.

I hope this article brings a glimmer of renewed interest in transmission line theory, and particularly smaller systems. There is no reason why those who lack the space for a classic line should be deprived of naturalness in their sound systems.

A LOW-COST TWO-WAY RIBBON SYSTEM

BY RICHARD PAINTER

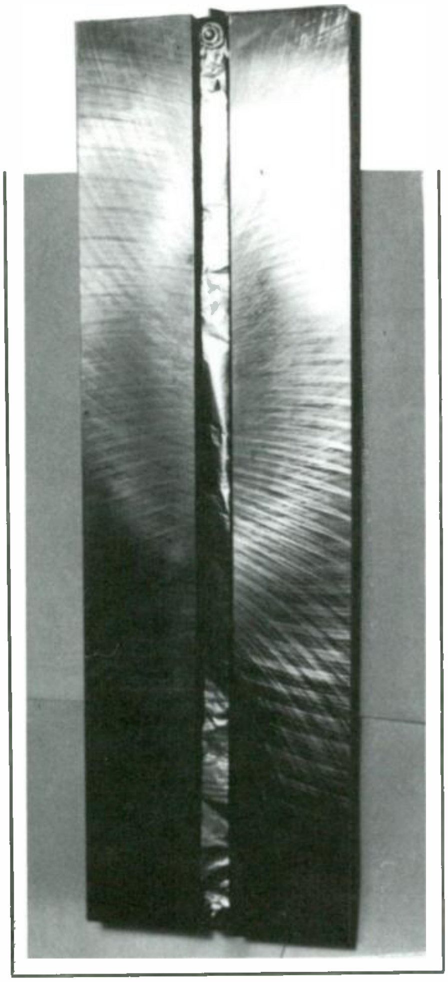

PHOTO 1: The completed ribbon speaker is 16″ tall.

Probably every *SB* reader has at one time or another read objective and subjective praise for ribbon systems. Audio magazines list ribbon systems among the best speakers obtainable at any price.

But the price! What's a poverty-striken audiophile to do? An Infinity IRS costs $45,000; an Apogee Duetta is about $3,000. In fact, very few systems of this type are priced at less than $2,000 per pair.

However, when I want a piece of audio equipment I can't afford, I turn to *TAA* and *SB*. In this case, I read the article by Michael Lampton and J. Henry Primbsch, "Simple Ribbon Tweeters" (*SB* 3/84). Their comprehensive work on the subject allowed me to construct a ribbon driver for a two-way ribbon/cone system which suited my needs (*Photos 1* and *2*).

I wanted to build a two-way system with full range frequency response. But why build a ribbon in the first place? There are cone or dome drivers to suit almost any need. And cone drivers for low-frequency reproduction are much easier to work with. For me this means using a 10-inch or larger woofer for adequate low-frequency power and a crossover at or around 1kHz.

However, I know of no dome tweeter that will give a response down to 1kHz. Also, the moving mass of a ribbon is much less than a dome tweeter. A ribbon typically has only a few milligrams of moving mass, compared to around 200mg for a dome tweeter!

A dome tweeter may have a 10 or 20-ounce magnet as well, while my ribbon currently uses 1.08kg or about 39 oz. of Alnico V. These two factors combine to produce the faster, more detailed sound characteristics usually associated with ribbon drivers.

After reading up on ribbon construction, however, I was a bit discouraged. The ribbons in the *SB* article were low efficiency, and were recommended only for 4kHz and above. I am a devoted tube freak and prefer high driver efficiency, to save on output tubes, and a 1kHz crossover, or lower. Looking at some commercial ribbon drivers convinced me the lower crossover point was possible, but to get the efficiency I wanted would be a challenge. I decided to try a 16-inch ribbon in a magnet structure consisting of two steel pieces, ⅛-inch thick, 2 by 16 inches; and a single steel backplate, 4 by 16 by ⅛″ (*Fig. 1*).

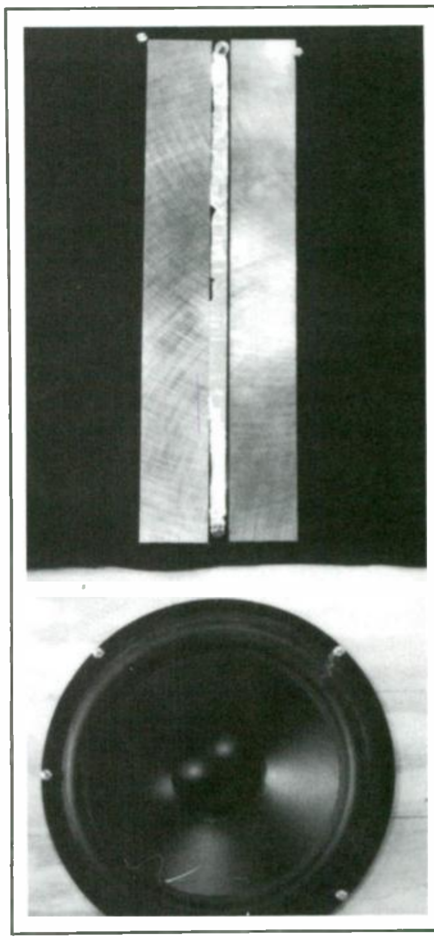

PHOTO 2: The ribbon midrange/tweeter atop my transmission line.

Fortunately, I know an excellent machinist, Arnold Kaczor, who made the plates from scrap steel. If you do not have the equipment to make these plates, I am sure most machine shops could produce the pieces, or you may contact Mr. Kaczor (see Sources).

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1. Colloms, Martin, "The Flat Response," *Stereophile*, Vol. 10, No. 1, p. 110.

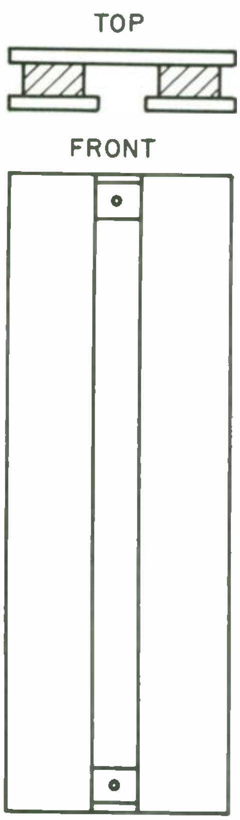

FIGURE 1: Construction design—16 by 4 by ⅛ inches (back plate).

MAGNETS. Edmund Scientific stocks a rectangular magnet which is suitable. I ordered 48 (about $25 per channel, plus shipping). When I had both the steel and magnets I arranged them as shown in *Photo 3*. Caution: Do not get your fingers, or anything else you value, between the two speaker plates. They will snap together like a guilliotine *with no warning*. For safety, while working I placed ½-inch wood blocks between them.

I now had to find a ribbon material. I broke open several cheap capacitors, and found a useable ribbon made from the foil of an old ceramic body, 0.1μF 600V cap *(Photo 4)*. I cut a 16-inch length of this material into ½-inch wide strips. This gave me a low mass and a load of about 0.5Ω. While this is quite a low impedance, I find it gives my old Dyna MK VIs no trouble at all. (I tried Adcoms and Haflers which also work well.)

Now, could I get the efficiency I wanted? I had already built a compact transmission line enclosure for a set of SEAS P-25 REX 10-inch woofers, and since these run 93dB/W/M, if I could get the ribbons to keep up with them I would be satisfied. I started with 12 magnets per channel and found I was about 6dB short of my goal. But with 24

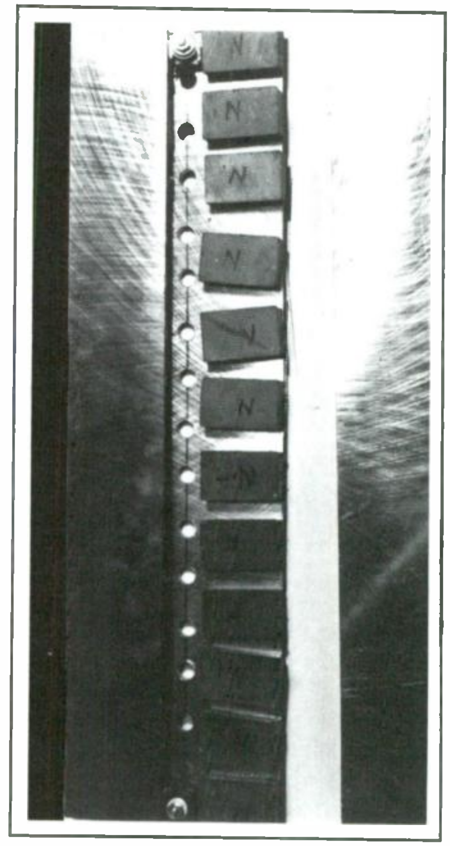

PHOTO 3: Magnet placement and plate construction.

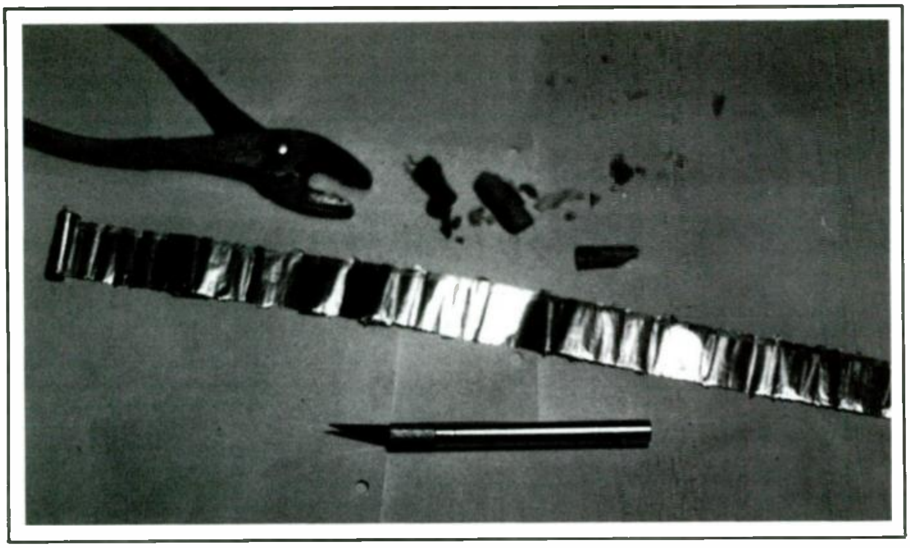

PHOTO 4: Ribbon material.

PARTS LIST

Qty.	Part
2	16 x 4 x ⅛″ steel with ½″ holes
4	16 x 2 x ⅛″ steel
48	Alnico magnets 1½ x 1 x ⅜″ 4lb. lift
2	16 x ½ x ⅛″ felt
2	5-way binding posts
1	Ceramic bodied capacitor (old TV set)
2	20μF Solen polypropylene caps
2	.01μF Siemens polystyrene caps

magnets per channel, the ribbons were every bit as efficient as I could wish.

I used 32μF capacitance in series with each ribbon and got a 6dB/octave, 1kHz crossover point. I paralleled a 22μF NP electrolytic, which I had on hand, with a 10μF WonderCap. Actually, without an enclosure, the ribbons roll off at 1kHz without a crossover, due to cancellation effects, but the caps protect the fragile ribbon material from excessive low-frequency excursions. An additional 0.01μF polystyrene bypass cap cleared up the high frequency detail considerably. Experimenting further, I found a 20μF Solen cap (polypropylene) bypassed with a 0.01μF polystyrene gave the best results.

I had the frequency response I wanted *(Fig. 2* shows the prototype's measurements); I had the efficiency I wanted; but could they handle power? Well, over the last four months I have used them with my system *(Table 1)* and pumped deafening music through them at over 120W/channel, without a ribbon failure.

The ribbons now worked very well, but I thought back-wave reflections caused some minor problems. At first I tried two layers of ⅛-inch felt over the steel backplate. This improved detail and reduced a very slight ringing sound. After looking further at some commercial systems, I decided to drill ½-inch holes to vent the ribbon *(Photo 3)*. This also allowed me to use a pair of output jacks from a discarded Hafler POOGE as ribbon input jacks. The ribbon then rests between two small washers held by two small nuts. I used some scrap plastic as an insulator, which also acts as a spacer to keep the ribbon frontplates apart. Eventually, I decided on one layer of felt, 16 by ½ by ⅛ inches, as damping material.

If you want to experiment with ribbons, I can offer some practical advice. At high volumes, ribbon widths much greater than ½ inch can produce a very audible and damaging "crinkle" sound, such as if you wad up a piece of tin foil.

Narrower pieces seem to be more fragile though they have more acceptable impedances. A ⅛-inch wide ribbon will run about 2Ω, all other things being equal. If you decide to try a taller ribbon, you may have trouble finding longer pieces of foil from a capacitor. Unless you have another source of ribbon material (sheets of aluminized Mylar, and so on), it may be better to stack two ribbons per side. Though I have not tried many other ribbon sources, I have yet to find one that works as well. Perhaps you will find a material that will give you a 4–8Ω impedance with equal or better efficiency than the foil found in some caps. If you do, please let me know.

THE SOUND. My design produces everything I had come to expect from a ribbon midrange/tweeter: clear, clean and dynamic sound with excellent imaging. My previous reference tweeter, the Dynaudio D-28, is a very good one, but does not bear comparison with the ribbons. There is however, one drawback. The ribbons, while having excellent horizontal dispersion, are quite directional vertically, and must be placed at ear level for optimum listening.

Another drawback, less important, is the somewhat "unprofessional" appearance of the ribbon itself. Actually, I can make the ribbon look quite good, but it requires a great deal of patience in cutting and smoothing the material. Since my system is usually in a state of flux *[pun intended?—Ed.]*, and the ribbon works as well if it looks wrinkled, I have rarely set up the ribbons to look their best.

Are these ribbons for you? Well, the most serious drawback for some people is the low impedance, but with tube power amps and high current transistor amps, it should be no problem. Flexibility is one of their better attributes. Want

Continued on page 58

TABLE I

TEST SYSTEM

AR Turntable	(Gary Galo's *TAA* mods) with acrylic platter and arm board.
Cartridge	Alchemist IA with Monster Cable tonearm interconnects
Conrad-Johnson PV-3 (POOGEd)	
GSI Dyna Mark IVs (modified)	

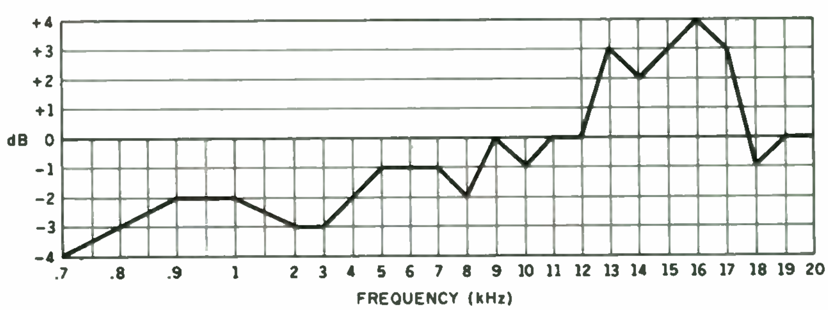

FIGURE 2: Frequency response, Keele near-field technique—1-20kHz, ±4dB; efficiency—93dB/W/M; moving mass—25mg; magnet mass—1.08kg.

KITS
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FILTERS & SPEAKER SAVER

KH-2: SPEAKER SAVER AND OUTPUT FAULT DETECTOR [3:77]. This basic two-channel kit includes board and all board-mounted components for control circuitry and power supply. It features turn-on and off protection and fast opto-coupler circuitry that prevents transients from damaging your system. The output fault detector has additional board-mounted components for speaker protection in case of amplifier failure. Each **$62**

KF-6: 30Hz RUMBLE FILTER. [4:75] This kit implements Walt Jung's 1975 design for a low frequency garbage filter. The filter knee is set to 30Hz. Roll-off below that knee is the 18dB/octave characteristic of its three pole design. Gain for the filter is unity (0dB) but can be simply adjusted for up to 12dB of gain. The reprint of Jung's article explores the use of the filter with other components in crossovers (see kits SBK-C1A, C1B, C1C). He shows how to obtain slopes of 6, 12 or 18dB in high and low pass filters. The kit contains all parts for building a two channel HPF including a board (3" x 3"), quad op amp IC, precision resistors and capacitors. Requires a bipolar supply of ± 15V, the KE-5 is suitable. Each **$28**

AIDS & TEST EQUIPMENT

KK-3: THE WARBLER OSCILLATOR [1:79]. This unit will produce a swept signal covering any ⅓-octave between 16Hz and 20kHz. The total harmonic distortion at the output is less than 1.5%. The output voltage is adjustable from 0 to 1V. When used with a microphone it is as effective as a pink noise source in evaluating speaker system performance. It also reveals the listening environment's effect on sound through reflection and absorption. The sweep rate is set at about 5Hz. The kit includes 3¼" x 3⅞" circuit board, transformer, all parts and article reprint. Each **$65**

KH-7: GLOECKLER PRECISION 101dB ATTENUATOR. [4:77] All switches, 1% metal film and 5% carbon film resistors to build prototype. Chassis, input/output jacks are not included. Each **$62**

KC-5: GLOECKLER 23-POSITION LEVEL CONTROL. [2:72] All metal film resistors, shorting rotary switch and two boards for a two-channel, 2dB per step attenuator. Choose 10k or 250kΩ. Each **$42**

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

KP-2: TWO TONE INTERMODULATION TEST FILTER. [1:82]. This filter is designed to isolate the two high frequency tones at an amplifier's input from low frequency intermodulation products present at the output. The high pass filter corners at 2kHz and rolls off at 24dB/octave. A 5kHz signal at the low pass input will be down at the output by 80dB. An article reprint detailing design and use is included with the kit. All parts are supplied including quad op amp IC, circuit board and precision resistors and capacitors. Each **$26**

SBK-D2 WITTENBREDER AUDIO PULSE GENERATOR. [SB 2:83] All parts, board, pots, power cord, switches 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 **$32**

CROSSOVERS

KC-4A: ELECTRONIC CROSSOVER, KIT A. [2:72] Single channel, two-way. All parts including C-4 board and LF351 ICs. Choose frequency of 60, 120, 240, 480, 960, 1920, 5k or 10k. KE-5 or KF-3 supplies are suitable. Each **$12**

KC-4B: ELECTRONIC CROSSOVER, KIT B. [2:72] Single channel, three-way. All parts including C-4 board & LF351 ICs. Choose two frequencies of 60, 120, 240, 480, 960, 1920, 5k or 10k. Each **$15**

KK-6L: WALDRON TUBE CROSSOVER LOW PASS: Single channel, 18dB/octave, Butterworth, [3:79] includes three-gang pot. Choose 1: 19-210; 43-465; 88-960; 190-2100; 430-4650; 880-9600; 1900-21,000 hertz. Each **$58**

KK-6H: WALDRON TUBE CROSSOVER HIGH PASS: Single channel, 18dB/octave, Butterworth, [3:79] includes three-gang pot. Please specify 1 of the frequencies in KK- 6L. No other can be supplied. Each **$60**

KK-7: WALDRON TUBE CROSSOVER POWER SUPPLY. [3:79] Includes board, transformer, fuse, semiconductors, line cord, capacitors to power four tube crossover boards (8 tubes), 1 stereo bi-amped circuit. Each **$100**

SBK-A1: LINKWITZ CROSSOVER/FILTER. [SB 4:80] Three-way crossover/filter/delay. 24dB/octave at 100Hz and 1.5kHz and 12dB/octave below 30Hz, with delayed woofer turn-on. Use the Sulzer supply KL-4A with KL-4B or KL-4C.
Per channel **$72** Two channels **$132** SBK Board only **$14**

SBK-CIA: JUNG ELECTRONIC TWO-WAY CROSSOVER. [SB 3:82] 30Hz filter with WJ-3 board & 4136 IC adapted as one channel crossover. Can be 6, 12 or 18dB/octave. Choose frequency of 60, 120, 250, 500, 1k, 2k, 5k or 10k. The KL-4A/KL-4B or KW-3 are suitable supplies. Each **$30**

SBK-C1B: THREE WAY, SINGLE CHANNEL CROSSOVER. [SB 3:82] Contains 2 each SBK-C1A. Choose high & low frequency. Each **$58**

SBK-C1C: TWO CHANNEL, COMMON BASS CROSSOVER. [SB 3:82] Contains two each SBK-C1A. Choose 1 frequency. Each **$62**

SBK-C2: BALLARD ACTIVE CROSSOVER. [SB 3,4:82] three-way crossover with variable phase correction for precise alignment. Kit includes PC board (5³/₈ x 9¹/₂"), precision resistors, polystyrene & polypropylene caps. Requires ± 15V DC power supply—not included. Can use KL-4A/KL-4B or KW-3. Two channel **$145**

SYSTEM ACCESSORIES

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 **$130** Two or more **$122**

KE-5: OLD COLONY POWER SUPPLY. Unregulated, ± 18V @ 55mA. Each **$20**

KF-3: GATELY REGULATED SUPPLY. ± 18V or ± 15V @ 100mA. Each **$48**

KL-4A: SULZER POWER SUPPLY REGULATOR. Each **$40**

KL-4B: SULZER DC RAW SUPPLY. ± 20V @ 300mA. Each **$42**

KH-8: MORREY SUPER BUFFER. [4:77] All parts, 1% metal film resistors, NE531 ICs, and PC board for two-channel output buffer. Each **$20**

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**

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**

KW-1: MAGNAVOX CD PLAYER MODIFICATION. Improves frequency response. Includes two Signetics NE5535s, 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-1: DISC STABILIZER. Set of 3 Sorbothane feet, 3 Tiptoes and Mod Squad's Disc Damper with 15 centering rings. Each Set **$75**

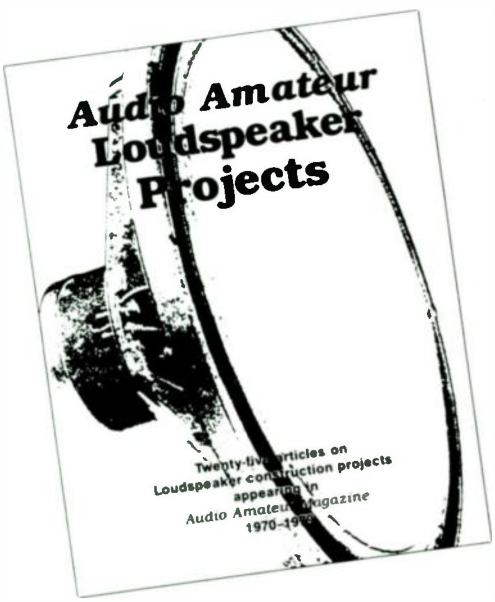
KX-2: POOGE-4 CD PLAYER MOD. [1:88, 2:88] Jung/Childress extensive rework of the Magnavox CDB 650. Each **$163**

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

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 *TAA* and *SB* 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.

Audio Amateur Loudspeaker Projects

Twenty-five articles on Loudspeaker construction projects appearing in ***Audio Amateur Magazine*** **1970–1979**



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Product Review

A Giant Leap: Beyond Thiele/Small

LEAP, *by Chris Strahm*, CNS Electronics, Box 42389, Portland, OR 97242, (503) 231-7247, runs on IBM PC, AT, XT; $149.

Reviewed by Vance Dickason

Virtually all the CAD programs for loudspeaker cabinet design, most of which have been reviewed in *Speaker Builder*, are basically calculator programs. They use the design equations from Thiele/Small second- and fourth-order models and grind out all the appropriate numbers. These programs also include graphic interfaces which allow viewing the predicted response on a log scale.

This predicted response, however, is not so much a picture of actual driver response in a given cabinet, as it is the filter-perfect projection of a second- or fourth-order network with different Q parameters. As such, these theoretically symmetrical driver rolloff pictures should be regarded as approximations of actual response, which give the user a fairly reasonable idea of what to expect out of a driver, but the actual situation definitely allows for much more sophisticated modeling. This brings us to the subject of LEAP (Loudspeaker Enclosure Analysis Program).

LEAP is as different from the above mentioned programs as a Wright Flyer is from an F-15. This program represents a substantial advance in low-end driver response modeling, and is an order of magnitude more accurate than the current, Thiele/Small industry standard. The Thiele-Small model, a major breakthrough for its time, ignores certain acoustic elements and treats others as fixed constants, unvarying with frequency. The real-life acoustic situation is forced into a usable, but simplified model of a second- or fourth-order electrical network. This allows many more professional designers and amateurs to quickly and easily come up with usable box designs, reasonably close to the predicted target response desired.

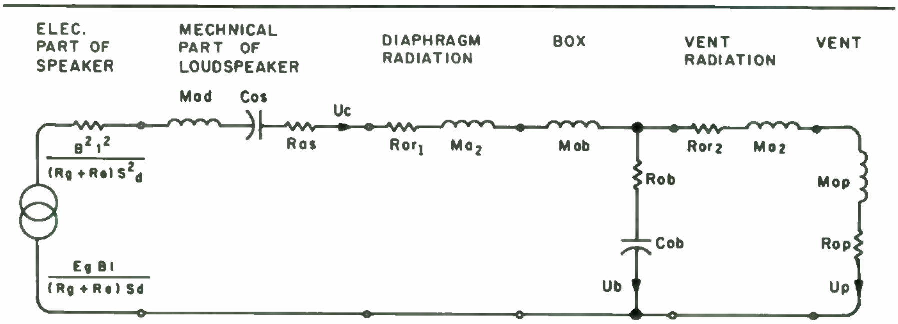


FIGURE 1: Driver/box model as originally presented by Thiele.

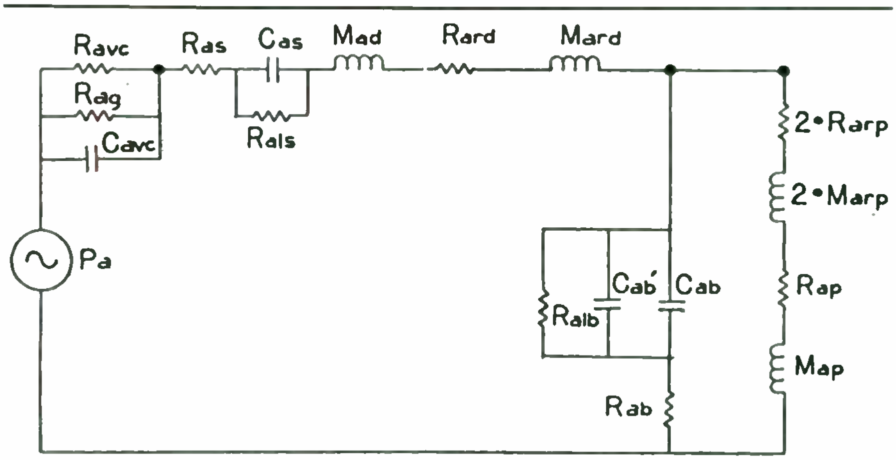


FIGURE 2: LEAP's basic acoustic model.[1]

LEAP however, is a full-scale acoustic model, taking elements into account which the Thiele/Small model removed, (and it's only fair to note these elements generally are not dramatically significant for single driver analysis), but also allows for the frequency dependent nature of many of the model's elements.

The real secret to LEAP's incredible ability to accurately model acoustic events has to do with some ingenious mathematical approximations of complex events which were carefully employed over relevant ranges of calculation. Otherwise, the kind of modeling in LEAP would literally take hours, not seconds, to compute.

The diagram in *Fig. 1* represents the driver/box model presented by Thiele in his 1961 paper, presented to the I.R.E. Radio and Electronic Engineering Convention in Sydney, Australia, and later, printed in the *JAES*, May, 1971. The same model is presented in Beranek's *Acoustics*, on page 241.

Figure 2 gives the basic acoustic model used by LEAP, minus the various mutual impedance circuits which model radiation impedance effects, caused by groups of speakers and ports used in an array.

1. Used by permission, *JAES*.

Comparing these two circuits, immediately you see the circuit topography of the LEAP model is somewhat different. It is beyond the scope of this review to explain the specific differences that exist, however, it might be helpful to point out the elements in the model which are treated as frequency dependent:

R_{avc}: represents the acoustical resistance of the voice coil. This element is normally thought to be constant with frequency, but it can be shown that counter electromotive force (EMF) voltage creates an effective increase in voice coil resistance that changes with frequency. Woofer motors with a higher BL tend to produce more change over a given frequency span than smaller motors.

R_{ard} and M_{ard}: these elements represent the radiation impedance of the diaphragm and also vary as a function of frequency.

R_{ab}: the resistive element usually measured as Q_l and strongly varies with frequency.

M_{arp} and R_{arp}: represent port radiation impedance and are also treated as frequency dependent.

R_{ap}: represents frequency dependent viscous losses through the port.

Again, the Thiele/Small model treats these variables as constants to produce a simplified, easily calculated approximation. It is interesting to note, many of the frequency dependent elements in the LEAP model were determined substantially by empirical means. As to the accuracy of this new model, *Fig. 3* shows the projected response (this was the earlier LEAP Version 2.1), sealed and vented, of a JBL 2225 woofer, while *Fig. 4* gives the actual measured response using outdoor, half-space sinewave sweep (the speaker pointed up, mounted in a pit, baffle at ground level). If you are interested in the details of the LEAP model, I recommend reading *Complete Analysis of Single and Multiple Loudspeaker Enclosures*, by Chris Strahm, AES Preprint #2419.

Figure 5 gives the complete LEAP model. The additional elements are necessary to consider the mutual impedance loading factors when using multiple ports, multiple drivers, and, in the case of sound reinforcement arrays, multiple cabinets. Of the four circuit elements which relate to the different mutual coupling effects, only the Speaker/Port Mutual (coupling) equations have been previously published. The speaker-to-port mutual coupling effect was recognized back in the 1920s; Bart Locanthi derived the equations for this element thirty years ago. The remaining mutual coupling equations are new, and to my knowledge, have never been published (actually they aren't being published now, but are part of the program code of LEAP). I can't stress enough what an incredibly important work and original program this is. Obviously, the ability to predict the response of multiple ports, drivers and cabinets also differentiates LEAP from the usual box design programs.

Besides utilizing a more complex model with frequency dependent elements, one more extremely important facet of LEAP is as a large signal simulator. As I stated before, all the previous Thiele/Small programs just calculate the usual small signal approximations. LEAP, however, is not a small signal calculation program, but rather will tell you just about everything you ever wanted to know about woofer operation in the real world, but were afraid to ask.

To explain the LEAP program, I'll walk you through the general procedures required for a cabinet design. Leap is divided into three basic operating modes, plus what is actually a second program called "Quick Cabinet." These operational modes are associated with three library functions: one for entering speaker parameters, another for entering cabinet parameters, and a design library for specifying the parameters for analysis. Functionally, a fourth mode is the actual analysis process performed on the information contained in the three libraries.

You start by entering the speaker parameters or by calling up one of the drivers loaded into one of the extensive driver libraries contained in LEAP. Fifteen parameters must be entered: Nominal Z, S_d, BL, R_{evc} (voice coil resistance), L_{evc} (voice coil inductance), V_{as}, C_{ms}, M_{ms}, F_i (driver resonance mounted on a baffle—LEAP calculates this for you), F_o (driver free resonance), Q_{ms}, Q_{es}, Q_{ts}, X_{max}, and P_{max} (driver power handling). The LEAP manual also includes some specific instructions on these parameters and measurement methods, but more on that later.

One critically important feature of LEAP is contained in the speaker parameter entry mode, and involves the cross-correlation of parameters as they are logged into this section of the program. As each parameter is entered, LEAP checks its integrity with other parameters and makes certain all 15 driver parameters properly interrelate. If you specify a certain BL, cone mass, and compliance, LEAP will immediately call your attention to an incorrect driver resonance or Q measurement.

You might be interested to know not all the specifications given by the various OEM manufacturers correlated properly, as initially entered into the LEAP data libraries. Amateurs are not the only ones who sometimes have trouble coming up with the right numbers. This certainly emphasizes we need to confirm driver specifications before proceeding with any design work.

The next step is to specify a cabinet design, which can come from several sources. Calling up the "Quick Cabinet" part of the program is easiest. Similar to calculator programs, it will give you a quick approximation of box design, but this complex optimizing routine doesn't just punch numbers into the usual Thiele/Small formulas. It will calculate volume for sealed enclosures, giving you box sizes for a Q_{tc} of 0.577, 0.7, or 1.0.

For vented designs, Quick Cabinet will

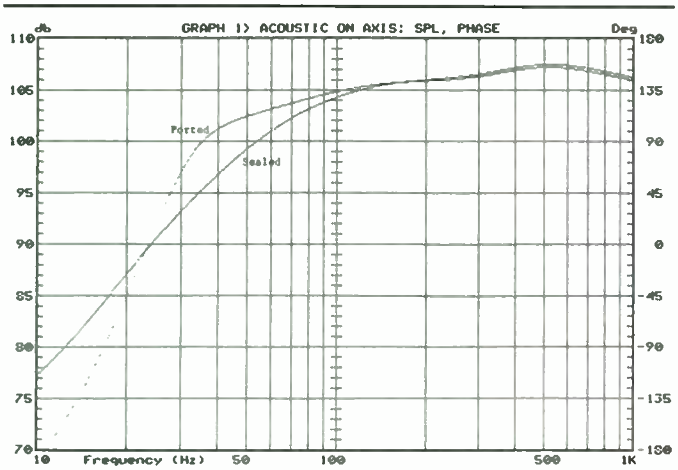


FIGURE 3: Earlier version's (LEAP 2.1) projected response, sealed and vented, for JBL 2225 woofer.

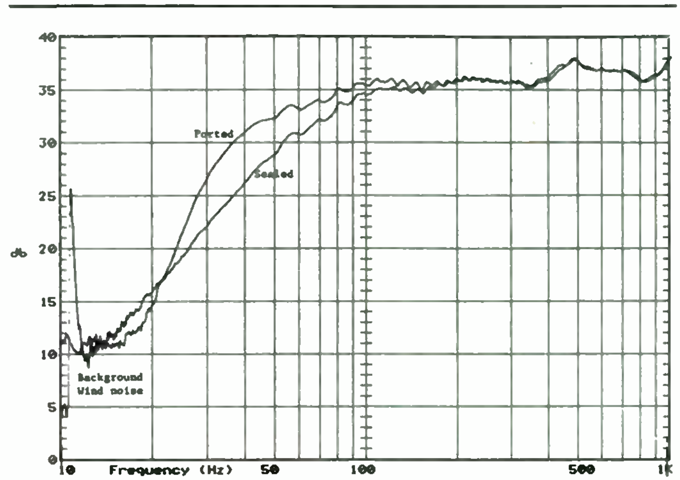


FIGURE 4: Measured response, 6.2 ft.^{3} box, JBL 2225 woofer.

give you box sizes for drivers with Q_{ts} less than 0.4 (overdamped) for three alignment families: Q_{B3}/Butterworth; Extended Bass Shelf/Butterworth; and Sub-Chebyshev/Bessel. For woofers with Q_{ts} over 0.4 (underdamped), it will give three box sizes for three other alignment families: Chebyshev/Butterworth; Enhanced Q/Butterworth; and Damped Chebyshev/Bessel. The overdamped Extended Bass Shelf alignment is a proprietary alignment only found in LEAP. Again, the Quick Cabinet designs are not derived from straight Thiele/Small equations, but are actually driver transfer function optimization routines.

Once you come up with a proposed cabinet size (from Quick Cabinet or some other Thiele/Small calculator program) enter the data into the cabinet library: cabinet dimensions; the number of drivers in the cabinet; whether or not the cabinet is ported and the number of ports; the size of the port or passive radiator data, including tuning frequency; any mutual coupling distances, such as speaker-to-speaker, port-to-speaker, port-to-port, or cabinet-to-cabinet; and the amount of filler material (LEAP has loss figures based on fiberglass only, so if you use some other type of material, you will have to empirically measure the result and come up with a percentage of fiberglass-fill equivalent).

In this section, and throughout the program, LEAP almost always calculates the answers and gives you a default figure to work with. In the case of cabinet design, if you enter volume, it will give you dimension defaults; with the port tuning frequency, it gives port dimensions.

After you enter both the speaker and the cabinet parameters, you are ready to enter analysis parameters in the design mode. Start by specifying your driver from the speaker library, and the box design you entered in the cabinet library. Next, indicate the number of cabinets (usually one for us home hi-fi people), flat or curved array (flat for a single speaker analysis), the simulated power to be applied, the amount of series resistance in the amp to speaker cable (and probably the crossover), voice coil temperature (set at 25°C, which is room temperature for a default), and the simulated distance the measuring microphone would be from the speaker.

So if you want 1W/1M, all you have to do is type it in. If you want to see what happens at 10W, or 20W, and show an increase in voice coil temperature from 25° to 75°, again, you simply type in the required parameters. Once you specify these parameters, LEAP automatically goes into analysis mode. This can take a few seconds, or somewhat longer, depending on the degree of precision you specify.

2. Used by permission *JAES*.

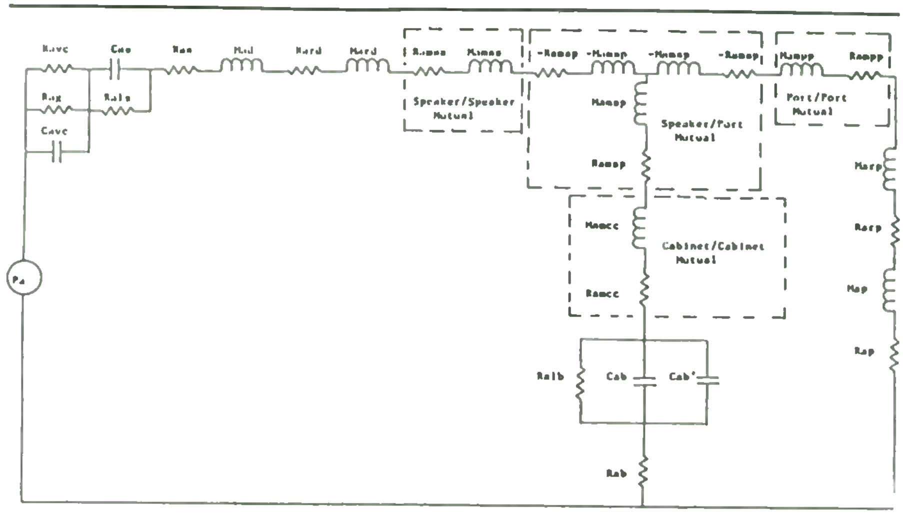


FIGURE 5: Complete LEAP model.[2]

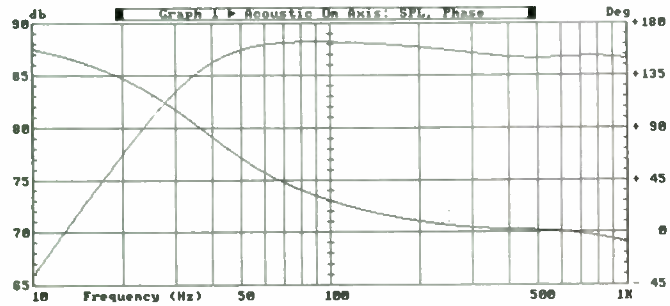


FIGURE 6: AC-10, 1W/88dB/25 °C, −3dB, phase angle 92.8.

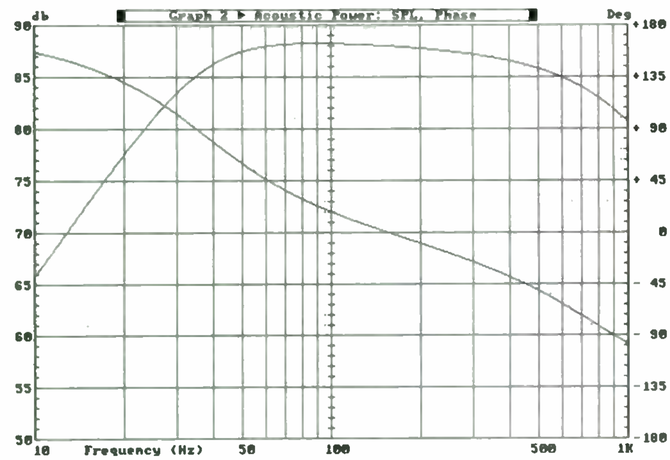


FIGURE 7: AC-10, 1W/88dB/25 °C, −3dB, phase angle 92.8.

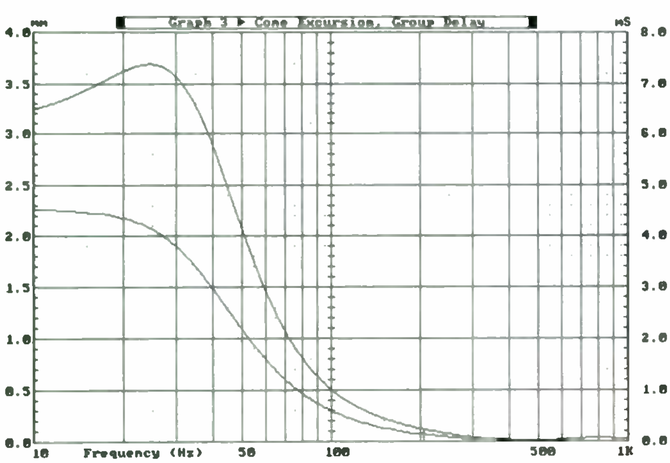


FIGURE 8: AC-10, 1W/88dB/25 °C, −3dB, phase angle 92.8.

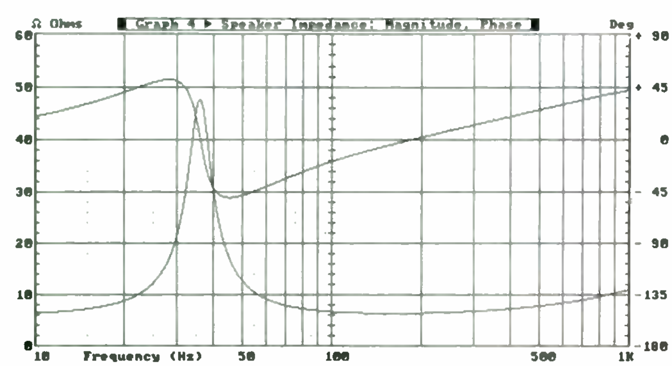


FIGURE 9: AC-10, 1W/88dB/25 °C, −3dB, phase angle 92.8.

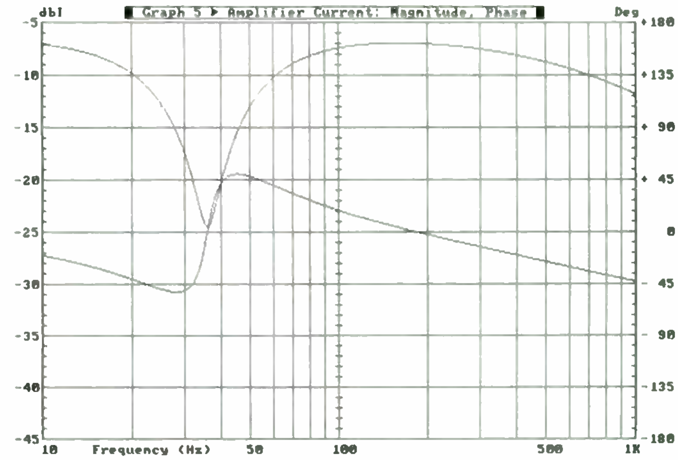


FIGURE 10: AC-10, 1W/88dB/25 °C, −3dB, phase angle 92.8.

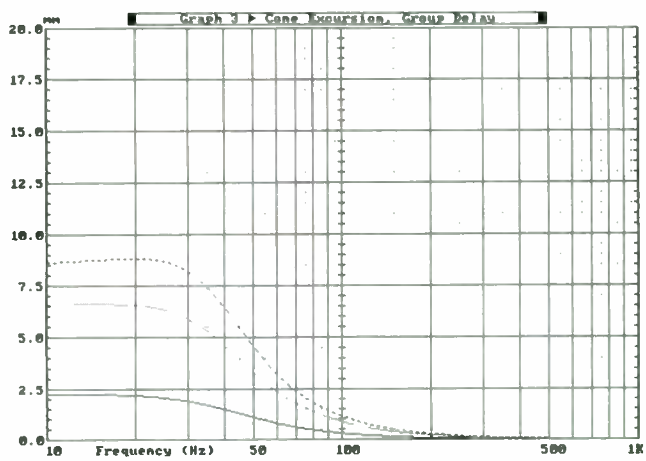


FIGURE 11: 1. AC-10, 1W/88dB/25 °C, −3dB, phase angle 92.8. 2. AC-10, 10W/98dB/50 °C, −3dB, phase angle 97.4 3. AC-10, 20W/100dB/75 °C, −3dB, phase angle 101.1.

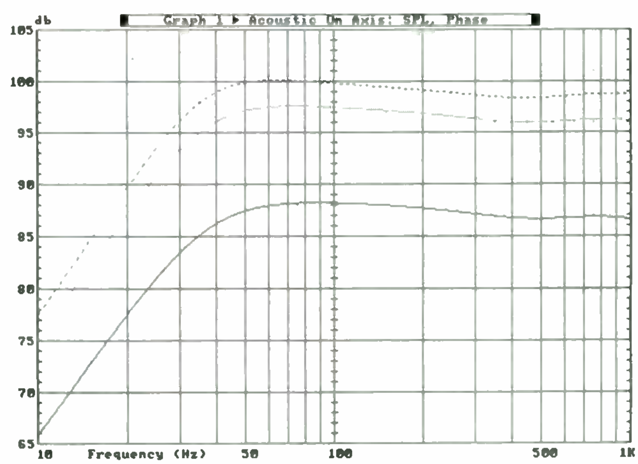


FIGURE 12: 1. AC-10, 1W/88dB/25 °C, −3dB, phase angle 92.8 2. AC-10, 10W/98dB/50 °C, −3dB, phase angle 97.4 3. AC-10, 20W/100dB/75 °C, −3dB, phase angle 101.1.

LEAP will either analyze each separate data point in its routine, or you can speed things up by telling it to skip every other point and interpolate. Actually, there are several different settings, each progressively faster than the full precision analysis. I have a 10MHz XT with a 10MHz 8087 math coprocessor, and the full precision mode takes only a little over a minute.

Once analysis is finished, you can then decide which of the 10 different curves you wish LEAP to display. On any given analysis, LEAP will display five pairs of data curves:

- Non-axis SPL/on axis acoustic phase;
- Acoustic power SPL/acoustic power phase;
- Cone excursion/group delay;
- Impedance magnitude/impedance phase;
- Current (admittance) magnitude/current (admittance) phase.

You can look at both curves on each graph, or turn one off. LEAP also allows you to compare designs, thus you can run a driver analysis using several different designs, and then select up to three to be displayed simultaneously on each of the five graphs (selected with the F keys of the keyboard).

To better appreciate what all this means, for an example, I have chosen the same driver I used in *Voice Coil's* User Report on LEAP (Part II, April 1988, Vol. 1, No. 6), the AC-10 woofer from Audio Concepts. *Figures 6–10* show the complete set of curves for a 3-ft.[3] cabinet, which is supposed to give a predicted Q_{tc} of 0.707, measured at 1W/1M, with a voice coil temperature of a nominal 25°C. If we measure the phase angle at the −3dB point on the SPL curve, which is easy to do with LEAP, we find we have about 92°, which is close to the expected 90° phase angle which occurs at the −3dB point in a theoretical Butterworth second-order filter. So far the program gives us pretty much what we would expect, and confirms the Thiele model.

The program also provides substantially more data than we usually get from a box program. Consider the Acoustic Power SPL curve in *Fig. 7*. Since LEAP takes into account the voice coil inductance, this curve begins to roll off about 400Hz. You can think of this as more or less the response at 45° off-axis, taken in a room situation. Such information is useful in determining desirable crossover points.

Suppose you want to know at what point excessive driver excursion will occur, the linear limit for this woofer being 7mm. All you must do is run successive analysis on the same speaker and cabinet at progressively increasing power input levels and voice coil temperatures. LEAP's file handling functions make this chore a snap. You simply copy the design file, at 1W, to another design location, change the input power level and temperature, and rerun the analysis. I did this at 10W, 50°C, and 20W, 75°C (I estimated temperature increases, which in this case are probably conservative). Once the three operations are completed, you can view all three designs simultaneously, as well as alter the graph to display only the cone excursion curves. The result is shown in *Fig. 11*.

You can easily see this driver is getting close to exceeding is linear capability at 30Hz or so, with a 20W input developing about 100dB of sound output at 1M. This is actually quite good for a single driver, and since we know the ear isn't particularly sensitive to distortion at low frequencies, this woofer would probably sound just fine up to 105dB, depending on its crossover point. If we look at the corresponding SPL curves for all three power levels, as shown in *Fig. 12*, we can see the rolloff gets progressively steeper as the power level and the voice coil temperature increase, indicating how driver damping is affected dynamically in a large signal situation. Notice the phase at the −3dB points is increasing, with increased SPL.

If this wasn't acceptable, we could try using two woofers, and see what happens.

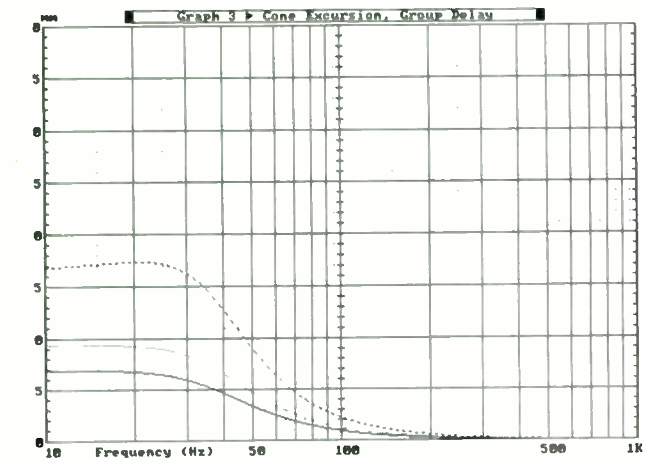


FIGURE 13: 5. AC-10 (2), 2.5W/98dB/35°C, −3dB, phase angle 95.7 6. AC-10 (2), 5W/100dB/50°C, −3dB, phase angle 97.5 8. AC-10 (2), 20W/106dB/75°C, −3dB, phase angle 102.

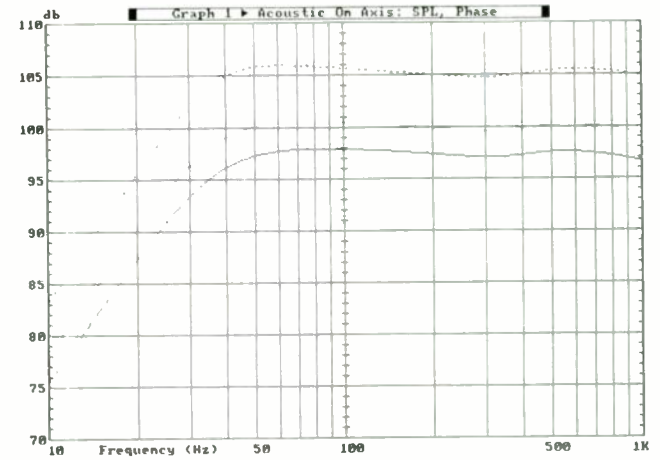


FIGURE 14: 5. AC-10 (2), 2.5W/98dB/35°C, −3dB, phase angle 95.7 6. AC-10 (2), 5W/100dB/50°C, −3dB, phase angle 97.5 8. AC-10 (2), 20W/106dB/75°C, −3dB, phase angle 102.

Figures 13 and *14* illustrate the results using two AC-10s in a single cabinet, twice the size of the original cabinet, but with one-fourth the power input (since two drivers give us a 6dB increase in sensitivity) and a correspondingly lower voice coil temperature. As you would expect, the required excursion at 100dB SPL is cut in half. Now the system begins to go nonlinear at about 106dB, and because the voice coil temperature will be lower, there would also be fewer changes in damping. Overlaying several curves at one time opens up many possibilities, such as matching woofer and midbass driver efficiencies for crossover frequencies between 100Hz and 500Hz.

LEAP comes with an excellent 185-page manual, which includes a tutorial, general explanation of the various program functions, and an extremely useful appendix, including a most interesting section containing recommended test procedures for determining driver parameters.

The author prefers the constant-voltage test method of determining impedance (actually the admittance), rather than the more commonly used constant-current method. He states several reasons for this, such as the ability to test at any drive level and the advantage of having the amplifier exert full damping control over the driver during testing.

Mr. Strahm's method for determining driver Q_{ts} also differs from the norm. Most sources, starting with Thiele, suggest measuring the −3dB points of the driver impedance above and below the resonance frequency. This method of measuring two low-frequency points so close together is prone to error magnification, and probably accounts, more than anything else, for the number of OEM manufacturers with noncorrelating parameters.

Success requires using good quality test equipment to measure the frequencies at extremely low voltages. The method described in the Appendix calculates Q_{ms} from this equation:

$$Q_{ms} = \frac{R_{es}}{6.283(BL^2)(C_{ms})(f_s)}$$

If you have never measured the BL (product) for a woofer, the manual covers that procedure also. It basically involves connecting a DC power supply to the driver terminals, placing a known weight on the woofer cone, and increasing the power supply voltage until the cone returns to its rest position. Measuring the amount of current used:

$$BL = \frac{g \times M}{i} \text{ tesla meters}$$

where

g = the force exerted by gravity (9.8 M/S^2)
M = the mass added to the cone (kg)
i = current (A)

After using both methods, I think this procedure is probably a lot easier for most people to perform, and as the manual points out, less likely to produce errors.

As if all of this isn't enough, you can expect a 4.0 version of LEAP, perhaps a year from now. This version will include nonlinear modeling of woofer behavior, which allows accurate prediction of driver operation beyond X_{max}!

LEAP is a powerful program which can greatly enhance and speed up box design work. It is a professional tool in every sense of the word, but is fortunately priced well within the means of many amateur builders and designers. Even more important, as software programs go, LEAP is laid out in an extremely logical manner, is straightforward, relatively easy to become familiar with, and fun to use. Compared to other software I have used, it is easier to learn than Microsoft Word, and as intuitive as Ventura Publisher.

LEAP's customer list now includes Polk Audio, Harman Motive, Design Acoustics, Acoustic Research, JBL, Scan Speak, Focal, OHM Acoustics, Cabasse, GNP, Marantz, TAD/Pioneer, RAMSA, Turbosound Ltd., Waldom, Solen Inc., and Audio Precision. I will be making extensive use of LEAP in my next book of construction projects, which I am currently working on, and I certainly give this product my highest possible recommendation.

High-Grade Fluff

Reviewed by John Cockroft

I recently tested a new, synthetic, long-fiber, sound absorbent material, which has very good qualities. I would like to share my impressions of Acousta-Stuf, marketed by Mahogany Sound's Larry Sharp.

First, I checked the fiber, pulling out individual strands over six inches long. In addition, each fiber is crimped during manufacturing at approximately 1/16-inch intervals, along its entire length. This ensures the fibers interlock with great consistency, and provides a regularity in density I have never seen in similar material. Even when pulled apart, Acousta-Stuf tends to remain homogenous, without holes and thin spots. In the four-pound sample I received, no hard or bunched up pieces had to be pulled apart, as is common with department store types of pillow stuffing.

The fibers lie in a natural pattern that would apparently give excellent frictional damping. The effort required to pull the material apart attests well to its interlocking ability.

I have been working on the Freeline modification, which uses a 4-inch woofer in a variant transmission line, open at both ends. This made it easy to remove the stuffing and replace it with Acousta-Stuf. After listening, I found the sound to be extremely smooth and well-extended in the bass region and well damped with no audible resonances.

Acousta-Stuf is a superior product, both sonically and mechanically and much easier to prepare for line insertion than any of the pillow stuffing grades I have encountered. Its density in the mass is much more consistent. The fibers are longer than most wools and there should be no problems with the material matting down in the line, or enclosure. In other words, no fishnets, dowels or other prosthetics will be required, to ensure the material won't sag. It is not a moth food, either.

I can't imagine other materials giving better sound quality. My little Freeline gives a really excellent account of itself, using Acousta-Stuf, at most reasonable levels, on the following difficult CD recording passages:

- Stravinsky, *The Firebird*, Nimbus Records NI5087. The crescendo at the start of section 5 comes across with tremendous solidity and great musicality in one of the most startling demonstrations of a 2¾-inch cone I have ever heard.
- Franck, *Encores a la Francaise*, Piece Heroique, the finale (section 4), Telarc CD-80104. Featuring Michael Murray at the organ, I experienced great solidity and deep bass extension and no sensation of artificial resonance.
- Poulenc: Organ Concerto in G minor, finale (same disc as above). Solid sense of great mass.

The above sensations are created with a single speaker (mono), as I have only one prototype of the Freeline. Even so, I have not heard better performances of the three passages with any other speaker. At least part of that I would attribute to Acousta-Stuf.

System designers and builders owe a great deal to individuals like Mr. Sharp, who see value in a product not readily available and have the incentive to make it available to others.

I am sure Acousta-Stuf would work very well in sealed boxes and with aperiodic systems as well. I prefer a more reactive material, such as fiberglass, for use in vented systems. However, this material damps the midrange resonances adequately while not impeding the action of the vent, as a more resisitive material might tend to do.

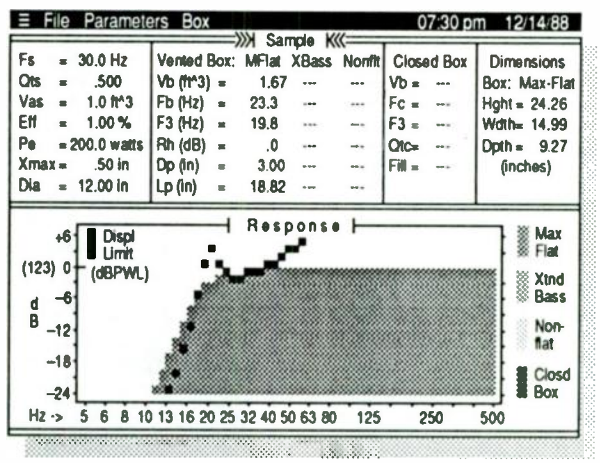
Mahogany Sound
2430 Schillinger's Rd.
Box 488
Mobile, AL 36695

(Acousta-Stuf, $7/lb.)

A LETTER

Asking For Help Without A Self Addressed Stamped Envelope Goes HERE!!

PLEASE!!



Craftsman's Corner

Three-Way Pyramid Design

My interest in loudspeakers is completely as an amateur. My opinions about loudspeaker design, materials and driver units are based on my experiences. They will not necessarily coincide with others' or manufacturers' opinions. I do not wish to offend anyone, only to explain my adventure.

I have been interested in speaker design for about 15 years and during this time I have owned speakers by Marantz, Monitor Audio, KEF and Spendor. I've also designed and built several pairs. Most recently, in 1976, I built a three-way, floor standing design, reflex loaded. The 75-liter cabinets were constructed from reinforced concrete, with panel walls approximately 1½" thick. The system had a big, powerful sound, good clarity and plenty of bass. I was delighted with them and immediately sold my Spendor BC1s.

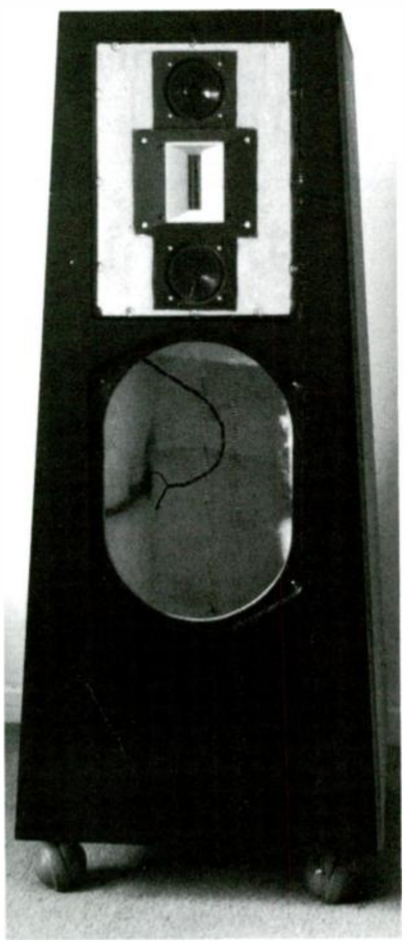
I enjoyed the music my speakers produced for nine years. No others sounded better to my ears. However, I gradually decided it was time to build another pair to incorporate recent design and technology.

My latest design evolved after much thought, listening and testing. I have adequate test equipment, including an oscilloscope, but I think ears are easily the best instruments. I've also read extensively on speaker design.

My aim was simply to build the best speaker that I could, and frankly, I have not spared time, effort or expense to get the results I wanted.

The following is a brief rundown of my system, with my reasoning for the methods I used:

- Size: 36"H by 15"W at bottom, tapering to 11"; depth, 18" at bottom, tapering to 15".
- Cabinet shape: A pyramid with a flat, removable top to avoid parallel surfaces and to conceal a sealed recess containing the crossover and mid/treble attenuators.
- Cabinet: Made from 1-inch MDF board, a heavy, dense material that doesn't have pits or voids and cuts beautifully. I braced the walls with mahogany struts and lined the inside with bituminous felt pads to further damp panel resonances. I stuffed the cabinet with BAF wadding. There is a separate enclosure within the main cabinet to house the mid and treble units.
- Bass Loading: I prefer a sealed box. I've yet to hear a reflex system that didn't sound a bit woolly in the bass. Reflex is okay for getting more bass from small speakers, but is not in the same league compared to quick bass transients of fairly large, sealed-box speakers operating at low Q.



- Total Q of bass unit/cabinet: 0.6 gives quick transient response with virtually no overhang. Bass drops off gradually instead of dropping like a stone, as in high Q designs.
- Bass drivers: the IMF B128/20 has the identical shape and is designed to replace the KEF B139 unit in IMF speakers, with excellent transient response and good power handling. I've tried both and prefer the IMFs.
- Mid-drivers: I paralleled two Jordan Modules to bring the output up to the bass driver's level and to help power handling. It is the best unit I've tried. I don't recommend it as a full-range driver because it has a response dip at 8kHz and a peak at 13kHz. My tests basically agree with Mr. Lampton's findings in *SB* 4/82 ("A Three-Way Corner Loudspeaker System"). I decided to use a larger enclosure for the Jordans than the manufacturer recommends, as it appears to improve the sound. I arranged the units in the D'Appolito configuration.
- Treble: The JVC HSW 1101-01A is simply the best treble unit I have tried. It has great definition and attack; again I agree with Mr. Lampton.
- Crossover: I chose a three-way 18dB/octave, per Bob Bullock's article (*SB* 2/85). I used heavy-duty ferrite core chokes for large mH values and air cores for small values; polypropylene capacitors for small values and nonpolar electrolytics bypassed with polypropylenes for larger values. The bass unit has a Zobel and the midrange has a trap. I use a switch to attenuate the mid and treble units. Crossover frequencies are 380Hz and 4750Hz. I prefer crossovers with fairly high cut-off rates; a bass unit crossing over at 500Hz at 6dB/octave, for example, still gives an output quite substantial at over 2kHz, far outside the useful range of a 10" or 12" driver.
- Speaker position: I placed these floor standing speakers away from the walls.
- General: The top cover will be assembled last. I plan to veneer the cabinets and fit a trim and front cloth grille. I'm also going to use a felt diffraction ring around the mid and treble units.

To describe the sound of my speakers is difficult; I will try to be objective. My design goal was to produce a speaker with an even frequency response with good clarity and attack. I believe my system is a success, and has great definition. I can now hear things on my records that were completely missing before. I can easily notice differences in turntables, tonearms, cartridges and amplifiers, for example.

They will not suit everyone. If you like a warm, mellow sound, with plenty of overblown bass, they are not for you. On the other hand, if you like to hear what's really on your records, this design will help you achieve that end.

I recommend these speakers be driven with a good quality amplifier. My system consists of a Heybrook TT2 turntable with separate power supply, Rega RB300 arm, Linn Karma cartridge, Naim NAP250 amplifier with Hi-Cap power supply and a NAC preamp.

A.M. Smith
Edgware, Middlesex HA8 6QB
England

SB Mailbox

MTM CORRECTION

I noticed an error in *SB* 2/88; Mr. Levreault's formula for R2 (p. 11) should be $(R_0^2/R_1) - R_0$, which I use to find a resistor value to parallel with another to arrive at a new desired lower value.

Steve Hauser
Iowa City, IA 52240

John Levreault replies:

Yes, the R2 equation has a typographical error, and should read as you suggest. Such is the difficulty of incompatible word processors between yours truly and our esteemed editorial staff.

JORDAN CABINETS: UPDATES

In Dave Davenport's "Kit Report" (*SB* 2/88) on our EJ Jordan HRM, he stated the cabinet work was a product of Princeton Acoustics, an affiliate of EJJ USA. While in a sense this is true, the cabinets are manufactured under our Option Audio company auspices. I may have misled Mr. Davenport on this matter, and although a minor distinction, we are trying to establish this company which does our cabinet work, which, by the way, is owned and operated by EJ Jordan USA.

Kirk Neal
EJ Jordan USA

TIPS FOR TIPS

Mr. Levreault suggests using Tip Toes, or the equivalent, to decouple his well thought out MTM array (*SB* 2/88) from woofer cabinet vibrations.

I think this approach will not decouple the enclosures, but will *increase* the energy transfer. To isolate the enclosures, I would use a compliant suspension, such as a sheet of foam rubber.

Tip Toes reduce the surface area of contact and increase the contact pressure at these points, and prevent *relative* motion between surfaces. Normally this prevents enclosure motion relative to the floor, especially when the speaker rests on a compliant surface, such as a rug. With a turntable, the spikes prevent the chassis from being excited by air borne vibrations, although it *increases* sensitivity to floor vibrations.

With your MTM array, you could couple the enclosure to a fixed surface to prevent the midrange drivers from causing the MTM enclosure to vibrate. Since the *edges* of the woofer enclosure are ideally vibration nulls, using spikes between the MTM enclosure and the woofer cabinet's top surface may be effective, but for different reasons than you stated. Large enclosures do not tend to be ideally stiff, and vibrations may still be transmitted to the MTM enclosure. Mass loading the enclosure with bricks should help by reducing the amplitude of vibrations.

Incidentally, in place of spiked points I use small hex nuts, which still reduce the contact surface area considerably, and don't mar the supporting surface. Spikes are useful for penetrating rugs.

Ralph Gonzalez
Philadelphia, PA 19143

John Levreault replies:

Although I tend to agree with your analysis, I disagree with your conclusions. The issue here is energy transfer. Let me explain.

Theoretically, the transfer of mechanical energy between two infinitely massive bodies will be zero, although they contact at a single point. In loudspeaker system design, we need to combine isolation with support, so we must call on this physical principle to help us.

For example, a woofer system must sit on a rigid structure so acoustic energy from the woofer system is not converted to mechanical energy in the structure. In real terms, we don't want the structure to vibrate with the speaker, which would smear the sound and distort the recreated soundfield. If a woofer is sitting on a carpeted floor, the carpet is the support structure, and the bass response of the system is very mushy sounding.

Placing the woofer on "tip-toes" or spikes solves this problem. The woofer has the support of the floor without cutting a hole in the carpet. Unfortunately, energy from the support can couple into the speaker. Such is life. If we bolt a woofer system to the concrete basement floor, a passing train can make the woofer wobble, but this wobble is asynchronous to the information being passed by the woofer, so we can live with it. Footfalls on a wooden floor can indeed create problems for a turntable even though it has been "tip-toed."

The MTM array has similar requirements and problems. The array must sit on a massive rigid enclosure. I chose the woofer, fully aware of the compromises you point out. I placed the spikes as close to the edges of the supporting woofer as possible to achieve the added benefit of the vibration nulls of the woofer enclosure. However, as I mentioned above, energy transfer is the issue. I approximated that mechanical energy in the woofer enclosure could be decoupled from the MTM array by increasing the mass of the array with bricks.

I'm open to suggestions for improving this system, but foam rubber or any other compliant materials is unacceptable to my ears. Hex nuts are better than foam rubber, but spikes are simple and effective.

I believe my system offers the following advantages:

1. A heavy and rigid woofer system mounted on a concrete floor with spikes, thus avoiding acoustical modulation of the woofer, although I acknowledge the possibility of mechanical energy being coupled from the woofer drivers themselves into the enclosure and from the floor into the woofer. (Nothing helps like a concrete basement floor!) The woofer won't sway in a self-induced breeze.
2. A relatively massive and adequately rigid high frequency array "tip-toed" to the rigid and more massive woofer system. Mechanical vibration from the woofer enclosure has been decoupled according to the above principles, although the MTM array must deal with its own spurious driver induced vibrations.

Consider what happens when a cannon shot is "played" through a woofer system sitting on a carpeted floor or a sheet of foam fubber on a concrete floor. The cone will lurch forward (or backward depending on the phase), and the air resists, creating a force that causes the woofer enclosure to lean back. As the support material attempts to restore the rest position of the woofer enclosure, the system wobbles, depending on the mechanical damping properties of the support material. The dynamic impact of the "musical" information will be smeared as will any information following the original transient event. I cannot live with this kind of sound.

PADS, DIPS AND BRACES

My congratulations to the *SB* staff for such an attractive and professional publication. I have several comments on the recent articles.

Mr. Leverault's array (*SB* 2/88), with the tweeter padded, will exhibit the same voltage sensitivity as a single D-76: 88dB. Has he considered wiring the D-76s in parallel, and using a padded 4Ω D-28? That would offer a voltage sensitivity of 94dB. How do his tip toes "mechanically isolate the woofer from the mid/tweeter," when he described using them to "firmly anchor the cabinet on a solid surface," in *SB* 2/87?

Mr. Hoffman, in issue 1/88, showed a response dip in the in the Radio Shack 4-inch woofer. I tested four units (40-1022) from three different stores, and they exhibit a severe dip (about 8dB) between 1.5 and 6kHz. Have Messrs. Cockroft and Weems, who have used the units, noticed such behavior also. I use mine in a D'Appolito configuration, wired in parallel. To correct the dip, I use a special low-pass filter (0.07μH and 28μF) and leave the driver impedance unequalized.

I think Mr. Sink should have noted in his "Primer" (*SB* 2/88) that by adding mass to a cone, the frequency extension occurs at the low end, and with some sacrifice at the high end. Maybe a future article could cover the interrelationship of voice coil diameter, magnet weight, sensitivity, and so on, and perhaps, how to spot phony Thiele/Small parameters.

Mr. Muxlow's 2/88 article, "Loudspeaker Cabinets," was a welcome addition, especially his many useful references. I can add two more, both by Maarten Vet, for readers to pursue: "Natural Frequencies of Thin Rectangular Plates," *Machine Design*, June 10, 1965; and "Thin Plate Natural Frequencies," *Machine Design*, June 9, 1966.

Mr. Muxlow says increasing the thickness of an enclosure adds mass, which pushes up the resonant frequencies. This is inaccurate; added mass by itself would force them lower. It's actually because the increased stiffness of a thicker panel more than overcomes the effect of added mass. By the way, I can see why some builders coat the interior walls with tar, although it adds weight without increasing stiffness, it also lowers the Q of the resonant frequencies. What would adding sand to the mixture do?

The tests in Reference 20 do not show cross-bracing to be particularly effective. I believe the result was colored by the very thin specimens used in the tests. I have found if the braces of opposing panels are cross-connected by a generously sized brace, the "knuckle rap" test indicates a remarkable improvement. I rationalize that panels present their weakest side to the deflecting forces, while cross-braces present their strongest. To resist the applied force, a panel bends, however a cross-brace must compress or stretch. Because a panel is weaker in its direction of loading than the cross-brace, its deflection is much greater. Since their deflection-resisting characteristics appear in parallel, those of the cross-brace will dominate, deflection will decrease and resonant frequency will increase. The article might have included that cabinet proportions, optimally selected, will present a pleasing appearance while spreading the resonant frequencies over a broad range. Weems quotes Thiele as recommending a width/depth/height ratio of 1/0.6/1.6.

David J. Meraner
Scotia, NY 12302

John Levreault replies:

I hope my reply to Mr. Gonzalez' letter adequately addresses your concerns over the use of spikes. I'm no mechanical engineer (although my degree is in physics) but I do know what sounds good. The above rationalization seems to make sense and I'm not trying to "tip-toe" around the issue. (Sorry about that, but I couldn't resist.) Any comments?

I chose to wire the drivers in series rather than in parallel for two reasons. First, the increased sensitivity of the midranges would require that I attenuate their drive by about 3dB to line up with the now less-efficient tweeter. I prefer to attenuate the highest frequency unit(s) so I can use lower power, high quality resistors. In any case, the array efficiency is determined by the lower efficiency driver, so paralleling midrange drivers would get you 91dB. Parallel midranges might be useful in a triamped system.

My second reason is more subtle. The lower load current demands of a high impedance system places proportionately smaller demands on power amplifier power supplies. I firmly believe the power supply is the cause of the significant differences I hear in audio amplifiers, both tube and solid state. Besides, my 100W MTM amplifier is more than adequate for full dynamic range, and it runs cooler too, although personally I negate this possible advantage by cranking up the bias current to run closer to Class A.

MTM SPACING

Mr. Levreault's recent MTM article was of great interest since I have built a similar unit. I wonder how the author determined the spacing between the units. I place mine as close together as possible to minimize phase interference.

I'm also curious why he mounted the midrange drivers in series. It seems that if they were mounted in parallel, he would require far less attenuation for the tweeter. This of course, would be a function of the amplifier and its relationship between power and impedance.

David Grant Willemain
Towson, MD 21204

John Levreault replies:

I determined the spacing between drivers exactly as you suggest: I placed them as close as possible to minimize diffraction, although this is less of a problem with the MTM configuration.

My rationale for series-connecting the midrange drivers is explained in my previous reply to Mr. Meraner.

ISOBARIKS

Your magazine is factual, noncommercial and objective. I consider myself a beginner, though I have been experimenting for about five years, and *SB* is just the ticket for technical information, which forces the beginner to think.

I am primarily interested in Isobarik designs. The idea of lower f_3 and reduced V_b all in the same package, as well as higher power handling and reduced driver distortion (push-pull configuratuion) is attractive. It's hard to believe such a pot of gold exists. Why don't I see more of these designs produced commercially? The design basics aren't that new. The only obvious drawback is a little more expense for the drivers, but since when did that stop anyone interested in sound improvements?

John Cockroft's Demonstrator (*SB* 2/87) and Little Dancer (*SB* 3/86) both favor the Isobarik approach. I would like more information on driver selection, tuning and alignment techniques, and formulas.

Rusty Lewellen
San Angelo, TX 76901

John Cockroft replies:

Isobarik, or compound speakers were discussed considerably in *SB's Mailbox* section in 1985-86. Some readers were disappointed the concept didn't offer a free lunch. I believe compound drivers are valid as tools toward a design. The more choices a designer has, the better possibility of meeting the required parameters. No concept is intrinsically better than any other, except in an explicit situation for an explicit driver, with definite desires and constraints.

With the compound speakers I designed and constructed, the vented configuration (the Demonstrator) was the most successful in lowering f_3. Part of the space savings realized by the reduced V_{as} is countered by the additional space required to house two drivers. But even so, the Demonstrator managed to end up slightly smaller than a similar simple speaker of the same calculated performance. In some cases, slightly smaller could be crucial; in others it may not matter.

In spite of the additional construction cost, several such speakers have been on the market. Jamo, Focal, Linn, and Audio Concepts are a few manufacturers who come to mind. Some are full-range systems, others are subwoofers. Compound drivers can work well in either situation.

The quality of the sound is not merely the result of the configuration chosen. It is the result of care taken in the many details and decisions required to bring a system from an idea to a finished product, incorporating as much of the original idea as the designer is able. The trick is to find the most proper environment for a given driver or speaker from the myriad of possiblities.

Outside of the drivers I use, I have little experience with many recent units, I prefer to use a few with the qualities I like that I can become familiar with through constant use, to the point that I can exploit them in many situations. There is a lot of hype in the commercial game, but no miracle drivers; all are electric motors that run within the contraints of their parameters, according to the laws of physics (not PR deparments). I'm amazed how some manufacturers extol the virtues of "tight bass" when drivers have Q_{ts} of 0.8 or higher, and often accompanied by claims of "powerful motors." How can such powerful motors be so poorly damped?

This is a good time to mention that the little Radio

Shack 40-1022, one of the drivers I have depended on for several years and used in many designs, is no longer available. This exceptional and consistent 4-inch woofer has been replaced with a driver designated 40-1022A. Unfortunately, Radio Shack has chosen to pretend it is the same driver. In fact, they include the same spec sheet with the new drivers, except they have overprinted an "A." This is unscrupulous, as the specs claim F_{sa}, 55Hz, and Q_{ts}, 0.35; but the new 40-1022A unit I measured had F_{sa}, 73Hz, and Q_{ts}, 0.74. I had measured 10 or 12 old 40-1022s, which were all within 10% or so, of the published specs.

The discrepancy rules out using the 40-1022A for my earlier compound woofer projects. However, if you modified a new unit by adding a weight to the cone, according to my Octaline article, it might well work out. With my 40-1022A, I tried adding about 5g; since the Q_{ts} is higher I was a bit conservative. It worked surprisingly well in my small lines.

The old drivers were made in Japan and came in a green box. The 40-1022A is made in Korea, comes in a blue box, and is not suitable for my designs to sound as I intended. Please take this as a requiem and a caveat. There still may be a few old units left on the shelves; I found none in my area.

Apparently Radio Shack didn't know what they had in the 40-1022, or didn't care. I'm having no luck finding a suitable substitute woofer for the Mini-Dancer, the Demonstrator and the Octaline, as I presented them. Radio Shack has also discontinued the 40-1376 tweeter, but the Audax TW74A is the same unit and available from many dealers.

ADDENDUM: DRIVER DESIGN

I congratulate Perry Sink for his informative article on driver basics (*SB* 2/88), but several technical inaccuracies need correcting.

First, I believe the magnetic flux in the air gap is one of the largest controllers of loudspeaker Q, since it determines the amount of electromagnetic damping. As Mr. Sink points out, the magnet strength also controls midband efficiency. If too strong a magnetic field is used, the Q becomes very small. Low-frequency response can also suffer because of overdamping. In fact, a driver with a stronger magnet, but all other parameters equal, will exhibit a *smaller* rise in impedance at resonance than will a speaker with a weaker magnet. (See Thiele's equation 72, "Loudspeakers in Vented Boxes," Part II, *JAES*, June 1971.)

Notice I consistently refer to magnet strength, not magnet size. The gap width and the magnetic saturation of the steel used for the top plate, back plate and pole piece, as well as the physical dimensions of these magnetic structure members, limit the maximum field strength; also, an optimum thickness-to-surface-area ratio exists, for ceramic magnets. Therefore, magnet size is a poor guide to magnet strength. In fact, some manufacturers are sufficiently unscrupulous to use magnets so large that the speaker cannot take advantage of them. Thus, they advertise a driver with a one-inch voice coil and a 30-oz. magnet, at a higher price, even though it performs no differently than with a 16-oz. magnet. I applaud companies such as Polydax, that list gap flux density rather than magnet weight, for truth in advertising.

Second, note that the Q of a speaker is the ratio of the resonant frequency to the -3dB bandwidth of the efficiency curve centered about F_o. Mr. Sink's statement, "a Q_t of 0.31 at 25Hz," is meaningless, since the Q is a function of speaker mass, stiffness and mechanical and electrical damping, and only refers to the shape of the resonance curve. Therefore, "Q at a certain frequency" does not exist. Also, trying to convert Q into relative response in decibels is very much like trying to convert the shape of a mountain top into the altitude part-way up the slope.

Third, the force factor or BL (product) is just the length of voice coil wire in the gap multiplied by the gap flux density. Therefore, BL *always* increases when more wire is added to the coil, provided the extra wire is within the gap. Here, though, is the rub (pun intended), because the gap width must be increased to accomodate more layers, somewhat increasing fringing and therefore reducing gap flux. The addition of wire to make the voice coil overhang the gap does not increase BL. However, it does increase inductance, which, as Mr. Sink points out, reduces high-frequency output. Voice-coil wire overhanging the gap also results in part of the voice coil's magnetic field being outside the gap, and thus wasted. This reduces efficiency in direct proportion to the amount of overhang.

The effect of added mass is negligible in all but the smallest speakers. In an 8-inch woofer with a 2-inch, 4-layer voice coil (an extreme example), only about one-fourth of the total moving mass is contributed by the extra two layers of wire. This difference in mass reduces midband efficiency by about 2.4dB; whereas doubling the number of turns in the gap would increase efficiency by 6dB, assuming the same gap flux density could be achieved with the four-layer coil as with the two-layer one.

The author's discussion of maximum excursion would benefit from several clarifications. First, the characteristics of an air spring are highly asymmetrical, resulting in a stiffer spring for compression than for expansion. The resulting even-order harmonic distortion components must be lived with or compensated for by means of a complementary nonlinearity in the loudspeaker suspension. This is very challenging and seldom attempted. Examining the second-harmonic distortion curves for an air-suspension speaker system is enlightening; also a little disquieting, if you happen to own one. In a ported, TL, or horn cabinet, the radiation resistance cannot be expected to provide excursion limitation at frequencies below the tuning frequency. Therefore, electronic excursion limiting is especially important for these types, as Mr. Sink implies.

The relationship of mass and suspension stiffness can be stated rather simply. At frequencies below resonance, the response of the speaker is primarily controlled by the suspension stiffness. At resonance, the mass "inductive reactance" and the stiffness "capacitive reactance" cancel. Thus, changing either the mass or stiffness will change the resonant frequency. At frequencies well above resonance, the response of the speaker is mass-controlled. The only effect of stiffness upon low-frequency efficiency is in the range below resonance, which is usually of little practical importance. In fact, a stiff suspension does not necessarily entail nonlinearities; it only raises F_o. Significant suspension nonlinearities result from requiring more excursion than the suspension is designed for.

Finally, the matter of cone mass should be addressed. While a more massive cone *per se* does not reduce high-frequency response relative to midband response, it is untrue that mass does not affect transient response. This is because more mass increases the tendency for the cone to go on vibrating after the signal has ceased; and results in more acoustic hangover, which is a designer's headache, since it reduces definition. Also, the theory does not account for the tendency of more massive cone bodies to be stiffer than less massive ones.

Although the extra stiffness does somewhat reduce the effect of standing waves in the cone, it also makes the cone behave more like a piston at higher frequencies, reducing the cone flexure that Mr. Sink correctly mentions. The result is that more massive cones do in fact have more restricted high-frequency response, although they have a flatter in-band response. Of course, a well-designed system will cross over to a higher-frequency driver when the piston limit of the woofer is reached.

Richard A. Honeycutt
EDC Acoustical Consultants
Lexington, NC 27292

Perry Sink replies:

Although my article was not meant to explore driver design to the degree your letter does, you make some valid points.

First, my statements regarding the strength of the magnetic field were made under the assumption than the magnet in question was properly designed and that size would be proportional to strength. This is not always true, as you said. My use of "magnet weight" was inaccurate, "magnet strength" would be more appropriate.

I hesitate to respond to your second comment, as I should look into the matter of Q further, to discuss it thoroughly. However, I can say I have observed a close correlation between Q_t and the response of the speaker near its resonant frequency; perhaps I oversimplified the issue (which usually happens in a "primer"). At any rate, I find the characteristics of woofers a bit more predictable than those of rock formations.

Your comments on BL need some qualifications. There are two ways to measure this. One is the method you refer to; another is obtained applying a current (I) to the coil, producing a displacement. A counteractive force is applied to the coil such that it is centered in the gap; given by the following equation:

BL = f (in newtons)/I (in amperes)

This method is not as cut-and-dried, but I believe it yields more useful data. BL product, when measured with either method, will not rise with an increase in the number of layers in every case, because the gap usually must be made wider. A good example of this is made by comparing the 2CA14 and 4CA14 versions of the Audax HD24S45TSMC woofer. The 2CA14 has a BL of 11.4 newtons per ampere (NPA); the 4CA14 measures 7.6 NPA.

I made my statements about cone mass because of incorrect statements made in many *SB* issues. It is difficult for many without a good background in physics to distinguish the various factors of displacement, velocity and acceleration, for speaker considerations of, for example, cone mass and transient response. I intended to clarify this complicated issue, which is especially difficult when the cone is working outside its range. I qualified my statements on this topic at the beginning, by specifying a perfect piston, which would not have standing waves. As to how cone mass actually affects high frequency and transient response, I should say that transient response is affected more by the quality of the surround and internal damping of the cone than by the mass of the cone.

Thank you for your comments. I hope my article and our interaction help to dispel some myths about driver design.

RUMBLE FILTER MODS

I have a few comments regarding Robert Bullock's article on sixth-order vented systems (*SB* 1/82). After reading the article, I too elected to use the Jung 30Hz Rumble Filter kit, modified for use as a response-shaping filter. The filter kit requires an external ±15V power supply, which is then zener regulated to ±6.8V on the circuit board.

The 4136 op amp supplied with the kit will typically swing ±2V less than its supply voltage. The TL075 is a bit better; ±1V less. Based on these numbers, I believe there is a real potential for clipping at the output of the filter, depending on the preamp level at the filter input, and the peak lift called for by the chosen alignment.

My preamp is a Hafler DH-100 and the Class 1 alignment calls for a gain of about 6dB. The DH-100's rated output is ±4.2V and its maximum output is ±11.3V. As Mr. Bullock points out, the actual amount of lift is dependent on program material. Still, I recommend increasing the rail voltages to ±15V. This provides a greater margin of safety and is easily implemented on the circuit board by eliminating the zener diodes and substituting wire jumpers for the limiting resistors, R9 and R10. Since the tantalum filter caps, C6 and C7, are rated at 16V DC, it would also be wise to substitute capacitors with a higher working voltage.

Thanks for the opportunity to comment. Robert Bullock's articles are excellent.

Robert Tooley
Rio Rancho, NM 87124

MCD25Ms MAKE A DIFFERENCE

McGee Radio ran an ad on the back cover of all the *SB* 1986 issues, for a three-way satellite/subwoofer kit. I ordered the kit, but was skeptical whether the system would turn out okay. The components were relatively inexpensive but good quality (Peerless, Polydax), with the exception of the high-end tweeter, which was fair at best.

I built the system, knowing I would eventually upgrade the tweeter, and was pleased with the results. Two years later I ordered and installed a pair of Madisound MCD25M titanium dome tweeters. It is amazing how much difference they make, with cleaner high-end response. The modification was simple, because the replacements are the identical size of the McGee originals. Thanks to Madisound and *SB* advertising these tweeters, I am enjoying a renewed happiness in my speakers.

I would be remiss if I did not mention my success in building a pair of John Cockroft's Mini-Dancers (*SB* 3/86). I built the type-4 configuration, with an inverted dome tweeter, and use the pair as satellites in conjunction with a 10," 2 ft.^{3} subwoofer. The strangeness of the loading inferred the most difficult enclosure I have built. However, the result is worth the effort, as the overall response is quite full; a fat-bottomed sound and warmth is evident on my favorite recordings. I built the system for a 155-ft.^{2} room, but I prototyped it in a room twice that size and had no lack of output level. I am amazed at the responsiveness of this system and thank John Cockroft for his efforts in enlightening his fellow speaker building compatriots to the virtues of Isobarik loading.

Mr. Dell: thank you for your ideals which bring forth this fine publication.

Jeff Cressionnie
Picayune, MS 39466

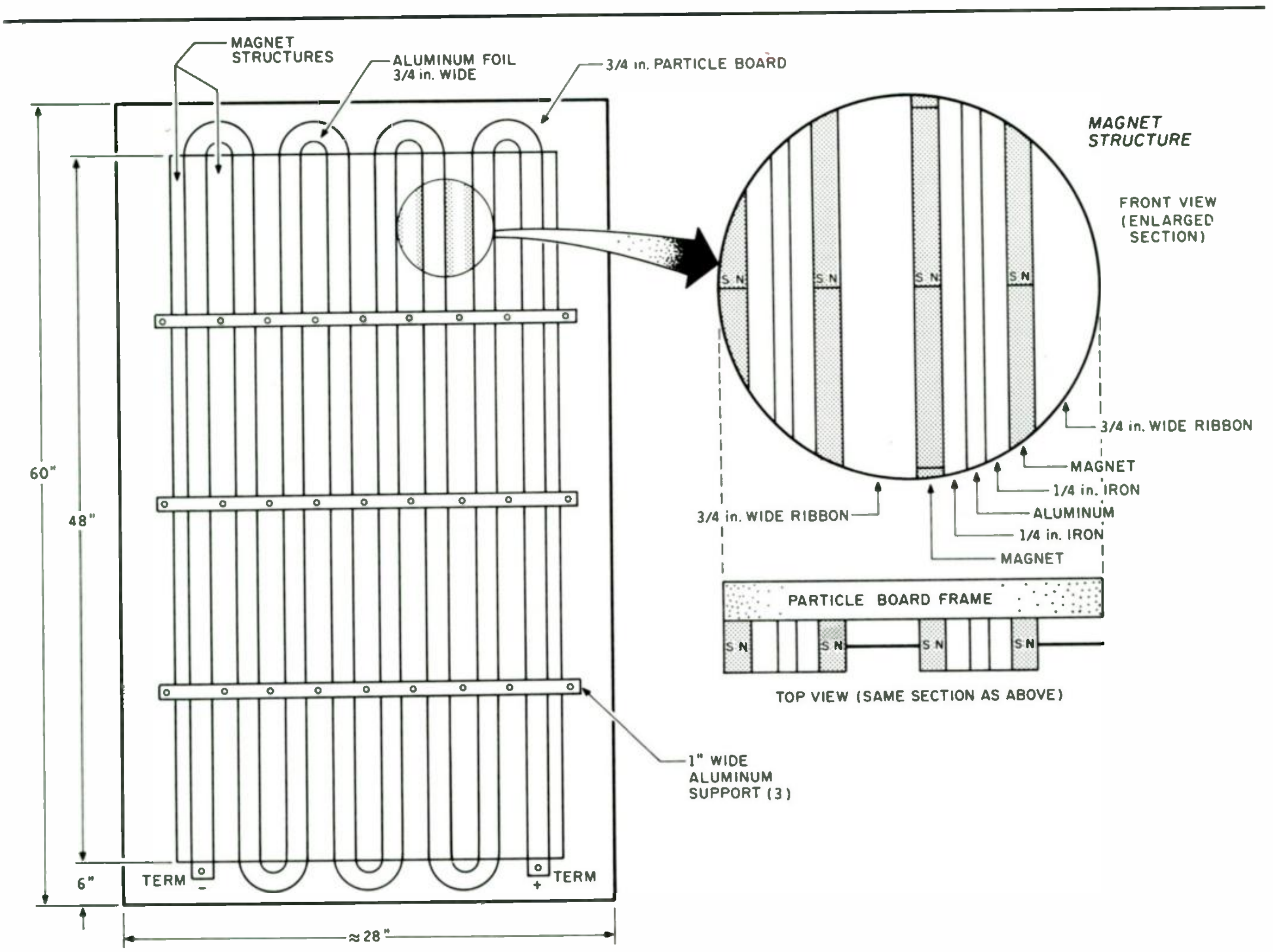


FIGURE 1: Full-range ribbon loudspeaker.

FULL-RANGE ESL

I was very happy to see the ribbon speaker article by Ole Thofte, ("La Folia," *SB* 1/88). I had everything designed for a Magneplanar woofer for my Strathearn driver and then priced magnets at over $1000. Magnepan sells the SMG-a loudspeakers for $500, so there must be a less expensive source. Can anyone suggest one? *[Edmund Scientific, 101 E. Gloucester Pike, Barrington, NJ 08007, is one source. Also see SB 3/84, p. 14 for a list; Source #11 is out of business.—Ed.]*

Mr. Thofte's full range design got me thinking and I redesigned my project. The ribbons are ¾ by 48 inches. The magnet structure is no longer short-circuited and much stronger. I use eight rows of ribbons with about 300 in.^{2} per side (*Fig. 1*). An array could be made on a slight arc rather than a straight line.

I made this setup with ½-inch magnets to reduce diffraction effects. The mild steel pieces complete the magnetic circuit, making a strong magnetic force. The aluminum between the iron pieces insulates the pole pieces from its back neighbor. Horizontal aluminum pieces every 12 inches are needed to keep the frame aligned due to the high magnetic forces involved.

If I could find magnets, under $200 per loudspeaker, I would build and test a prototype. The magnets would be 1 by ½ by ¼ inches; as it looks now, it is not feasible.

William Wagaman
Mertztown, PA 19539

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 Postage Reply coupons (available at your post office) instead of stamps on your envelope.

Please leave room in your letter for replies. Your questions 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.

PUSH-PULL EQUATIONS

I am inquiring about the appropriate design equations to use for a sixth-order push-pull vented design as described by John Levreault in *SB* 2/87. My similar design, twin drivers wired in parallel, reversed polarity to the inward-facing driver, is based on a Ted Jordan design published by Transcendental Audio (no longer in business).

I initially used the Audax HD20B drivers, but had no power capacity. I changed to Becker 908A327 units, with $Q_l = 7$: V_{as}, 2.08; Q_{ts}, 0.47; &, 0.5798; V_b, 3.59. I am using a box of 3.88 ft.[3] for both, which means I ignored the second unit. *SB* articles range from doubling to halving V_{as}.

What is the correct formula to calculate V_b for the simple push-pull alignment?

Richard Mallin
Ambler, PA 19002

John Levreault replies:

When designing a system with two drivers sharing the same volume, system V_{as} is doubled and hence box volume, V_b, must be doubled to obtain a particular value for alpha. It is as if you stacked two identical systems, and then removed the top and bottom panels of the separate enclosures. You should conclude, correctly, that the response will remain unchanged. The alpha is unaffected.

I too was once interested in the Jordan design. I searched my archives and found his design discussion, which basically proved the superiority of a design with a Q_{tc} of 1.0, which further required a driver Q_{ts} of 0.6. The Audax HD 20B25J2C12 was recommended for this alignment. Although I never built the system, I have worked with these drivers and concur that they are short on power capacity. I've blown four of them, but they sounded quite good, despite a high Q_{tc} above 1.1.

Sixth-order alignments, including design equations, were completely discussed in Bob Bullock's article (*SB* 1/82). I suspect your Becker devices may need a fairly large box since they have a relativly high Q_{ts}. I suggest a session with BOXRESPONSE, to see if you can arrive at a design, acceptable for your listening requirements and preferences.

MIX 'N MATCH RESPONSE

In response to Mark Butcher's letter ("Mix 'n Match Amps," *SB* 4/87, p. 56), I doubt there are any power amplifiers on the market which have significant phase shift in the range of frequencies where crossovers appear. If the amplifiers in a biamplified system have differing polarity (one inverts and one doesn't) you can simply invert the polarity of the corresponding driver.

Active crossovers can simplify crossover design and reduce the demands on power amps and passive components. Yet, in many cases, it is not only possible but actually easier to obtain the same results with a well-designed passive crossover and a high-quality amplifier. Contrary to Bob Ballard's letter ("In Praise of Active Crossovers," *SB* 4/86, p. 52), passive crossovers need not degrade a speaker's low frequency transient response, and in some cases (e.g. an overdamped woofer) actually *improve* it. In addition, knowledgable designers can fully compensate for the complex driver impedance over the relevant range of frequencies, and readily produce accurate passive bandpass filters.

For speaker builders who have the confidence to "go active," the advantage is in the economy of choosing amplifiers whose power and distortion characteristics complement the frequency range over which they are used: for example, a low-power, Class A vacuum tube tweeter amp and a high-power (Class AB) transistor woofer amp.

Ralph Gonzalez
Philadelphia, PA 19143

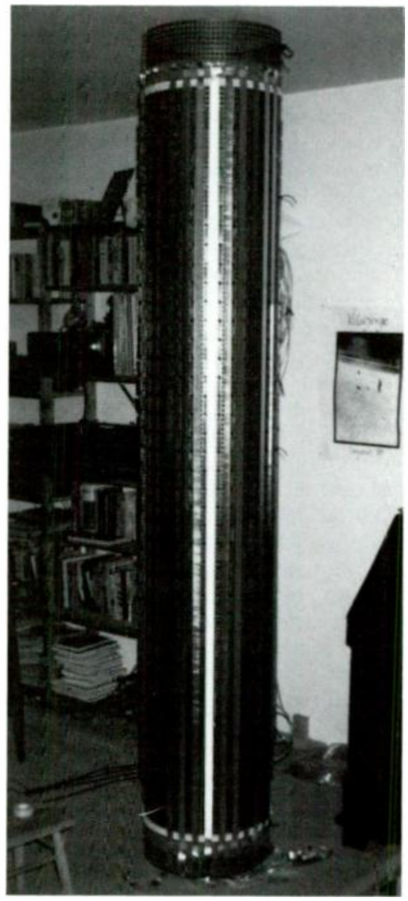


PHOTO 1: A pulsating cylinder.

LA FOLIA UPDATE

As you can see from the *Photo*, my latest ribbon experiment is tube-formed and works as a pulsating cylinder. It has 30 ribbons (2.5m by 2cm) and 1200 magnets (42 by 7 by 9mm). I thought this speaker would be rather powerful, but it turned out *very* timid and inefficient. Is my magnet pattern too weak? Too much air—short-circuiting?

I welcome a good explanation. At this modest sound level, it seems the sound stage capabilities of this speaker are very good. My next step will be, instead of the ribbons, to make a planar membrane as a pulsating cylinder. Also, I think I'll experiment with the diameter of the tube. It might end up something like a Martin-Logan shape.

I hope my article ("La Folia," *SB* 1/88) stirs development. Last year *High Fidelity* (a Danish publication) had three interviews with people who built La Folia. Several improvement were made and I heard some builders are now focusing on a better finish. It is really interesting to see this spread of ideas.

Ole Thofte
Menstrup, Naestved
Denmark

TL CHALLENGE

John Cockroft's Shortline (*SB* 1/88) is a novel hybrid design using transmission line loading as part of the concept. Mr. Cockroft compares data on the Shortline with my article ("An Experimental Transmission Line," *SB* 4/85). While it is a welcome addition to the meager fund of published data there are some statements in the article that call for more discussion.

Mr. Cockroft states "the purpose of the line is to attenuate the backwave of the speaker and to damp impedance peaks," which implies that resistive damping by line stuffing is the principal goal of all transmission line design, and I disagree.

Several different concepts of optimum transmission line length design might be grouped generally as:

1. Longer is better, but one-quarter wavelength in air at desired low frequency cutoff is a minimum length. Translation if this concept into actual line length design is often somwhat vague.
2. Driver mechanical resonance (described by the electrical impedance peak) should be opposed by an acoustical resonance of similar amplitude and Q, but opposite phase. Resistive damping by the line stuffing is limited to undesired line harmonics.
3. Resistive damping as measured by impedance frequency curve is more important than line length.

Exponents of the longer-is-better concept say longer length provides lower system cutoff and more effective damping. Longer lines do provide more effective resistive damping. However, this higher level of resistive damping lowers line output at all frequencies, including desired low frequencies.

The second concept requires opposition to the driver resonant impedance peak by reflected out-of-phase acoustical resonant impedance from an open-ended quarter wavelength line. Optimum line length for the driver/line *system* to provide quarter wave reflections (not free air quarter wavelength) may be determined by experiment. This means looking at impedance valleys rather than peaks and this was the intent of my article.

A.J. Bradbury related the effect of stuffing density on sound speed (*JAES*, April 1976). However, as my article indicates, I have not been able to confirm more than a minor change in effective wavelength of a driver/line system using a stuffing density of 0.5 pounds per cu. ft. Therefore, the major function of stuffing in this concept is heavy damping of undesired line harmonics, combined with minimum damping of desired low-frequency backwave.

The third concept relies on line stuffing to provide resistive damping of both driver resonance and line harmonics. As Mr. Cockroft noted, these shorter lines generally require higher density stuffing. As stuffing density increases, the line frequency/impedance graph begins to look like a sealed-box graph, with the disappearance of the lower impedance peak.

I agree with Mr. Cockroft that a "leaky line" is better than a sealed box for general reproduction of low frequency sound. However, I would expect reduced low-frequency output due to increased stuffing density.

As I said, the Shortline is a unique and interesting design. I basically challenge (with words, since we don't fling gauntlets anymore) the implications of his comments on general transmission line design objectives.

T.E. Cox
Wilton, CT 06897

John Cockroft replies:

I'm glad Mr. Cox appreciates my data, although I am a bit disconcerted he thinks I have trod upon his transmission line toes. This was not my intent.

While I regard resistive damping an important and effective means of achieving the sort of sound that pleases me and, apparently, others; I strongly doubt it is considered, much less approved of, by many. I think (of course, I don't know) that Bailey would not subscribe to the concept of opposed resonances. He was heroically attempting to remove all resonances.

Mr. Cox is correct in Concept 1, when he refers to vagueness. I can also live with his Concept 3, but I certainly would not mandate it for the general public. Peaks and valleys cohabitate. I don't believe it matters whether the Shortline has a 1dB peak, or a 1dB valley.

Perhaps if the backwave was damped along with the harmonics, the frontwave would have a chance to speak.

While it's true that as stuffing density increases, the impedance graph begins to resemble a sealed box's graph, it doesn't sound like a box and that's what is important.

My only interest in speaker systems is the way they sound. Although I don't recall making any statements regarding the general reproduction of low-frequency sound, I'm glad Mr. Cox agrees with me. I'm not even sure what is meant by a "leaky line." Neither I nor those who have heard my densely-stuffed systems have ever complained about the absence of low-frequency output. In a moment of candor, I admit I expected reduced low-frequency output; I just didn't find it.

A DELAYED RESPONSE

While running through some back issues, I came across a *Mailbox* item ("Magnet Malady," *SB* 4/83) in which Rafael Lopez notes a white corrosion on his driver magnet plates. When I first read his letter, I recognized the problem and meant to send off a note. I don't know whether anyone is still curious, but here is my reply.

I'll bet his magnet plates, and other parts of his driver's structure, were a bright white metallic color, perhaps with a brassy tinge. Often steel components, not intended to be painted, are treated with a zinc or cadmium coating. If not a true hot-dipped galvanized or electro-plate, the process involves a fast and inexpensive "flash-coat" ionic vapor deposition, or sherardizing, in which the part is tumbled in a hot bath of zinc powder.

This provides a measure of anti-corrosion protection, as the oxide of zinc clings, rather than sloughing off like iron oxide (rust). The coating is intentionally sacrificial, meant to oxidize rather than prevent oxidation altogether. It will prevent the corrosion of steel not exposed to severe conditions. Outdoor conditions, as in roofing sheet metal, culvert pipe, door frames and so on, require thicker coatings from hot-dipped galvanizing.

I cannot say what caused Mr. Lopez' corrosion problem, without doing a thorough investigation. But I had the same experience with some Polydax units; the corrosion developed surprisingly quickly, and I realized corrosion is a chemical reaction and fundamentally, with a magnet producing a constant electrical polarity, an electrical phenomenon.

If my reasoning is correct, I see no cause for concern. If you wish to clean this oxidation off, remember you are removing the protection provided in the first place, and you must, absolutely, provide the paint job, as Mr. Lopez did. Spraying the parts lightly with with clear lacquer when new, might be best. Be sure to keep paint away from where it doesn't belong.

I apologize for my four-year delay to this query. This is, after all, a forum for sharing ideas, and corrosion protection is more important than I had realized and worthy of further discussion in these pages.

Paul Graham
Independence, MO 64050

FRONT-REAR LOADING

I am a subscriber since the first issues of *SB* and *TAA* and I would like to suggest a tutorial series about rear- plus front-loading loudspeaker enclosures, including computer software and mathematical analysis.

Besides some famous designs such as Tannoy's GRF Autograph (front and rear horn loading) and Western Electric/Altec Lansing A7 (front, horn loaded; rear, reflex loaded), we have the following:

- The F.A.S. Air Coupler, Fowler, Allison, and Sleeper (*High Fidelity*, No. 1, Vol. 1, 1951);
- The Karlson enclosure, rear, reflex; front, slot loaded;
- The RJ enclosure, Frank Robinson and William Joseph, (*Audio Engineering*, Jan. 1953);
- The Janis Woofer enclosure, rear, acoustic suspension; front, slot loaded;
- Electro-Voice's MTL-4, manifold technology bass enclosure;
- Elipson's Charge Simetrique, now marketed by KEF, bandpass loudspeaker enclosure (R107);
- Metastatic Slot, dual driver from Linear Power and Langley Theater, National Air and Space Museum, (*B.A.S. Speaker*, No. 1/2, Vol. 14, p. 14);
- and the "goose neck" proposed by KEF and Speakerlab, for auto installation.

Could anyone supply me the address of the manufacturer of Credence raw drivers? Is Sherman Research out of business? *[Yes.—Ed.]*

Nestor Natividade
04581, Brooklin, Sao Paulo
Brazil

The Editor Replies:

Although I have access to copies of the original "Air Coupler" articles in the early issues of *High Fidelity*, both Charles Fowler and Roy Allison have long since disowned the design. Many of the designs are patented and not reproducible here. I have omitted two designs from Mr. Natividade's list by a large Massachusett's manufacturer who does not allow any reproduction of technical details.—E.T.D.

TWO-WAY RIBBON

Continued from page 40

less efficiency? Use fewer magnets or add a 1Ω or 2Ω resistor in series. Want more efficiency? Add magnets. A lower crossover point? Use longer ribbons and enclose the back to eliminate acoustical cancellation.

I hope you experiment with a ribbon midrange/tweeter. In my 10 years or so of speaker building I have never been as delighted as when I first heard my own strip of aluminum singing.

SOURCES

Edmund Scientific (magnets)
101 E. Gloucester Pike
Barrington, NJ 08007
(Part #33,027)

DRH (crossover caps)
2275 East Bay Dr. #1205C
Clearwater, FL 33546
(813) 536-2904, after 6pm

Old Colony Sound Lab (binding posts)
PO Box 243
Peterborough, NH 03458

Arnold Kaczor (steel plates)
1137 E. Huron County Line Rd.
Ubly, MI 48475
(Mr. Kaczor has offered to produce the plates needed for a stereo pair for $50, plus shipping.)

CORNER FREQUENCY AND PLOTS

In Mr. Gonzalez' first LMP article (*SB* 1/87) he states the corner frequency comes from "the frequency at which your driver has the same magnitude relative to the flat portion of its response." I'm not sure what this means. How does he arrive at the corner frequency from the manufacturers' curves? Is it the 3dB down point at each end of the driver's response?

I have another question on interpreting the phase plots the program generates. Mr. Gonzalez mentioned in the second LMP article (*SB* 2/87) one of the goals is to "get the phase angles of the driver/crossover sections within 90° of each other at the crossover frequency."

I don't know what the plots should look like for first-, second-, third- and fourth-order responses when this goal has been achieved. Does each one's phase plot appear different? If so, what do they look like? I can't tell when the 90° per section goal is reached, but I know the magnitude response is supposed to be flat at that point. Also, since even orders have a different phase response than odd orders, how is a 90° per section achieved with all of them?

David Del Zotto
Seattle, WA 14726

Ralph Gonzalez replies:

Your questions are reasonable and I hope this reply helps clarify the issues.

1. In Fig. 5a of my article (*SB* 1/87, p. 21), the corner frequency is where $f/f_c = 1$. The corner frequency does *not* equal the 3dB down point of the driver's response, except in the case where the damping ratio, $d = 0.7$. If the damping ratio is less than 0.7, then there may actually be a *peak* at the corner frequency. For example, suppose your woofer's high frequency response resembles that in Fig. 5a with $d = 0.3$. In this case, the magnitude at $f/f_c = 1$ is about +4dB. You now return to the manufacturer's frequency response curve to find the value of f_c: this frequency should be near the +4dB peak of your driver's high frequency response.

2. When I say the phase angles should be within 90° of each other, I mean the *difference* between these phase angles should preferably be *less than or equal to* 90°. First use LMP to plot the woofer and its crossover alone (reduce the tweeter's SENSITIVITY by about 60dB) and note its phase angle at the crossover frequency. Repeat this for the tweeter with its own crossover. Now subtract the smaller of these two phase angles from the larger one. For example, with a second-order Linkwitz-Riley crossover using *ideal* drivers (enter zero in response to each of the DRIVER INFORMATION prompts) the phase angle of the woofer/crossover is −90° at the crossover frequency, and that of the tweeter/crossover is +90°, so the phase difference is 180°—clearly undesirable. By inverting the tweeter's polarity (as is recommended for an ideal second-order crossover), both phase angles will appear at −90°, so the phase difference will be zero.

Unfortunately, due to driver limitations, you will never obtain an *ideal* second-order crossover, and you will have to repeat the experiment incorporating models for your own drivers.

It is usually undesirable to have a phase difference much greater than 90° because obtaining a flat, net-magnitude response in this case may require the individual drivers to show a peak around the crossover frequency. There will also usually be a peak in the *reverberant* response and undesirable vertical lobing.

A UK REPORT

More and more persons are recognizing the benefits of bi- and tri- wiring; a number of manufacturers bringing out speakers here in England with split crossovers: Rogers, Music Fidelity, B&W, and so on, as well as home builders, of course.

An upgrade of this has been introduced by David Rusby in a new quarterly, *Audio Conversions*, a UK publication written by leading designers. *[We are inquiring about this publication. We will publish the address when we learn it.—Ed.]* The article on "star wiring" includes upgrading capacitors, inductors and resistors, and then tri-wiring. The author decided to try star wiring to stop the crossover currents modulating the driver returns, and he found the improvement was immediately apparent, for the cost of a few plugs and leads.

An article by Russ Andrews includes a discussion of Ray Kimber, of Kimber Kable fame, and his research into copper conductivity. His measurements show ordinary copper 10 times more conductive than solder, and recommends screw terminals, binding posts and crimp-on connectors, plus contact enhancer (Tweek, Cramolin) for the best sound quality.

I remember three years ago the controversy began on the merits of using solid core cable. Its initial promoter was James M. Hughes, a reviewer in *Hi-Fi Answers*, inspired by the research of Denis Morcroft. His view that solid core was superior to stranded cable first created a fair amount of incredulity. I initially tried 1mm solid core mains cable on one speaker, with the usual stranded cable on the other, with a mono signal. I prefered the the solid core almost immediately, switching between the two for a couple of days to make sure. I also rewired internally with solid core, again with improved sound.

One explanation for this shows complex, rapidly changing magnetic and electric fields cut through the individual strands inducing stress currents, smearing the signal currents. A thorough explanation, by Dr. Malcolm Hawksford in *Hi-Fi News*, stated the optimum diameter for audio cable was around 0.8mm.

Another generally accepted idea which is generating big business here is the use of speaker stands. The stands shown in Ralph Gonzalez' article (*SB* 3/87, p. 40) Heybrooks HBSIs, are one of the most popular for small speakers. The general consensus is they should offer a highly rigid, stable platform, and include spiked feet for greater stability and improved sound. The cabinets should be supported at points of maximum rigidity, about ½" in from the corners. Most stands are now available with upward facing spikes. You could use cross-head screws inserted into the cabinet corners to locate the spikes, or bolt the cabinet directly to the stand, to improve stability.

The ultimate example appears to be building the stand and cabinet as a rigid integrated structure, as adopted by Naim Audio SB2s and Mourdant Short HH2s. The ambitious home builder could use 2 by 3 timber to form the internal frame and integrated stand, and rigidly attach the front baffle and cabinet to the frame, possibly decoupled by rubber or cork to reduce vibration energy from the baffle.

On a final note, some manufacturers are using vertical figure-eight braces, attached to the inside top, bottom and side enclosure panels, as stronger alternatives to horizontal braces.

On an esoteric level, adverse effects from the ionization of the air, which generates complex charge potentials in and across audio equipment, furniture, LPs, and within the cabinet stuffing, should be neutralized for better sound quality, according to Peter Belt, who has proposed a number of methods. This led some to remove the stuffing, claiming improved sound. Controversial, to say the least.

I hope some of this will be of interest to other readers.

R.D. Lewis
Tarporley, Cheshire, CW6 9RB
England

DRIVER PHASE RESPONSE

Unlike Mr. Knittel (*SB* 1/88, p. 49), I have not found impedance compensation (Zobels) makes driver phase response more constant with frequency. In my experiments I found it had very little effect on the waveform produced by the speaker. I used:

1. Direct driver connection to the power amp, a McIntosh MI-75;

2. As above, with a 0.5Ω resistor in series with the speaker, to represent the loss a passive crossover could cause;

3. A high resistance source of 1kΩ in series and driven by a Hewlett-Packard 200AB sine wave generator.

I consider conditions #1 and 2 realistic. I observed each condition with and without a Zobel. With condition #3, there was slightly more phase shift, with the Zobel tending to decrease this slightly. The Zobel did reduce amplitude at higher frequencies.

I tested a Dynaudio 17W75, using a subminiature Audio-Technica condenser mic (AT 803S) placed about 1mm directly over the cone (with the 17W75 really an inverted dome), where the voice coil attaches. This minimized room and cone breakup effects and time delay. It also contributed to a clean mike output. I observed phase and amplitude waveforms on a Tektronix 502A dual beam oscilloscope. The reference beam was triggered by the amplifier output (#1 and 2), or the signal generator (#3). I used a Heathkit IM-2420 frequency counter to accurately set frequency.

If we look at phase at the speaker terminals with a high impedance source and Zobel (#3), the observed phase difference is nearly 0 above 400Hz, as it should be. This does not mean the speaker is putting out a 0° waveform, but a passive crossover would *see* a resistive load in this frequency range. I actually use this setup to determine Zobel values, and I simply choose values to get the frequencies where the rising part of the speaker impedance curve is as near to 0° phase with the source as possible. I used 14μF in series with 7.5Ω for the 17W75.

The dual beam oscilloscope-microphone setup can also be used to align speakers in a system. I can move the mike perpendicular to the listening axis plane, using a stand and clamping apparatus. The oscilloscope reference beam is triggered by the amplifier output. I clamp the mike in front of the low-frequency speaker, set the sine wave generator at the chosen crossover frequency and mark the position of the mike output on the oscilloscope screen with a wax pencil. Then I move the microphone in front of the high-frequency driver and move the driver back until its output position matches the low-frequency mark. This places the two drivers in-phase at the crossover frequency.

I think it is common to focus on phase changes caused by the crossover, but from my experience I realize the difficulty of making a linear phase speaker system where time delay is either zero, or constant with frequency. So, if you use a first-order crossover, the summed electrical output adds up to the input in phase and amplitude; the speaker themselves will almost always ruin the desired phase relationship. *JAES* articles by Leach (Anthology II, pp. 149-160) and Heyser (Anthology I) have more to offer on this.

Robert L. Shultz
Loma Linda, CA 92354

Max Knittel replies:

We are not in disagreement; I am guilty of poor writing. I intended to say that for driver phase response with compensation networks, the phase at the driver terminals is nearly constant, not that the phase of the acoustic output is nearly constant. The compensation network produces a nicer (more resistive) load for the crossover so it behaves more as you would expect—just what you found.

Drivers themselves have large phase shifts. As usual for resonant systems, any resonance causes fluctuations in both the magnitude and phase of the output. The drivers with the least acoustic phase variations would seem to be those where spurious resonances—those at frequencies above the fundamental free-air resonance—are best controlled. It would seem that driver manufacturers should start publishing acoustic phase vs. frequency curves, in addition to the normal acoustic amplitude vs. frequency curves.

The problem of phase changes intrinsically caused by the drivers is why it is not possible to "physically" align the drivers by merely calculating ¼ wavelength at the crossover frequency (the 90° phase shift of a first-order crossover) and allowing for the distance of driver depth (that is, aligning the voice coils). Although this is a good starting place, you must empirically move the drivers around, as in your method, because the of the known phase shifts in the drivers themselves. In a frequency region around the crossover point, try to find the best compromise high-frequency driver setback relative to the low-frequency driver.

Although the acoustic phase response is always less than perfect for any particular frequency, I have found the desired phase relationship is not completely "ruined" and does add to the spatial accuracy and stability of a stereo image.

POLK 10 MODS' EFFECT

In reference to Mr. Frane's article, "Modifying the Polk 10s," in *SB* 4/87, I suspect his modification may have some unforseen effects. If Polk uses a passive lowpass crossover for the woofers in this speaker, then the modification may require a change in the component values to reflect the new effective impedance. Also, the presence of the other channel's crossover may alter Mr. Frane's calculations of the resultant impedance after making the modification, and I'm not sure whether other problems may result due to the electrical coupling of the "inboard" and "outboard" speakers, particularly near the crossover frequency.

Perhaps the best solution would be to disconnect the *positive leads* of the outboard woofers as well, and attach them directly to their respective amplifier terminals, bypassing the crossover (possibly inserting an inductor as a low-pass filter). Then, recalculate the component values *or* wire a high-power 8Ω resistor in parallel with the inboard speaker. The latter approach uses Polk's original crossover but will present a lower impedance to his amplifier.

In addition, since only the inboard

woofer carries the main signal, he might need to attenuate the tweeter up to 6dB with an L-pad, and may also need to retune the enclosure volume and/or the passive radiator for optimal bass. Good luck!

Ralph Gonzalez
Philadelphia, PA 19143

Jim Frane replies:

Thanks for your thought-provoking comments. Polk does use a passive low-pass crossover for the Model 10s which I modified. As you correctly point out, the high frequency cutoff point of a given inductor, or choke, that serves as low-pass filter will vary with the impedance of the driver(s) involved. Assuming Polk has used a 0.3mH inductor for a 3kHz crossover with a 6Ω load, then the change to 5.33 with the same inductor would change the crossover point to about 3.2kHz. Since the positive leads to both the inboard and outboard speakers come from the same point in the same crossover, I don't believe the crossover will see them separately as far as the impedance is concerned.

This modification was an empirical experiment; I attempted to improve ambience reproduction for these particular speakers. I did not attempt impedance measurements of either a single modified speaker or the pair working together. Unfortunately, I no longer own or have access to these speakers. I would encourage you and other readers to pursue your own theories and report on the results.

ITERATIVE MEASUREMENTS

When calculating crossover values for a three-way system, how do you handle the situation when the impedance at the lower frequency is very different than at the higher crossover frequency? For example, in the Dynaudio D-76 (second-order, APC):

impedance at 500Hz = 6Ω
impedance at 3kHz = 10Ω

Do you use the average midrange driver impedance? Or, do you figure the *bandpass* in two stages, with the high-pass at low frequency based on the low frequency impedance and the low-pass at high frequency section based on the high frequency impedance?

If this is not correct, please describe the correct procedure; I am math proficient, simplicity is not essential.

C.E. Sadler
Dover, DE 19903

Robert Bullock replies:

What you want to do is design a passive filter, by means of formulas, that will operate correctly into a nonresisitive load. In general this cannot be done

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using closed-form formulas. An iterative process such as Schuck's optimization, or a laboratory trial-and-error procedure is the only way to deal with the problem. Either technique requires moderately sophisticated measurement capabilities.

Your alternative is to equalize the driver according to my directions in "Passive Crossover Networks," (*SB* 1/85, p. 19, *Fig. 19*). The object is to make the driver appear as much like a resistor as possible, at least in the neighborhood of the crossover frequencies. The more nearly you can approach a resistive load, the more nearly the filter will behave as advertised.

For your crossover frequencies I would try to equalize to a nominal 6 or 7Ω load, from 250Hz or lower to 6kHz or higher. By 7Ω nominal, I mean within 20%.

POSITIVE FOR PASSIVE

I first began using biamp and triamp designs for my hi-fi system back in the early sixties, but abandoned this approach due to the complexity of the electronics. Recently, after reading some of the comments by Ralph Gonzalez and Contributing Editors Bullock and D'Appolito, I decided to go back to this approach. It is so much easier to do this now than it used to be, with the advent of op amps.

While current state-of-the-art speakers sound much different driven directly from the amp, rather than through a passive network, they do not always sound better that way. Now, before I'm besieged by comments that state "any difference you hear with passives must be negative," let me say I realize this involves a modification of the electrical signal. However, this is not always objectionable.

An example is a well-damped bass system, ported or closed box. There's a good chance that if it sounds perfect when driven through a passive network, it will sound overdamped, driven directly from the amp. This can be rectified by placing a small resistance in series with the speaker, maybe 0.1 to 0.2Ω. The Q of the system is back to where it was with the passive device. I have seen this effect several times with closed-box systems using top-of-the-line drivers with efficient magnet systems.

I have not been at all happy with my 3/2 D'Appolito satellites when driving the speakers direct from the amp using high level crossovers. I am using the Dynaudio 17W extended range in this application. The system sounds too bright and the term "glare" seems to fit it best. Some might describe the sound as more open, but the effect wears thin after hours of listening. It also sounds awful on anything but pristine program material.

Conversely, when driven through a passive crossover with Zobels in place the sound is much too dry and constrained. Interestingly enough, the best sound is with no Zobel at all, using a network designed for the impedance of the speaker pair at the crossover frequency. This also produces the flattest measured system response.

The electronic crossover between the sub and the 3/2 unit is 100Hz with 18dB up and 6dB down. This sounds much better than the 12 up and 12 down that I used previously.

I mention this to encourage readers to listen and make changes to determine what sounds best. The best sounding amp/speaker combination is not always the most perfect electrically. Another note on Zobels: they sound and measure differently, depending on whether the resistance or capacitance is placed first in the circuit. This is probably caused by interaction with other reactance in the crossover.

I would like to compliment *SB* on a fine periodical with excellent input from its contributing editors.

Judson Barber
North Augusta, SC 29841

Joseph D'Appolito replies:

Both active and passive crossovers are legitimate approaches to frequency division. The preference for one over the other is application specific. Usually, either approach can be made to work well, but in any particular application one may be simpler than the other. The argument used to be that active approaches were better because the amplifier isolated the crossover from the effect of nonideal driver impedance loading. With the advent of modern passive crossover optimization software, which accounts for both driver impedance and response variations, this argument is no longer valid.

I have been biamping for almost 20 years and find the greatest benefit comes from biamping between the bass drivers and the remaining upper range drivers, providing the system has been designed from the beginning for this approach. On the other hand, I have also found that the response anomalies of midrange and tweeter drivers, which must be accounted for in any design, are often more simply and directly corrected with passive networks. For example, the high-frequency rolloff of the Dynaudio D-28 driver in my 3/2 satellite is not the inverse of any simple 6 or 12dB/octave active network. However, a simple parallel RC passive network in series with this driver combines with its rising voice coil impedance to provide exactly the right compensation to flatten and extend its response to 21kHz.

Our goal is an overall correct acoustic response. Properly applied, both active and passive crossovers can be made to work well. Blind application of standard crossover circuits which do not account for driver response irregularities or frequency dependent input impedance variations will usually lead to poor performances. Although active approaches eliminate the effect of impedance variations, we must still deal correctly with response anomalies. I think paragragh five of Mr. Barber's letter tells it all: the crossover which produced the flattest response sounded best.

DENSER PLYWOOD

I built a pair of Octalines for my daughter, using John Cockroft's suggestions for finishing (*SB* 3/87). They are very nice sounding and looking. For my next project I plan on using Apple-ply, an American, multiply plywood made to compete with Baltic birch plywood at a much lower cost. It has nothing to do with Apple wood, but has a birch face, is much denser than regular plywood and has no gaps.

Can anyone tell me how to put preamp outputs on my Sony receiver with remote control so I can biamp my speakers but still use the remote volume control? Is there a place on the circuit board that can be tapped?

Michael Dunn
Eugene, OR 97401

BOXRESPONSE UPDATE

I am an avid user of BOXRESPONSE and note from Bob White's letter (*SB* 1/86, p. 54) that he has some additional codes to display X_{max} for a given frequency and input power. I would like to have a listing of the code for inclusion in my version of BOXRESPONSE (I have converted it to run under Lotus 1-2-3). Thank you for a wonderful program and in advance for the new code.

Chris Edmundson
Seattle, WA 98102

Bob White replies:

It's nice to hear from people who are still using the original version of BOXRESPONSE. So many things have happened to that program since Bob Bullock first wrote it, I figure I should give you the changes you requested and bring you up to date.

First of all, eight lines differ from the original code and I added eight new lines:

Astute observers will notice the equation for R9 and R9R are not used in the current version. The value R9 is the voltage limited excursion rather than the required excursion at the thermal power limit specified at the beginning of the program. The value of one number or the other is a matter of taste. Experiment, substitute R9R for RYR in line 490 and see what happens. As a side note, if you have a purchased copy of the disk, you have both BOXRESPONSE and BOXRESPONSE XMAX on your disk.

This next section is not for the faint of heart. In my attempt to improve the original program, I searched high and low for an easy way to generate graphics on the Apple II screen. Anyway, the answer I finally found is easy. A program distributed by Roger Wagner Publications, Inc., called CHART 'N GRAPH TOOLBOX, lets you define and scale your own screens (called windowing). Picture this, with a simple statement you can put log scaled data on the Apple II screen. It's a bargain, too. I found it in a mail order ad for $27.

The upshot of all this is if you get a copy of this Toolbox for your own use, I can supply a copy of the listing of BOXRESPONSE XMAX that does graphics. The graph shows excursion in millimeters and displacement-limited SPL. A dashed line on the display shows the maximum linear excursion, which you input. This version of the program is very useful in getting the maximum "bang for your buck." With the proper choice of box volume and tuning you can make a driver with a a small displacement achieve high SPL before being excursion limited. I would have gotten the Toolbox distribution license and made this version available, but the price would have been about the same.

Contact the Editor (if you are not faint of heart) and listings and diskettes will be available if interest warrants.

```
330 PRINT "                                              MAXIMUM"
331 PRINT "                   MAXIMUM     MAXIMUM     REQ'D"
332 PRINT "FREQ.  RESP.    INPUT      INFINITE     EXCURS"
333 PRINT " IN      IN       IN       RESPONSE       IN"
334 PRINT " HZ     (DB)     WATTS      IN (DB)       MM"
335 PRINT "=========================================="

490 PRINT F9;TAB(6);Y2R;TAB(15);P1R;TAB(23);Y3R;TAB(33);RYR

550 DATA 1,5,12,15,20,25,27,30,32,35,37,40,42,45,47,50,52,55
         60,65,70,75,80,85,90,95,100,110,125,150,175,200,210


New lines:
155 INPUT "ENTER DRIVER NAME ":D$
437 RY=Y1*K1*E*1000
456 R9=Y1*K1*E2*1000
485 RYR=FN R(RY)
486 R9R=FN R(R9)
561 F4=R1*25.4
562 PRINT "XMAX IN INCHES = ";R1
563 PRINT "XMAX IN MM      = ";F4
```

Classified Advertising

TRADE

FALCON ACOUSTICS is one of the largest independent manufacturers of crossover networks and speaker system accessories in the UK, supplying manufacturers, retailers and export. We have outlets in both the USA and Canada. Your guarantee is the trust our manufacturing customers put in the quality and reliability of our networks. **Inductors:** Ferrite (up to 200W), air-cored and transformer; **Capacitors:** Reversible electrolytic 50V/100V, polyester, polycarbonate and polypropylene; **Networks:** choice of over 70 for different unit combinations; **Components & Accessories:** most except for the wood! If you have supply problems, please send for details, $2 air, $1 surface, to our mail order retail outlet. ***FALCON ELECTRONICS***, Tabor House, Norwich Rd., Mulbarton, Norfolk, England NR14 8JT. TTF

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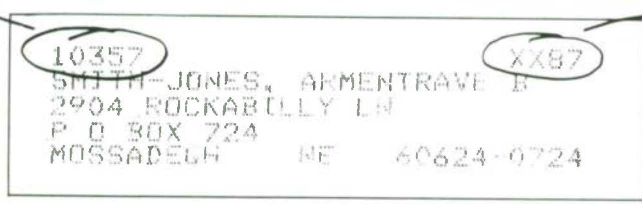

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5¼ INCH CAST FRAME, Neoflex cone, vented pole piece, 40mm Kapton former aluminum, edgewound flat wire. Voice coil length, 18mm, 15Ω driver, Fs = 43; Qts = 0.274; Vas " 11.4 liter; 87dB/W/M. These special drivers are brand new and the original contractor could not take delivery. These are a special one-time offer at a cost far below market value. They are ideal for line array systems. For prices and quantities, call ***FOCAL AMERICA***, (818) 707-1629. T4/88

FOR SALE

Borbely preamp kit all boards stuffed, careful work, silver soldered, no time to complete, $375; Lang power amps, 20W full kits, stuffed boards, $250/pair; Falcon Acoustics Tocatta speakers, two KEF B110s 5¼", one 1½" T52B tweeter, vented enclosure, excellent condition, $250/pair; Becker 10" woofers, 910A100, new, $50/pair. Dan Hildreth, 1514 E. Stephens, Tempe, AZ 85283, (602) 838-2789.

New Morel MDT-30 tweeter, $20; used MDT-30, $12; pair used Focal 7N501s, $40; Omnisonic 801 image enhancement unit (useful for making tapes for your car), $15. Duke, (504) 866-0191.

Pair of 30W-54 12" poly woofers, almost new, $240. John, 122 Forest Hills Rd., Springfield, MA 01128, (413) 783-7113.

Heil air motion transformers, several styles, $150-300/pair; 12" ESS woofers (cast), $200/pair; ESS crossovers, $50/pair; Heil replacement diaphragms, $35. Richard Paino (916) 988-0673.

New JBL 15" woofers 2235H (LE15-pro version), Prosound JBL 2215A, Electro-Voice 30" subwoofer drivers; Eton 8" Kevlar, 280Hex, Focal 10" Kevlar 10K515, Audax 8" Bextrene. Limited number of 10μF Siderealkaps. Bob Pickard, 1720 Lilac, Walnut Creek, CA 94595, (415) 930-9104.

'81 Eminence 38 oz. 12", $25/pair; new Audax 8" H4C12, $39/pair; perfect Dalesford 5", $39/pair; old KEF B110, $12/pair; Dynaudio D28, $28/pair; SEAS 1" ferro tweeters, $17/pair; Audax super tweeters, $12/for 4; Audax 6.5" H4, $12/one only. Lee Arbach, 1987 Autumn Gold, San Jose, CA 95131, (408) 258-7971 evenings.

Speaker cables: 3 pair MIT PC-18 cables with Esoterica gold bananas, 2 pair @ 8', $100/pair, 1 pair @ 12', $125; 3 pair Livewire Teflon 12 with gold spades, 6, 8, 12 feet at $25, $30, $40; 2 pair Livewire BC-12 with gold spades, 8' @ $50, 12' @ $75. Larry Cartwright, (412) 846-7419.

Heathkit three-way time-aligned speaker system model ASX-1383, 10" woofer (B2 aligned with f_3 of 45Hz), 5" Peerless midrange, 1" Peerless dome tweeter, 6dB/octave crossover at 750 and 4000Hz, in beautiful rosewood cabinet, $300. Bruce Beyer, (313) 455-1728.

Linn, Ittok and AT-OC7 mc, $600; Nitty Gritty II record cleaner, $175; JBL 2221 10" speaker, $125/pair; KLH 27 receiver, $100; Sherwood S-5500 IV integrated amp, $50; Dyna PAT-4, $35; EV/Realistic ARS-4 four channel decoder, $35; pop and classical CDs, $8-$10 send SASE, all ppd. Steve Hluchan, 21 Westward Rd., Woodbridge, CT 06525, (203) 397-3888.

Audio Control industrial road case for SA 3050 analyzer, custom designed and made with heavy duty hardware, storage for microphone and cables, etc., brand new, $200; Acoustilog Impulser signal alignment measurement "black box," use with scope and microphone, very accurate, $125. Misc Altec, E-V and JBL components. Tom Young, (203) 274-2202.

Two Morel 224R woofer-mids, closed box Qtc 1.1, infinite baffle, or auto sound, 20 hours use, $50 U.S. postpaid. Mark Thompson, 194 St. Davids Rd. W., St. Catharine's Ontario Canada L2T 1R4.

SAE 180 stereo parametric equalizer, two bands per channel, $80 or trade for woofers, mids, tweeters, crossovers, anything interesting. Rocky Kinnison, (815) 657-8488.

Mobile Fidelity out-of-print rock/pop LPs. Approximately 20 including Dead/American Beauty, CCR/Cosmo, Little Feat/Columbus, Clapton/Slowhand, etc. All factory sealed. Price below market. Amperex 6922 industry rated equivalent to 6DJ8. Gold pins, $4.75 each. Need current address of Worldwide FM/TV DX Assoc. Lowell Thomas, 3405 N. Sixth St., Fresno, CA 93726.

Teac A-6100 MK II 15ips, ½-track, reel-to-reel, like new, $750; Ampex TDK 10½" tape; Pioneer MA-62A 6x2 mixer, mint, $150; SWTP 540 amp, $25 or trade for Dyna SCA-35; 185Hz speaker crossovers, $15. Make offers, this stuff just taking up space. Paul Becker, (509) 327-8859.

Hermeyer ESA-3 parts: oil capacitors, bridge rectifiers, cable, plate/bias transformers, call for details; ESA-3 circuit board set, $15; 70,000μF 25V DC Mallory type CGS, $25 each; Hafler XL-280 chassis, DH-200 rack kit, $20 each; Dyna FM-5 tuner, $75; Soundcraftsman 20-12 equalizer, $100. Nick Mastrobuono, 50 Copperfield, Sarnia Ontario N7S 5K8, (519) 336-9160.

Bedini 25/25 Class A amplifier, ideal for original Quads, $360 or best offer; Stax Sigma headphone with SD 7 box, $440 or best offer. Mark, (408) 244-2886.

dbx 228, $75; Quadpod, rackmount, level controls, $75; Newmark MK II power indicator, 4 channel, rackmount, $50; Sound Organization TT stand, $50; 15 pair X-Terminators, $10/pair. Larry Cartwright, (412) 846-7419.

Two 18" Electro-Voice EVM18B Series II bass drivers in original sealed cartons, $320/pair; two 18" Oaktron M18Y2 guitar speakers, $85 each; two 15" CTS 15W38C woofers, $80/pair; two 12" CTS 12W54C woofers, $60/pair. All items mint, unused. C. Orme, 1027 Valley Rd., Colorado Springs, CO 80904.

Hewlett Packard oscilloscope 120B, $150; Sony ST J75 stereo tuner, $290; Audio Research D100 amp, $575; Kenwood KT7000 stereo tuner, $250. All excellent condition. WANTED: Scott 335 Stereo Multiplex adaptor or similar McIntosh unit. T. Hafkenschiel, (415) 851-2779.

Richter Scale Series III, $250; Infinitesimals, $200; EPI LS841 auto speakers, $185; SEAS CA25RE4X/DC, $45/pair; Gold Ribbon Concepts ribbons, 2 pair sealed; Heath A9 mono integrated amps. R. Omegna, 44 Cold Hill Rd., Morristown, NJ 07960, (201) 543-2971 evenings until 10 p.m.

WANTED

Altec Lansing 800Hz horn and driver and Altec Lansing 15" woofer. Jack Rumora, PO Box 368, Senecaville, OH 43780, (614) 685-2612.

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Acoustat monitors X, 3 or 4 or Koss 1A and information on modifying the Acoustat ES servo amp, schematics, etc. Also electrostatic speakers for parts, panels, transformers, etc. Tom Bourdeau, 10870 Eastcourt Dr., Windsor, Ontario Canada N8R 1E8, (519) 735-9023.

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Sanyo Plus T55 AM/FM stereo tuner. Stanley Grycz, 2935 Crehore St., Lorain, OH 44052, (216) 288-9480.

Pair of replacement diaphragms for Altec drivers model No. 808-8B. Andrew Lewis, 3929 S. Sherman, Englewood, CO 80110, (303) 781-5573.

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 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.

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.

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 monthly *B.A.S. SPEAKER* with reviews, debates, scientific analyses, and summaries of lectures by major engineers. Read about Apogee, Nytal, Conrad-Johnson, dbx digital, Snell, music criticism and other topics. Rates on request. PO Box 211, Boston, MA 02126.

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 started in October 1983 and has regular meetings on the third Sunday of each month as well as special programs with leaders in the industry, such as Mr. William Conrad of Conrad-Johnson and Mr. William Johnson of Audio Research. We are currently looking for additional members in the Southeast. all members receive the minutes of each meeting and program, as well as other relevant announcements and correspondence. For full information and membership packet, write Atlanta Audio Society, PO Box 92130, Atlanta, GA 30314, or call Howard Royal in Newnan, GA, (404) 253-6419.

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.

WASHINGTON AREA AUDIO SOCIETY (N. VA, MD and DC) is looking for sincere audiophiles who are eager to devote their time and get involved with the direction of the society and the publication of a monthly newsletter. Please contact: Horace J. Vignale, 13514 Bentley Circle, Lake Ridge, VA 22192-4316.

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.

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An increasing number of readers are telephoning for information about their personal classified ads. Ads are filed as received, set in type late in the magazine production cycle, and put on page approximately one month before the magazine is mailed. Delivery of the mailed magazine can take from four days to three weeks.

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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.

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.

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.

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. Tonawanda, 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.

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.

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.

ORGAN MUSIC ENTHUSIASTS: If live recordings of fine theatre pipe organs or electronic organs are your thing, I have over two thousand of them. I lend you the music on reels or cassettes. All operation is via the mail. A refundable dollar will get you more information. E.A. Rawlings, 5411 Bocage St., Montreal, Canada H4J 1A2.

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, 2113 Charger Dr., San Jose, CA 95131.

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.

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Illustration depicts FOCAL 280DB Kit. T-120 Tweeter and 7N402DB Dual Coil Driver.

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Fast Reply #HC61

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Fast Reply #HC29

morel

high fidelity range

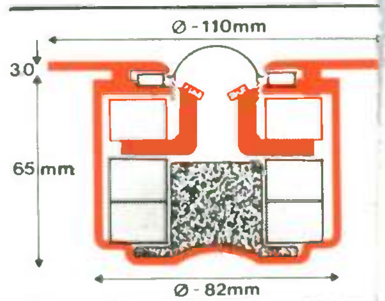
The MDT 33 is an extremely fast Tweeter, using a 28mm (1⅛") diameter voice coil and a chemically treated soft dome, and is ideally suited for two way systems with the possibility of a lower than normal crossover frequency, as well as for three and multiple way systems.

Incorporating the Morel Hexatech voice coil technique, aluminium wire wound on an aluminium former and using flexible wire termination ensures excellent high frequency performance with exceedingly high power availability. The power handling is further enhanced by using Ferrofluid in the magnetic circuit.

The magnetic system itself is an ingenious Morel double magnet design and is completely enclosed. By venting into the enlarged area of the double magnet system, a low resonant frequency of 500Hz is obtained with a remarkably smooth roll off from 1000Hz through this damped resonance area. The subsequent wide range response of 1400-20000 ± 0.6dB is obtained with a harmonic distortion of below 0.8% over the entire range. The distortion figures quoted are with an input power giving an output level of 96dB at 1 metre. The MDT 33 sensitivity is 92.5dB for 1 watt 1 metre, and a power handling capability of from 100 to 500 watt subject to crossover frequency.

With such a dome tweeter design, the acoustic qualities at lower than normal crossover frequencies are excellent with an absence of honking, and even at the more normal crossover frequencies this excellent acoustical behaviour is evident to the ear. With the lower crossover frequency available and high capability, it is ideal for consideration in two way systems using a 10" or 12" woofer.

To utilise the dome at the lower than normal crossover frequency available makes it necessary to have a sharp roll off below 1400Hz of minimum 12dB per octave to protect the tweeter from mechanical damage. This makes it ideal for use with active systems.

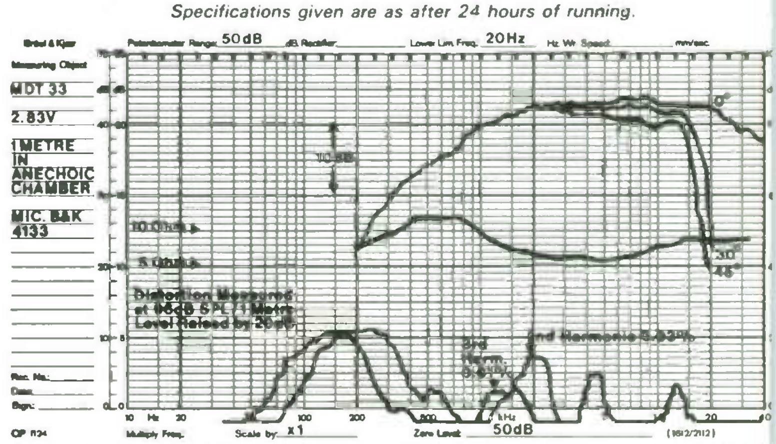


Specification

Overall Dimensions	Ø - 110mm x 68mm
Face Plate Thickness	3mm
Voice Coil Diameter	28mm (1⅛")
	Hexatech Aluminium
Voice Coil Former	Aluminium
Number of Layers	2
DC Resistance	5.2 ohms
Nominal Impedance	8 ohms
Voice Coil Inductance @ 1 Khz	0.09mh
Air Gap Width	0.75mm
Air Gap Height	2.5mm
Voice Coil Height	2.7mm
Flux Density	1.95T
Force Factor (BXL)	4.76 WB/M
Rmec	2.09ns/m
Qms	0.66
Qes	0.38
Q/T	0.24

Vas	0.016 lit
Moving Mass including Air Load	0.44 gra
Effective Dome Area	8.5 c
Dome Material	Treated Fab
Frequency Response	1400-20000 ± 0.6
	(1000-40000 – 5c
Resonant Frequency	500
Power Handling Din:	
X-Over 1400 Hz	100
X-Over 5000 Hz	500
Transient Power 10ms	1500
Sensitivity	92.5dB (1W/1
Rise Time	10
Intermodulation Distortion	
for 96dB SPL	<0.2
Harmonic Distortion	
for 96dB SPL	<0.8
Nett Weight	1.2

Specifications given are as after 24 hours of running.



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