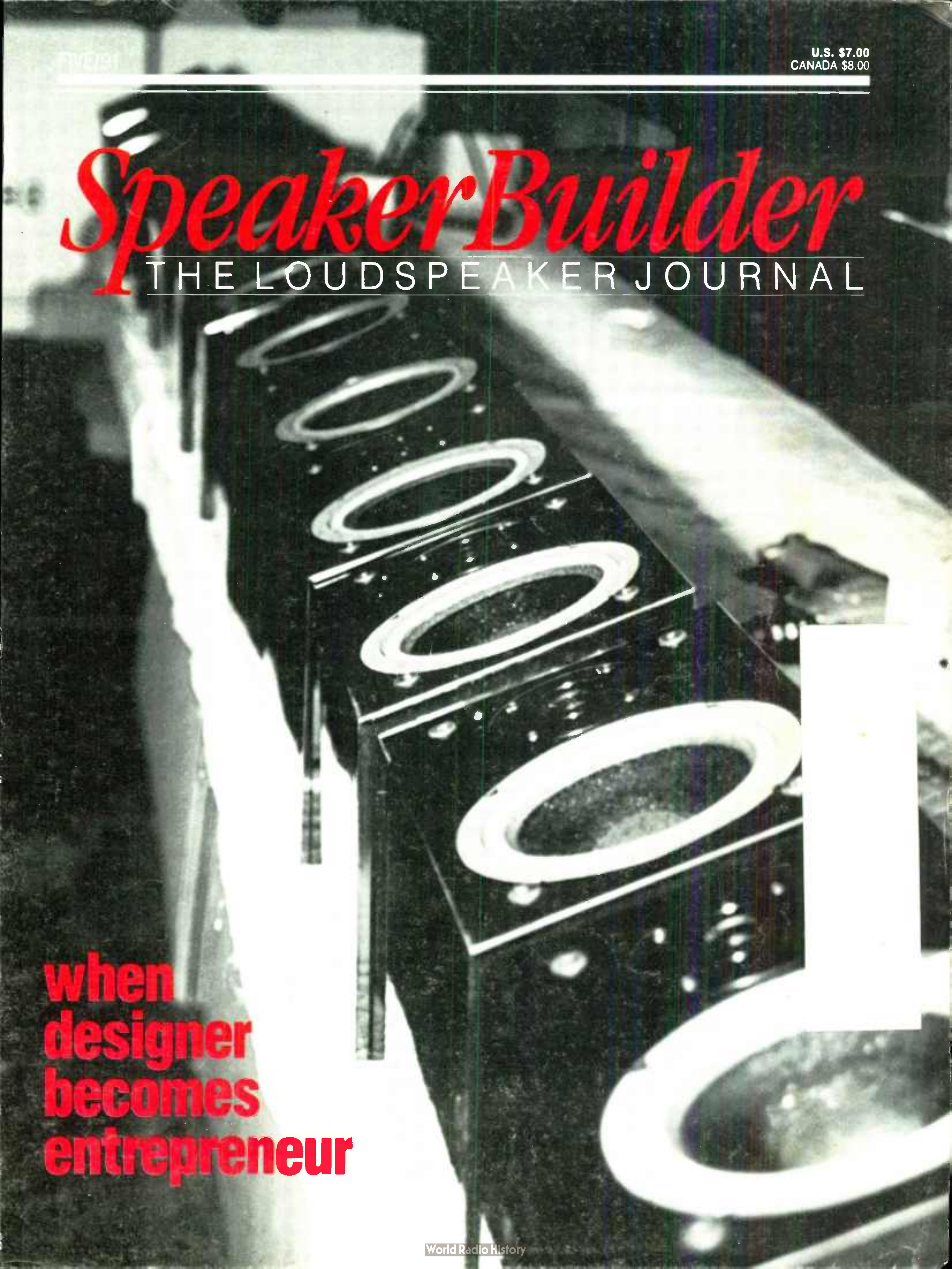
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Good News

AUDIOCONTROL has published a technical paper by Glenn White (author of *The Audio Dictionary*, available from Old Colony) entitled "Small Room Acoustics De-Mythologized." It studies the behavior of sound in real rooms and how the room and materials affect objective and subjective acoustics. Also discussed is the philosophical differences in various styles of recording and music. Finally, the paper shows you how to apply this information to optimize your listening environment.

For a copy of the paper, request technical paper #107 and enclose a 9 by 12 SASE (postage 52¢ from AudioControl, 22313 70th Ave. West, Mountlake Terrace, WA 98043-2165, (206) 775-8461, FAX (206) 778-3166).

SONANCE has introduced several in-wall speakers. The Ambient Imaging System 500s are ideal for use as the main speakers where an environmentally enveloping sound is desired or as the effects speakers in surround-sound systems. Since the drivers are set in the mounting baffle at an angle, the main thrust of sound can be aimed in any direction. The AIS 500s have a frequency response of 60Hz-20kHz ±5dB, a sensitivity of 89dB 1W/1m, a nominal impedance of 6Ω, a minimum power requirement of 5W, and a maximum power of 75W. They sell for $499 per pair, including brackets and grilles.

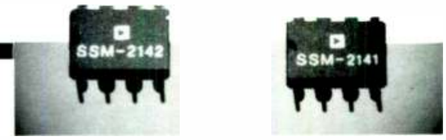
The M30V ($310/pair with brackets and grilles) is an upgrade to the M30 two-way speaker system. It is designed for use in close proximity to a video screen. The system is shielded to prevent interference to the video picture from the magnetic fields present in loudspeakers. It has a frequency response of 65Hz-20kHz ±2dB and a crossover point of 3.7kHz, and it can handle from 5-60W of power. The M30V has a nominal impedance of 8Ω and efficiency is rated at 88dB 1W/1m.

Derived from Sonance's rectangular footprint M10 and II, the M10R and S2R round in-wall speakers use the same drivers as the earlier models. They are intended for ceiling or wall installations where a wide and uniform dispersion pattern is desired. The S2R has a nominal crossover point at 3kHz and a frequency response of 55Hz-20kHz ±3dB. The M10R has a frequency response of 75Hz-15kHz ±5dB.

For more information on the company's speakers, contact Mitch Simon, Sonance, 961 Calle Negocio, San Clemente, CA 92672, (800) 582-7777, FAX (714) 361-5151. From Canada, call (604) 873-4475.

Fast Reply #IF1373

ANALOG DEVICES' SSM-2142 is a highly integrated line driver system for audio, telecommunications, and industrial applications. This monolithic device replaces large, expensive transformer-based solutions and eliminates the need for expensive external trim circuitry. Housed in a single 8-pin miniDIP, the complete SSM-2142 provides complementary differential outputs from a single-ended source, and can transmit 10V RMS signals into impedance loads as low as 600Ω (differential or single-ended), over cable lengths up to 500 feet.



The SSM-2142 is well-suited for a variety of applications that require superior noise immunity and common-mode rejection, including professional studio mix consoles, telecommunications, and industrial instrumentation. Total harmonic distortion plus noise from 20Hz-20kHz is 0.006%; output common-mode rejection is typically 45dB. When coupled with its companion, the SSM-2141 differential line receiver, the device offers a completely balanced low-cost, high-performance driver/receiver system for less than $5.

The SSM-2141 and SSM-2142 cost $1.95 and $2.95 in 100s, respectively. For complete information, contact Analog Devices, One Technology Way, PO Box 9106, Norwood, MA 02062-9106, (617) 329-4700, FAX (617) 329-1241.

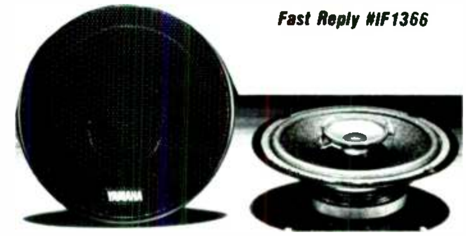
Fast Reply #IF70

YAMAHA ELECTRONICS has introduced two round coaxial car audio speakers. The YCS-6023 is a 6½″ model with a frequency range of 40Hz-20kHz. It delivers better low-frequency response than its predecessor, the YCS-6020. The speaker's sensitivity is at least 89dB and requires a mounting depth of 1$\frac{11}{16}$″. The YCS-6023 sells for $100/pair, including snap-on grilles and speaker cords.

Yamaha's other speaker, the YCS-5021, is a 5¼″ coaxial design. It delivers a frequency range of 70Hz-20kHz, has a sensitivity of at least 89dB, and requires a mounting depth of 1$\frac{11}{16}$″. It sells for $90 a pair, including snap-on grilles and speaker cords.

Contact Yamaha Electronics Corp., USA for information: 6722 Orangethorpe Ave., Buena Park, CA 90620.

Fast Reply #IF1366



← ***Fast Reply #IF29***

CIRCUIT SEARCH has announced Version 1.09 of their database of references to articles containing practical electronic circuit designs. This dBase III/III+ compatible database references nearly 13,000 articles and papers from more than 300 technical and scientific journals and magazines. As an interdisciplinary reference source, it can locate circuits by key words from journals in fields as varied as electronics, astronomy, agriculture, physics, chemistry, nuclear science, education, biomedicine, and many more disciplines where electronics can be applied.

The database is installed on a hard disk on an IBM PC or compatible, and used with either its own menu-driven front end or a dBase-compatible database management program. You can locate references by title, circuit description, device type used, author, author affiliation, journal, date, and more. The database occupies about 7Mb and comes on a series of disks.

Additional services offered by Circuit Search include provision of hard copy (FAX or mail) of articles that qualify under Copyright Clearance Center (CCC) provisions, as well as update and revision services on a semi-annual basis.

Circuit Search sells for $375 (US dollars), including one free semi-annual update/revision. Further updates cost $40 for one or $65 for two. A sample disk including 400 references, pricing, and additional information is available for $7. Contact Peter Sawatzky, Circuit Search, PO Box 268, Breslau, Ontario, Canada, N0B 1M0, (519) 241-1252, FAX (519) 742-4594.

Fast Reply #IF1371

Continued on page 8

MONSTER CABLE has announced the Custom Installer SCIR™ series of cables. They are UL approved Class 2 fire rating for running in walls. The cables consist of the two-conductor, single-channel SCIR 14-2 14-gauge and SCIR 18-2 18-gauge, and the four-conductor dual-channel SCIR 16-4 16-gauge.

The cables' identification system—color-coding of inside and outside jackets—provides for simple error-free hookup. Two color-coded outside jackets provide flexibility and multiple configuration cable hookup. Their low friction jacket makes these cables simple to pull through walls and tight spaces.

Monster Cable also provides a connector compatibility chart as a reference to all its cables and connectors. For information, contact Gary Reber, Monster Cable Products, Inc., (714) 677-4668.

R.F. ENGINEERING has announced the VI series of automatic A/V switches. The VI allows two video devices (a laser disc player and a VCR) to be connected to a single monitor. It detects which source is delivering a signal and routes the signal to the monitor.

The VI can detect either video or audio signals, so it can be used as an expansion input on an audio preamp. The device is designed so that if both signals are present, it will give priority to the device on the Video B connection. A sensitivity adjustment allows the VI to filter out noise that can cause erratic switching. A delay adjustment is used to prevent switching on missing signals, such as between tracks on a CD.

The VI is available with RCA connectors at $89.95 or S-VHS connectors at $119.95. Contact R.F. Engineering, Inc., 9215 Lowell Blvd., Westminster, CO 80030, (303) 430-8281, FAX (303) 430-4023.

Fast Reply #IF1365

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About This Issue

Robert Spear and **Alexander Thornhill** have collaborated on a general discussion of the effects of fibrous tangle in transmission lines, in particular the effects of stuffing on attenuation and the speed of sound (p. 11). In this piece, the authors have balanced theory and practice, and have kept math and formulas to a minimum.

Paul Becker, dissatisfied with expensive solid-state commercial crossovers, has built a simple, inexpensive, "purist" crossover that allows "tube preamp through power amp" signal flow and allows single channel or stereo subwoofing. Turn to page 18 for the details.

Contributing Editor **Bruce Edgar** has re-interviewed Ken Kantor, since his venture into starting his own company, NHT. To gain some insight into a company's birth, turn to page 22.

Disillusioned with the "one-side-of-the-room" acoustics so often heard at social gatherings, **Joseph Demers** designed an omnidirectional speaker. His discussion of his "DOALS" system begins on page 36.

Beginning on page 42, **John Jackson** describes how to construct an easy-to-modify high-quality loudspeaker cable.

Donald Scott's speaker-to-ear interface (p. 44) provides him with a listening situation free of an undercurrent of room sound since his system produces an acoustic mix of 90% direct to 10% reflective sound.

Reviewed is Audio Concept's Little V, which Contributing Editor **Gary Galo** claims is an impressive addition to AC's line of speakers. See his comments beginning on page 46. Also read **David Davenport**'s remarks on page 54 about the new edition of Martin Colloms' *High Performance Loudspeakers.*

On our cover: NHT drivers produced on the assembly line. Photo by **Bruce Edgar**.

SpeakerBuilder
THE LOUDSPEAKER JOURNAL

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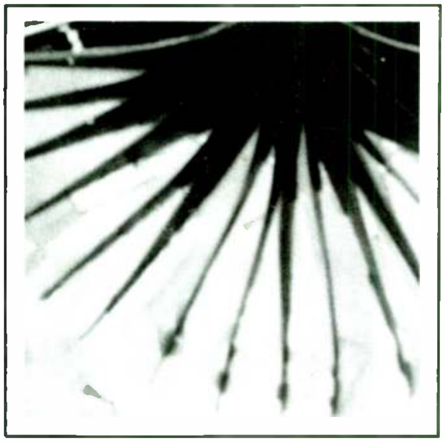

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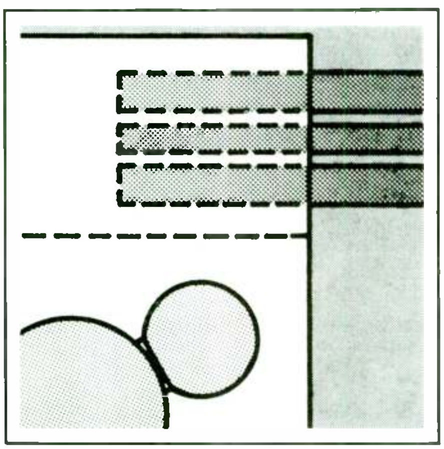

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FEATURES

DEPARTMENTS

Intermezzo, featuring noted Canadian pianist Robert Silverman playing Brahms' piano Sonata in f, Opus 5 and the Intermezzo Opus 117 No. 1, is a limited-edition LP available from ***STEREOPHILE***. Recorded with vacuum-tube microphones and tape recorder, the recording provides a fascinating comparison between early and late Brahms. *Intermezzo* was recorded by the renowned recording engineer Kavi Alexander under the supervision of *Stereophile* editor John Atkinson.

Best known in his native Canada, where he has performed from coast to coast, Robert Silverman has also appeared with the Chicago Symphony and the Boston Pops, and has performed in New York, Washington, London, Paris, Budapest, Hong Kong, Rio de Janeiro, and the Soviet Union. He has recorded for Orion, Musica Viva, and Marquis. His instrument for *Intermezzo* was a majestic 9′ Steinway "D" concert grand.

The recording is $20.45 from *Stereophile*, 208 Delgado St., PO Box 5529, Santa Fe, NM 87502.

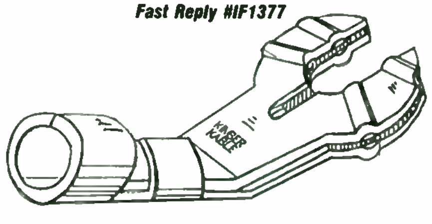
Fast Reply #IF69

KIMBER KABLE has announced a patent-pending connector. The PostMaster™ is a spade-like crimp terminal for use on five-way binding posts. It has radial ridges to concentrate and intensify the contact pressure, resulting in a gas-tight contact area. The PostMaster is spring loaded with a high-rebound, high-temperature sandwiched wafer that maintains dynamic pressure with finger-tight torque. The wafer damps vibration that might exist if the binding post is on a speaker. The high temperature specification assures consistent pressure under changing thermal conditions, such as on a power amplifier.

The connector is available in two sizes; the PostMaster-25 fits post sizes from 0.2–0.265″, the PostMaster-33 fits sizes from 0.28–0.345″. Wire size accepted is gauge 16 through 6. The base metal is cold rolled OFC copper and the plating is the same UltraPlate™ used on Kimber Kable's RCA connectors. It is also available on special order without plating.

The PostMaster sells for $20/pair. Contact Kimber Kable, 2752 S. 1900 West, Ogden, UT 84401, (801) 621-5530, FAX (801) 627-6980.

Fast Reply #IF1377



The FOCUS speaker system is available from **REEL TO REAL DESIGNS**. The speaker uses controlled directivity to improve image resolution. This special driver array minimizes colorations due to floor and ceiling reflections.

Two 7″ midbass drivers with two-layer Kevlar honeycomb construction are strategically separated to provide a null off-axis vertically. A treated 1¼″ woven dome tweeter with a 48 oz. magnet structure operates out to 12kHz where it hands off to a 3″ ribbon supertweeter.

The bass range is covered by three 12″ subwoofers with carbon-filled polypropylene cones. The long-throw drivers offer more than 300 in² of piston area to each channel. The system is tuned to 20Hz via rear ports and is bi-amp capable.

FOCUS is available in walnut, oak, rosewood, black lacquer, teak, and ribbon mahogany and comes with a ten-year warranty. Frequency response is 16–28kHz. Impedance is 4Ω and sensitivity is 96.5dB. Peak power handling exceeds 500W/channel.

The cost of the FOCUS speaker system is $4,450. Complete information can be obtained by contacting Reel to Real Designs, 3021 Sangamon Ave., Springfield, IL 62702, (800) 283-4644.

Fast Reply #IF1372

TANNOY has announced the Tannoy Sixes, a seven-speaker line with four models featuring dual-concentric driver technology. The units have irregular hexagonal cabinets with mineral-loaded copolymer end caps, dual-concentric drivers, a captive bi-wiring system, and internal DMT (differential material technology) bracing.

In the dual-concentric design, the high-frequency driver shares the same chassis as the bass driver, making the entire drive unit operate across the complete audio bandwidth as a single point source. The signal is supplied to each driver by a time-compensated crossover, which aligns high- and low-frequency sound sources to a single point on the same axis.

The discrete high-frequency driver is designed to provide zero surround radiation. Its deep-drawn, extruded aluminum diaphragm is suspended by a nitrile rubber surround. Also included are an integral injection moulded dust seal and piston stiffener, integrated anti-diffraction rings, high-power anisotropic barium-ferrite magnets, and a high torsional strength steel chassis.

The Sixes feature a hard-wired crossover network with a bi-wire terminal panel and integral captive slide-switch mechanism. It uses low-order slopes for minimum phase operation and low-loss inductors.

Bookshelf or stand-mounting models cost from $279–$599 a pair; floor-standing models cost from $799–$1,300 a pair. For more information, contact Tannoy, 141 Linden St., Suite G10A, Wellesley, MA 02181, (617) 239-1692, FAX (617) 239-0096.

Fast Reply #IF1368

Guest Editorial

MY TWO FAVORITE AUDIO TEST INSTRUMENTS

By Sid Tetins

I've subscribed to *SB* since its inception, and I hate to be picky, but the 2/91 cover frightens me. Perhaps "annoys" me would be a better term. To save you digging it out of your library (you do save your *SB*s, don't you?), let me describe it and its associated article and see if you share my feelings.

The cover photo shows an array of 16 10" woofers mounted in what appears to be a plywood and plastic box, mounted in the back of a Chevy van. On pages 48–49 of that issue's "Craftsman's Corner," H. Chris Edmondson describes this system, which he built. Eight 2 × 50W RMS/channel amps drive the woofers. Other amps, arrays of drivers, crossovers, and a CD player complete the system, which, according to Mr. Edmondson, has achieved an SPL of 139dB.

Now, let me describe a couple of audio test instruments I own. Hang on, I realize this may seem confusing at the moment, but bear with me, the pieces of this puzzle will fit together.

My audio test instruments are a matched pair, and, although I've owned them longer than I can remember, they remain at the leading edge of technology, principally because their designer has seen fit to freeze the state of the art at the current level. They are difficult, often impossible to repair, and spare parts are unavailable. Due to these limitations, I strive to keep them in as close to original condition as possible. In fact, they are so valuable to me I carry them with me everywhere I go. They are called "ears," and that brings me back to Mr. Edmondson and his 139dB system.

Sound, particularly at high levels, and its effects have been the subject of much study and debate. A quick trip to any well-stocked library should produce a number of books on the subject. The ones I found came to the same conclusion: that we, as a society, are deafening ourselves by continued exposure to high sound levels. Medical praticioners who specialize in ear problems, and others who have reason to study such things, have assembled impressive empirical evidence to support this conclusion.

This evidence has not been lost on the captains of industry, government officials, or the writers of occupational health insurance. Look around a pistol range, an airport ramp, or a noisy factory floor and you'll see lots of smart folks protecting their precious "audio test instruments" with earmuffs or plugs.

Although ear-damaging sound levels may be on their way out at the workplace, not everyone is getting the message, and there remain plenty of ways for the unwary to render themselves deaf or hearing impaired.

Dancing, that time-honored boy-girl social grace/romantic ritual, has become a contest of endurance at the nightspots in my area. To even ask the lady to dance requires a vocal exercise akin to shouting over a blast furnace. As far as whispering "sweet nothings" in her ear, you might as well forget it, unless you shout. Then the music (I'm using that term loosely) will stop right when you get to the delicate part and she won't be the only one who knows your intentions.

Among other entertainment venues, rock concerts, long noted for their ear-shattering sound levels, have now been joined by country acts in the decibel race. Two shows I attended at our local fair were so loud the songs couldn't be heard; the performances were simply walls of unbearable noise. While this might have been understandable with over-amped, amateur garage bands, these were big names from Nashville, with chart records, Silver Eagle tour buses, and sound engineers—who should've known better.

The audio onslaught continues on the sidewalks and streets with the never-ending battle of the boom boxes. The shoulder-borne variety has now reached the size of a console television, its mass seemingly limited only by the ability of its owner to carry it. Its automotive counterpart, the boommobile, can be heard blocks away. Some states, noting that a driver enveloped by the blasting speakers cannot possible hear emergency vehicle sirens and the like, have passed laws restricting their use. My home state, Florida, has a rather timid anti-boommobile ordinance on the books, but from the continuous parade of thundering, thumping vehicles roaming the Sunshine State, law enforcement is either unable or unwilling to make it stick.

Somehow it strikes me as odd that people will spend a great deal of money, and in Mr. Edmondson's case, a good deal of time and effort, developing audio systems capable of destroying the very instruments of sensory perception that give us reason for having an audio system in the first place. Is it just me or is the tail really beginning to wag the dog?

I used to run a large, computerized printing machine. During one of its periodic maintenance downtimes, one of the technicians and I were discussing the tape decks in our cars. The other tech, overhearing us, said, "I've just got an AM radio in my car. A fancy sound system doesn't mean much to me." He smiled, turned off the hearing aids in his ears, turned on a vacuum cleaner, and returned to his work. How he lost his hearing isn't the point. The point is, must we lose what we have before we value it?

Do not mistake what you've read as a call for more rules and regulations. Government already has its nose stuck way too far into everybody's business. I realize the fact I do not like boom boxes does not give me the right to tell you how to think, or how to conduct your life. A lot of good folks have paid the ultimate price to see to it that that doesn't happen here (or at least it's not supposed to).

However, for Mr. Edmondson and anyone else who enjoys high decibel technology, a couple of parting thoughts. With freedom comes responsibility. If your "sonic impact" renders you hearing impaired, remember that hearing aids are expensive, and I have no spare money (that is, taxes) to buy them for you. Also, sound does not respect property lines; therefore, if you want me to respect your right to listen to whatever you wish, you must respect my right not to listen. In short, your freedom stops where my wallet, and my ears, begin.

FIBROUS TANGLE EFFECTS ON ACOUSTICAL TLs

BY ROBERT J. SPEAR
ASSISTED BY ALEX F. THORNHILL

Acoustical transmission lines (TLs) are among the more recent loudspeaker enclosure designs. Despite their claimed advantages, they have never gained the popularity of vented or sealed systems. One reason might be that TL enclosures must be stuffed with acoustically absorbent fiber material to realize their full design potential. The function of the fiber tangle has not been studied extensively and is not well understood. The purpose of this article will be to investigate the effects of fibrous stuffing on attenuation and the speed of sound, to attempt to understand how it affects a system's performance, and to provide practical applications for the findings.

BACKGROUND. A.R. Bailey, whose 1965 article in *Wireless World* first described the transmission line, sampled the sound damping properties of many substances. Bailey came to prefer long-fiber wool, but apparently performed no exacting tests on the materials he tried.[1] He understood that wool fibers acted as a low-pass filter, but viewed the stuffing more as a means of eliminating internal enclosure resonances and giving the woofer a gentle rolloff, a trait he believed was highly desirable.[2]

In April 1976, a paper entitled "The Use of Fibrous Materials in Loudspeaker Enclosures" by the English aeronautical engineer L.J.S. Bradbury was published in America by the Audio Engineering Society. Bradbury gave formulas for predicting the absorptive characteristics of wool and glass fibers, given their specific gravity, cross-sectional area, and packing density. He showed that fiber delayed sound differently at various frequencies. His simplified equations did not account for tapered lines and were still too complex for most amateur speaker builders. To our knowledge, no works similar to Bradbury's have yet appeared.

SOUND IN AN EMPTY TL. To understand a stuffed TL, it is helpful to know how it functions empty. For this discussion we will describe a TL as an untapered duct of a certain length and cross-sectional area that has a driver mounted at one end. The moving cone excites the air in a series of compressions and rarefactions which move longitudinally down the line. The enclosure dimensions of most TLs are small enough to prevent the sound wave from expanding spherically, so the wave within the duct is a plane wave (a wave with a flat front).[3]

When the line terminus is blocked, the sound waves are reflected back toward their source. Upon reaching the woofer most of the wave energy is reflected forward again, augmenting the energy from the next push of the woofer. This process continues until an equilibrium of sound level is reached.[3]

When the line terminus is open, you expect the sound simply to flow out of the line and into the room, but this is not what happens. Because air in the room has a different impedance than air in the pipe, a portion of the sound wave reflects back up the line and can cause a standing wave to arise. Standing waves occur freely at frequencies that are multiples or divisions of the line length and cause nulls and resonant peaks. The process of augmentation and reflection occurs even though the terminus is open, with some of the energy reflecting back up the line and some escaping.

Acoustical attenuation occurs in an unstuffed line mainly due to viscous drag (friction) along the enclosure walls and absorption by enclosure panel resonances, particularly at bends.[3] Air alone is not a strong attenuator of sound. The speed of sound normally does not change in air, but within a duct other factors can come into play. Even the texture of the enclosure walls will affect the passage of sound. More than a century ago, Kundt observed the tendency of waves in a duct with roughened walls to slow down as the frequency decreases.[4] In a smooth walled plastic tube, we noted that low frequencies passed through relatively unaffected, while high frequencies appeared to be slightly slowed down and attenuated.

Low frequencies are most confined by the enclosure and thus are best guided through the line. At higher frequencies and shorter wavelengths, the enclosure affords more room for the wave to expand spherically. A path with many diagonals can result in an apparent reduction in the speed of sound because the path to the terminus is longer than a straight path. The diagonal paths also

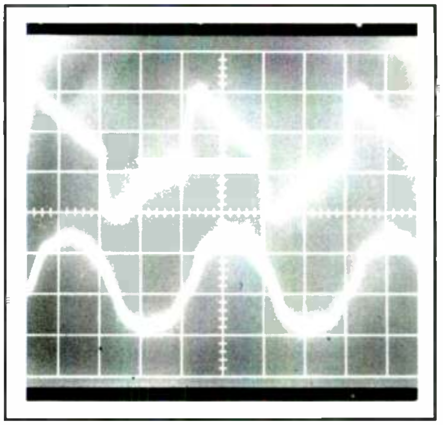


PHOTO 1: Dual oscilloscope trace. Upper trace shows a square wave at 146Hz taken with the microphone 5mm above the woofer dust cap. The lower trace is at the same frequency amplified 4x with the microphone flush with the line terminus. Screen divisions show 1mS/cm.

present many more points at which the air can encounter the enclosure walls and afford more opportunity for the sound wave to lose energy.

EFFECTS IN A TL. When a fibrous tangle is stuffed into a TL, it acts as a filter. The fiber's characteristics and packing density determine the filter's symmetry and its degree of effectiveness. Two consequences arising from stuffing are the attenuation of acoustic energy and a reduction in the speed of sound. Both are frequency-dependent and vary in proportion to each other.

Attenuating acoustic energy in the line damps resonances and harmonics standing in the pipe and diminishes internal reflections, thereby reducing impedance peaks in the woofer. The filter's acoustic attenuation is greatest at higher frequencies and least at lower frequencies. Reducing the speed of sound in the line modifies phase point relationships depending on frequency, and delays the arrival of sound at the line terminus. The filter's delay of the sound is greatest at lower frequencies and least at higher frequencies.[1] The twin effects of energy attenuation and speed reduction are distinct, but not inseparable, and at low frequencies it should be possible to manipulate their relationship to advantage.

SPEED OF SOUND THEORY. It is claimed a stuffed TL slows the speed of sound such that a quarter wavelength line behaves as if it were three quarters of a wavelength long. Sound propagating at 1,130 feet per second (fps) must then be slowed to 282.5 fps, or conversely, if sound takes 7.9mS to travel a 9′ line, it should take three times longer (23.7mS) over the same distance. A lingering misconception is that the reduction in speed is uniform at all frequencies, a notion that has led many TL builders astray.

In his second *Wireless World* article, Bailey stated the speed of sound in the line was slowed by a factor of 0.7 to 0.8.[5] It was not clear to us whether Bailey meant the speed was reduced *by* 20–30% or *to* 20–30%, nor did he explain how he measured this phenomenon. Bradbury's paper was ambiguous on the subject. His formula showed the speed of sound falling under 300 fps below 50Hz, but this result is theoretical.[1]

Bradbury states that sound waves traveling through a fibrous tangle are subject to aerodynamic drag proportional to the air velocity across the fibers, and the slow air velocities arising at low frequencies change the behavior of the fiber tangle.[1] The fibers appear stationary to the sound wave at high frequencies, but at low frequencies they have time to couple to the air. The mass of the fibers adds to that of the air, creating a denser medium.[1] Under these conditions, the speed of sound can be reduced with little loss of acoustic energy.

ATTENUATION THEORY. Lore has it that a TL absorbs all the woofer's rear energy. The idea has emerged that a transmission line is the ideal expression of an infinite baffle where the cone behaves as if it were in a line of endless length, an ideal environment at all frequencies. Unfortunately, the cone is not so easily fooled. Bailey constructed an 8′ line with a pivoted flap at the terminus. When the line was stuffed and the flap closed, the expected damping of the woofer cone did not occur. Cone excursion doubled for each halving of the frequency and bass response weakened. Opening the flap restored the bass, diminished diffraction effects, and reduced cone excursion at low frequencies.[2]

The fiber's attenuation potential at a given frequency remains fixed. If a certain amount attenuates by 10dB, a 90dB woofer input produces an 80dB port output. At 50dB input, the attenuated port output is 40dB, again 10dB less. Higher energy levels in the line might set the

Continued on page 14

Appendix A

If you have not seen Bradbury's formulas, we present the pertinent ones here. His fiber diameter and density for fiberglass and wool have been replaced with Hillman's and Bullock's values for Dacron. All arithmetical operators are separated by spaces to help distinguish the symbols for division (/) and subtraction (−) from slashes (/) and hyphens (-) used in the abbreviation of terms. *Table A* lists the parameter symbols and values.

The first value to be determined is λ, the drag parameter. Bradbury seems confident the value for the exponent (n) is 1.4, but the value for A is less certain. Investigations by Bradbury and others have suggested a range from 12–50 with an average of 27, which Bradbury used in his paper.

The drag parameter formula is:

$$\lambda = A \times (\mu / d^2) \times (P / rhoF)^n$$

λ = 1010.8758 kg/(m³-sec) for our highest packing density of 6.694kg/m³ (0.417 lbs./ft³). A value of 27 is used for A. Using the extreme values for A produces λ values of 449.278 and 1871.99 for this packing density.

Bradbury expresses the behavior of sound in a fiber tangle as a complex number, and his equations are written to separate the real and imaginary parts. The real part, α, represents the velocity of sound in the tangle, and the imaginary part, β, represents the attenuation. We have used only the α equation which yields the velocity in air relative to the velocity in the fiber (V_A / V_F). However, probably for ease of graphing, Bradbury expressed this as V_F / V_A, which equals 1 / α, where:

$$\alpha = (((1 + P / rhoA)^2 + (w \times P / \lambda)^2) / (1 + (w \times P / \lambda)^2))^{0.25} \times \cos(\Theta)$$

$$w = 2 \times \pi \times frequency\ (f)$$

and

$$\Theta = .5(\arctan(w \times P / \lambda) - \arctan((w \times P / \lambda) / (1 + P / rhoA)))$$

Various values for A were tried, seeking to match the calculated curve to the curve of the data points. The best fit for the number of tries made was obtained for A = 50. Our calculated velocity ratios do not coincide with the measured values, the latter being slower. *Figures 1a* and *b* show the curve and data points (as ratios of velocity in fiber (V_F) to velocity in air (V_A)) for the set of data points shifted up in velocity and for the actual values.

The data points were shifted up to make the comparison between theoretical and measured data easier. Our figures show the alignment of measured and calculated values at least equal to or possibly better than the corresponding figures in Bradbury's paper. The shape of the curve seems to be in satisfactory agreement even though the absolute values do not coincide.

TABLE A

PARAMETER SYMBOLS AND VALUES

PARAMETER	SYMBOL	VALUE
Fiber diameter	d	0.0000225m
Fiber density	rhoF	900kg/m³
Air density	rhoA	1.11kg/m³
Air viscosity	μ	0.0000181kg/m-sec
Stuffing density	P	in kg/m³
A constant	A	
An exponent	n	

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64	**Emminence** 10" paper, Fs 23, Qts .33, Vas 269 Ltr., 92db 4Ω	$25.00
450	**Dan Mar** 12" sealed box or replacement woofer, 1" vc, 50 watts, priced below 1960 level!	$12.00
215	**Peerless** 1759 10.5" vented box long throw poly woofer, Fs 22, Qts .34, Vas 125, Xmax 8.5mm peak, 4 Ω only.	$45.00
200	**Emminence** 12" Poly, Fs 24.8, Qts .32, Vas 209 Ltr 93 db 8Ω	$25.00
360	**Famous Maker** 6.5" sealed box paper cone woofer, Fs 69, Qts .55, Vas 10 Liters.	$7.50
104	**Dynaudio** 30W100 8Ω, 4" voice coil, 12" poly woofer with foam surround, Fs 20, Qts .48, Vas 409 Liters The very best!	$175.00
145	**Vifa** D25AG05 aluminum dome tweeter, ferrofluid cooled, 90db, cutout 3 1/8". Try a metal dome at a modest cost.	$14.00
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Continued from page 12
fibers into motion sooner (at a higher frequency), and this could also influence the system's sound characteristics.

MEASURING SPEED OF SOUND. We could not make direct measurement of sound velocity, so we decided to measure delay in the line instead. We used a dual-trace oscilloscope, two matched mikes, and an audio generator set to produce a square wave at 15Hz. The onset transient of the wave was audible as a sharp click which triggered the scope. We superimposed the wave from a port microphone over a second wave from a microphone at the speaker and used the oscilloscope's screen grid to obtain a reading of the delay in milliseconds.

It takes 7.9mS for sound at 1,130 fps to travel 9′ in air. After tests on optimally stuffed lines, we measured delays only 1 or 2mS greater. We also encountered other problems. We could not use the port mike to trigger the scope because the fiber filtered out the spike. We then realized the spike did not even contain audio information at 15Hz, but merely occurred 15 times per second. Whichever part of the spectrum reached the mike first appeared on the scope. This did not give us the greatest delay in the line, but the least—and at an unknown frequency.

After thinking it over and re-reading Bradbury, we measured phase points every 90° from 1kHz down as far as we could. Our intention was to stuff the line gradually and take readings after each stuffing so we could track the shift of phase points. From these phase points plotted on a graph we could then compute the speed of sound.

We constructed a special line for this purpose, the 96-SP-X, a straight 9′ section of 6″ diameter schedule 35 gasketed sewer pipe. On one end we added an 8″ diameter elbow section in which we mounted an 8″ woofer, a Radio Shack 40-1024.

Once again our initial discoveries were not what we had expected. At higher stuffing densities, we often got no measurements at all (see following section on attenuation). At low frequencies, accurate measurements became extraordinarily difficult. We were always amazed at the consequences of even a light stuffing. The reduction in the speed of sound at 0.238 lbs./ft³ appeared so great at first we became convinced our data was inaccurate. We restuffed the pipe and measured again, obtaining nearly the same results.

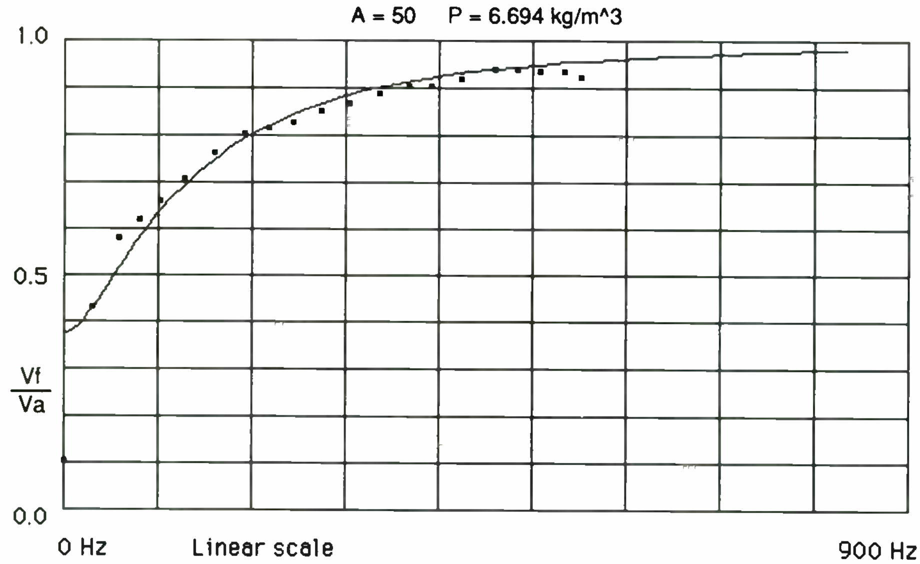

FIGURE 1a: Theoretical (solid line) vs. measured data (black squares) plot for the speed of sound in a stuffed TL with measured points shifted up for easy comparison of curve shapes. See Appendix A for values of A and P and explanation of ratio V_F/V_A.

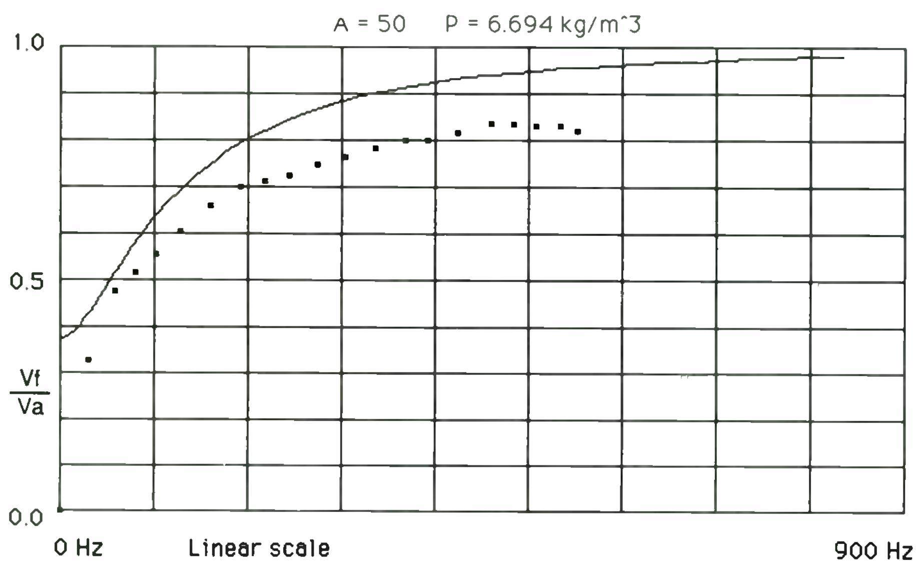

FIGURE 1b: Same data as *Fig. 1a* with black squares in their measured position.

After our tests were completed, we had to determine whether or not our data resembled Bradbury's theoretical projections. He had measured only wool and fiberglass, and his equations reflected their characteristics. We were measuring Dacron, which is somewhat different. Using the constants for Dacron given by Bullock in his computer TL modeling program[6] and the information presented by Hillman,[7] Alex modified Bradbury's equations to fit Dacron fiber.

Figure 1a shows the speed of sound versus frequency for Dacron according to the modified formula. Our measurements at 0.417 lbs./ft³ are shifted up to lie on the plotted line. The fit is acceptable as far as the general shape of the plot is concerned.

Figure 1b shows the actual relationship of measured data versus theory. The measured speed of sound at 500Hz is about 12% slower than projected, but Bradbury's measured data tends to lie below theory at high frequencies and above it at low frequencies.[1]

At 0.417 lbs./ft³, the speed of sound at 500Hz was about 950 fps, while at 34Hz it was just a bit less than 373 fps. In theory the speed of sound in a stuffed line continues to fall as frequency decreases, but we were unable to confirm this with our present techniques. The accuracy of our results must be weighed against the difficulties we encountered obtaining data and the necessity of guessing at certain values in the equations. For simplicity we omit discussing those estimates

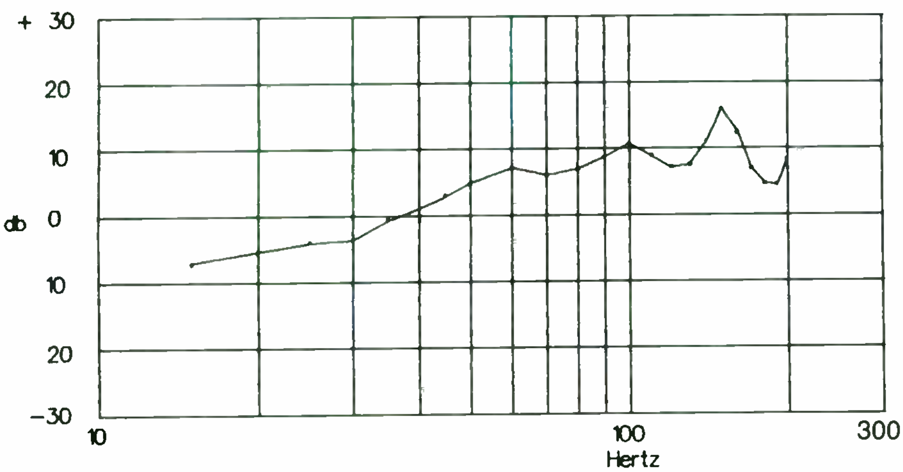

FIGURE 2: Port output (unstuffed) plot of 96-SP-X, a 9′ line constructed of 6″ diameter gasketed sewer pipe.

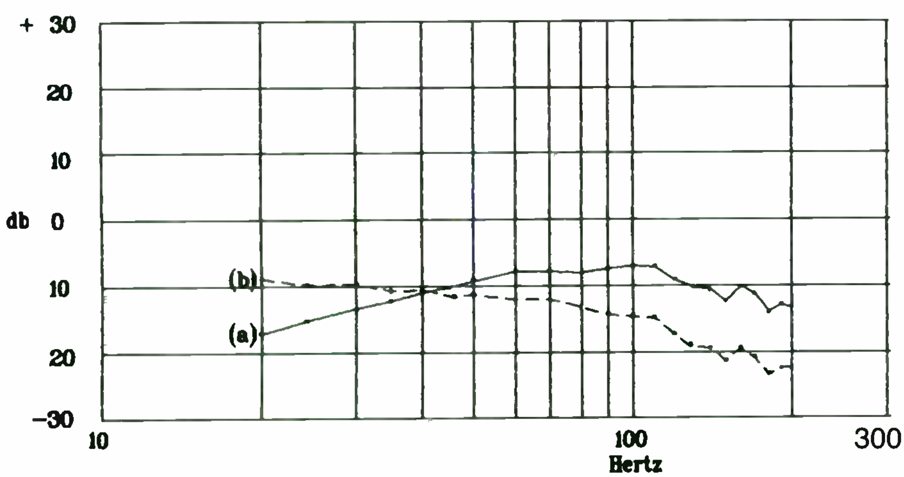

FIGURE 3: (a) filter attenuation with woofer voltage constant; (b) filter attenuation with woofer amplitude constant.

here and refer you to Appendix A for amplification.

The reduction of speed within the fibrous tangle does not depend on line length or the actual amount of stuffing, rather it is the stuffing ratio which governs this effect. If sound travels at a constant speed through an unchanging medium, the speed of sound entering the fiber mass must change over a small distance. Once within the fibers, the velocity remains the same whether the line is 1′ or 10′.

We have not yet managed to record sound at 282 fps at system cut-off, but we came close. We and others have found that significant reductions in the speed of sound can be achieved at modest stuffing densities. Further reductions are possible with small increases, but even a few extra ounces of fiber are often enough to overdamp highly compliant woofers. The old rule of thumb that it is better to overstuff than understuff must be taken with a grain of salt.

MEASURING ATTENUATION. We used two closely matched microphones to take measurements using near-field techniques. The speaker mike was suspended 5mm over the woofer's dust cap, and the port microphone was centered at the mouth of the exit in line with the enclosure frame. The TL system was placed on its back atop a rectangular absorbent surface (the Queen-sized mattress in my guest room) to avoid floor reflections. A tall heap of soft fabric was placed between the two microphones to isolate them from each other, a precaution which was not wholly effective.

Sine waves were read from 200Hz–15Hz in 10Hz decrements except below 50Hz where 5Hz decrements were used. We did not try to measure attenuation above 200Hz often because room reflections began to interfere with our readings. The port output of the 96-SP-X unstuffed is shown in *Fig. 2*.

Sound decays exponentially (a linear rate expressed in decibels) in a fiber-filled line.[1] The measured frequency should be above 200Hz where attenuation is the dominant factor. Even four ounces of fiber (0.086 lbs./ft³), an amount so small it nearly disappeared in the line, produced measurable attenuation. Above 200Hz it was not unusual to measure a drop of 5–8dB in peaks at the port at this density.

By the time the stuffing densities reached about 0.25 lbs./ft³, attenuation rates of 15dB were not uncommon. The volume of sound from the port was so diminished that woofer sound predominated, and consequently the frequency of our highest usable readings decreased with each successive stuffing. We might have done better if the speaker end of the apparatus had been in one room and the port end in another with the pipe passing through a tightly sealed hole in the wall, but my normally supportive spouse drew the line at this.

The descent of the high-frequency stopband was skewed in shape indicating that the higher the frequency, the greater the attenuation. At a stuffing density of 0.417 lbs./ft³ the effect of the fiber above 500Hz was considerable. At the line terminus, portions of the acoustical spectrum were unmeasurable and attenuation rates were beyond the 50dB range of our equipment.

The attenuation curve of the fiber-stuffed pipe is shown in line a of *Fig. 3*. Note that the port output peaks near 100Hz. This seems to indicate attenuation at 100Hz is weakest and becomes stronger on either side. However, such a curve does not show the true symmetry of the filter.

The output to the woofer was 0.6V at all frequencies. Woofer rolloff began between 120 and 140Hz, and the decline appears as a component of the port output. We calculated a second series of port output plots to correct for woofer rolloff and to show the actual attenuation values of the fibers at frequencies below 200Hz. The calculations are described and some examples given in Appendix B.

The solid line in *Fig. 3* (a) shows a generalized filter symmetry for a round pipe stuffed with Dacron polyester when the voltage level to the woofer is constant. The dashed line (b) shows the shape of the filter in the same pipe when the decibel level is constant.

LOW-PASS FILTER EFFECTS. The low-pass characteristic of the fiber helps to explain the good bass performance of a TL. If a pitch fundamental lies in the passband, it will pass through relatively unchanged while its overtones will be attenuated according to the slope of the filter. Since musical instruments are distinguished by their harmonics, their identity becomes obscured as their overtones are filtered out. The low frequency information reaching the line terminus will be featureless. In other words, it would be difficult to tell a tuba from a string bass.

If a pitch of 36Hz were present in the line, the third harmonic would be sounding around 146Hz. The top trace in *Photo 1* shows a 146Hz square wave taken from a microphone at the front of the woofer. The bottom trace shows the same wave taken from another microphone at the port. The stuffing density is 0.25 lbs./ft³. Note that nearly all the characteristics of the square wave have been filtered away, and the wave is now sinusoidal.

The attenuation caused by such a small amount of fiber is a graphic demonstration of its capabilities. This filtering effect should go unnoticed by the listener since the front wave of the speaker supplies the missing information. When conflicting higher harmonics are absent and substantial low frequency energy is available at the port to augment fundamentals, the resultant bass output from the system tends to be very strong, clean and well-defined.

Appendix B

The following explanation and *Table A* show how we corrected for the filter slope once the woofer began to roll off. Readings shown were taken from microphones at the port and the woofer. The readings are in decibels and are signed either plus or minus.

To obtain the corrected attenuation results, the woofer output is subtracted from the port output. Subtracting a negative number from another negative number gives a less negative result (a smaller figure) because it subtracts attenuation; subtracting a positive number from a negative number gives a more negative result (a larger figure) because it adds to the attenuation.

The corrected difference between the woofer and port shows how much the fiber attenuates at a given frequency. Note that as the woofer output falls, the port output also falls. However, the corrected response rises, consequently showing less attenuation.

TABLE A

READINGS FROM THE 96-SP-X*

FREQ. (Hz)	WOOFER (dB)	PORT (dB)	CORRECTED (dB)
200	+9.26	−13.10	−22.36
190	+9.48	−12.85	−22.33
180	+9.19	−14.10	−23.29
170	+9.57	−11.20	−20.77
160	+9.20	−10.12	−19.32
150	+8.86	−12.40	−21.26
140	+8.77	−10.70	−19.47
130	+8.77	−10.23	−19.00**
120	+8.19	−9.00	−17.19
110	+7.70	−7.00	−14.70***
100	+7.39	−7.10	−14.49
90	+6.54	−7.30	−14.27
80	+5.28	−7.97	−13.25
70	+4.45	−7.80	−12.25
60	+3.65	−7.70	−12.02
50	+2.08	−9.07	−11.15
45	+1.18	−10.30	−11.48
40	−0.42	−11.10	−10.68
35	−1.48	−12.23	−10.75
30	−3.80	−13.40	−9.60
25	−5.40	−15.18	−9.78
20	−8.30	−17.10	−8.80

* A 9′ line stuffed with Dacron polyester fiber at 0.417 lbs./ft³ showing the attenuation of sound.
** Woofer rolls off.
*** Lowest port reading.

MIDBASS FADE. In many TLs the midbass output seems to recede. Authors have called this "midbass recision" or characterized it as the "lean and mean" midbass sound of a TL. The cause of this infamous trait could be due partly to the fiber stuffing. As stuffing densities increase, the woofer's rolloff point moves higher in frequency while the attenuation corner of the filter tends to move lower. The ear is generally insensitive to this until the woofer has faded between 1.5 and 3dB, a point lying around 70 or 80Hz in our 9′ lines and near the attenuation corner of the filter. If woofer output is falling from 140Hz and the system's bass lift manifests itself best below 70Hz, a natural dip occurs between these two points.

SUBWOOFER VARIATION. One way we manipulated the fiber tangle to advantage was in a small three-way TL where we rolled off the woofer at 150Hz at 24dB per octave. We were then free to manipulate the fiber type and stuffing density to allow for the best low bass output. We could ignore the performance of the fibers in the upper ranges (2–3kHz) where the woofer would still be operating in a two-way system.

CABINET BOOM. Our lines are intended for low and moderate volume levels, so we employ lighter stuffing densities. We have been able to induce low-frequency booming in our systems at high loudness levels. Because TL enclosures tend to be inherently well-braced, most of the booming is due to cavity resonances and can be controlled by judicious stuffing. While increased packing densities can restore clean performance at high levels, at low levels they can overdamp compliant woofers similar to the type we used in our lines. The overdamping was plainly audible to the listeners and apparent in our test measurements.

IN CONCLUSION. Transmission line theory and design have not progressed much since the time of Bailey's original article. Reading Dickason will underscore how small is the body of information available for TL designs compared to that for vented and sealed enclosures.[8] Speaker manufacturers have avoided the TL format, concerned about the costs and difficulties of cabinet fabrication and probably convinced TL performance lags behind the others.

The home constructor can find rich opportunity in this period of neglect. With a little effort he can build a unique high-performance system that will have no commercial counterpart. The design of TL systems is one where both trial and error and intuition can still produce great leaps forward. Alex and I continue to explore various aspects of TL performance, and we hope you will do the same. We look forward to hearing about your results.

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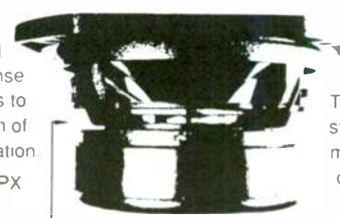
ABOUT THE AUTHORS

Robert J. Spear holds a Master of Science degree in music from Ithaca (NY) College and began his musical career as a bass player and teacher. Since 1980, he and his wife have lived near Washington, DC, where they hand build violins, violas, and cellos for professional musicians. Bob is an active member of the Catgut Acoustical Society, an international organization devoted to researching the acoustics of violin family instruments. An avid audiophile, his current preoccupation is trying to coax deep bass from small woofers.

Alexander F. Thornhill holds a degree in electrical engineering from the University of Maryland. He is semi-retired from a career at the Naval Research Lab in Washington, DC. Alex designed the radio transmitter for the Vanguard satellite, and has also worked on radio navigation devices and AF filters, among other things. His interest in audio predates stereo, and he divides his listening time between his "old" mono system with an RCA duo-cone speaker mounted in a Karlson enclosure with his "new" system, an AR-1 paired with an AR-3.



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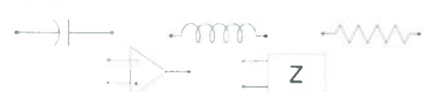

QUASI-ELECTRONIC CROSSOVER

BY PAUL BECKER

During my years as a music major, I developed a keen avocation for recording musical events and the equipment necessary to reproduce that sound at home. As I began to use hi-fi gear, a friend got me interested in building my own speakers. Then, about the time Sony introduced V-FETs, another friend introduced me to a Mac-60, which I tested side by side with the Sonys at the local hi-fi emporium. Despite sales hype regarding the V-FET technology, something in those tubes brought me closer to my aural remembrances of live performances.

Since that time, I've experimented with speaker designs and have built and modified Dynaco tube amplifiers and the crossovers that go with the various configurations. I've preferred to use tube electronics for their virtues in the midrange and high end, and solid-state amplification for the bass.

We start talking about bi-amplification. But what do we do for an affordable crossover? Commercial ones are usually solid state—negating the purist approach to tubes for the higher frequencies. Tubed crossovers were beyond my budget. With this article, I will present a simple crossover you can build for less than $50. All the parts are available at Radio Shack. It allows purist "tube preamp through power amp" signal flow and allows single channel or stereo subwoofing.

UNSATISFACTORY METHODS. As background, here are a few methods that were not satisfactory to me.

1. The basic passive crossover with chokes and capacitors between the amplifier and subwoofer/satellites robs power, does not allow tubes for HF and solid state for woofing, and is mushy overall.

2. *Figure 1* represents a method in which the stereo signal can be summed at the 16Ω taps of the tubed power amps with R1 and 2; R3 and C1 provide the low-pass crossover and R4 supplies the level control to feed the bass amp. While this method works, again, it is mushy. You are going through those transformers to feed the bass amp, when you really want the signal to be tight. If your satellites are basically okay and you need only a little bass fill, you might like this approach, since the satellites run full range.

3. You can modify method two with a passive crossover between preamp and power amps (*Fig. 2*). C1 and 2 provide the high pass for the treble amps, and R3 and 6 the level controls. Although this method worked out sonically for me while I had stereo subwoofers, you must design it specifically around the input impedance of the amps you are using. It has signal loss and getting the system balanced is a tedious process.

DILEMMAS SOLVED. *Figure 3* shows my current bi-amping crossover. I use a pair of Dyna MK IVs for the satellites and a single SWTP Tiger .01 for the center-channel subwoofer. I modified the Dynas according to the instructions in my previous article (*GA* 2/89, p. 8) and beefed up the Tiger with considerable power supply capacitance.

C1 and 2 are the high-pass filters for the MK IVs. It's completely passive, rates about as pure as possible using tubes from pre-to-power amps with no extra amplifying stages, and has no signal loss. This is where the tonal quality and clarity come from, that which defines the essence of the sound.

R1-3 and IC1 provide unity-gain stereo-to-mono summing. I chose 470k because that is the MK IV's input imped-

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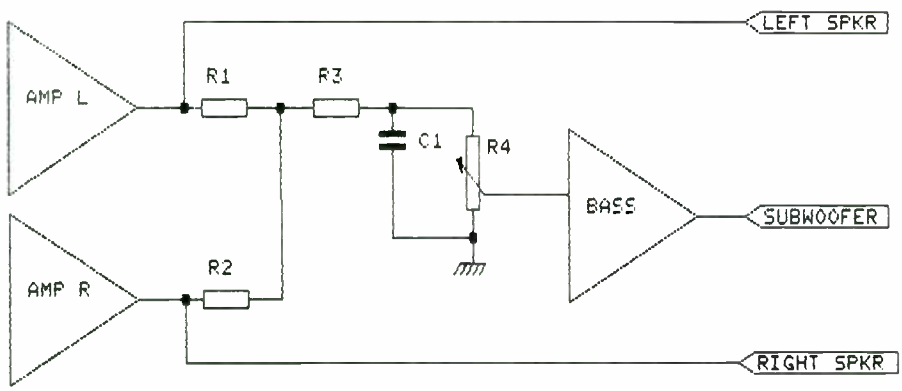

FIGURE 1: Passive center-channel crossover.

TABLE 1

PASSIVE CENTER-CHANNEL CROSSOVER PARTS LISTS

Resistors	
R1, 2	600Ω
R3	1.8k
R4	50–100k
Capacitor	
C1	1μF

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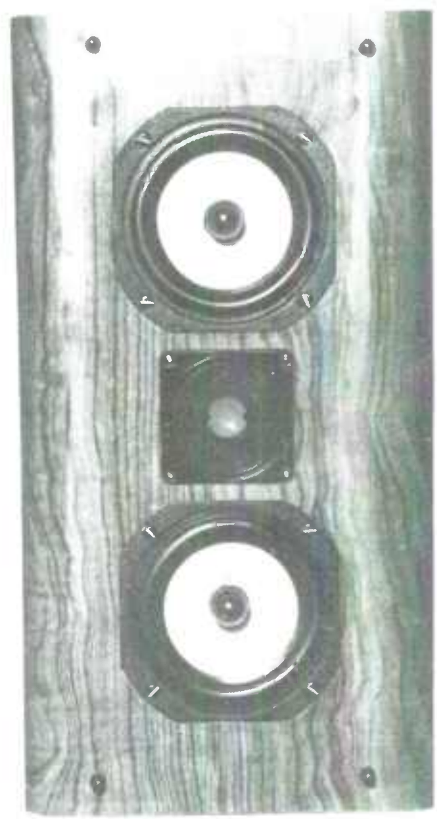

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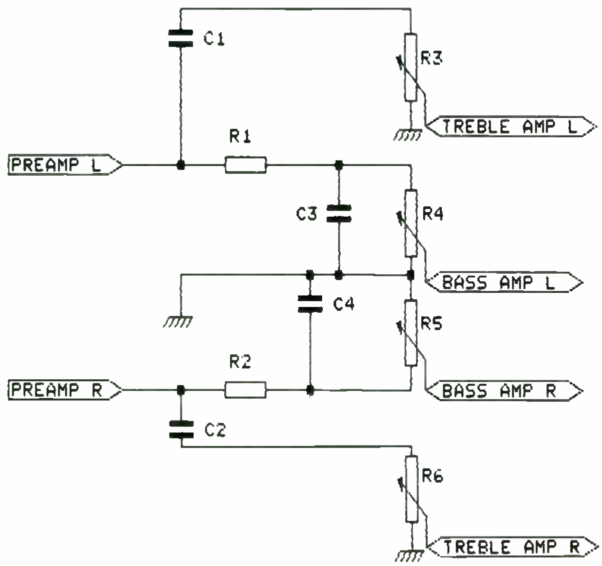


FIGURE 2: Passive stereo crossover.

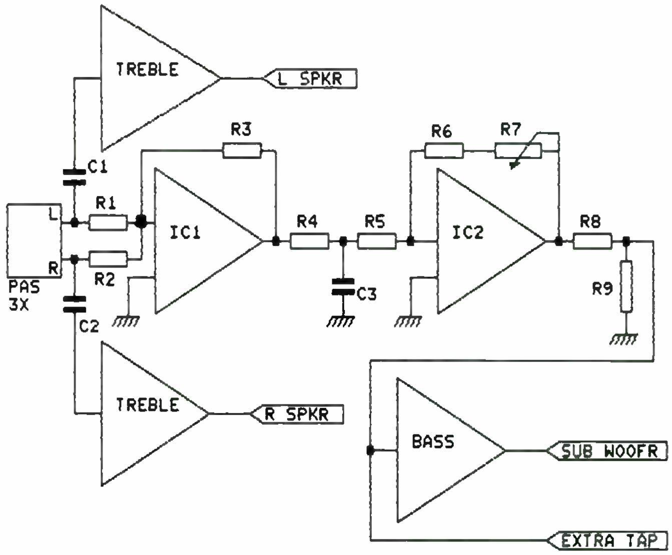


FIGURE 3: Quasi-electronic crossover.

Continued from page 18

ance; it provides maximum load impedance for the PAS 3X. A larger value could be detrimental in terms of RF noise.

R4 and C3 supply the low-pass filtering for the bass amp. Since I was aiming for a 90Hz crossover, the values of RC come from the formula T = RC, where T for 90Hz is 1,800µS. (*Table 3* shows other frequencies.) I used the same formula to determine C1 and 2 related to 470k input impedance for the MK IVs.

The circuitry surrounding IC2 provides a buffer and "versatility" stage. It brings the signal back to the polarity at which it entered the crossover. R9 provides a pad in case the subwoofer system is more efficient than the satellites. Keeping R5 and 6 the same value allows for unity gain. R7 gives versatility if the bass amp or speaker lacks sufficient efficiency, making a signal boost necessary.

TWEAKING. You can adjust all these values according to your application and needs. For instance, if you're using two subwoofers, you can eliminate the stage surrounding IC1 and make two IC2 stages. If you're using an IC that is faster than the 741, you can add filter caps in parallel with R3 and R6 to hold down potential oscillations.

All the ideas presented here provide a 6dB/octave slope. After considerable reading and experimentation, I decided that a center-channel subwoofer would provide the necessary bass fill if crossed over at 100Hz or less. A 6dB/octave slope allows a lot of bass information to be carried by the satellites for directional cues, and yet is sufficiently rolled off to keep mid and trebles clean. A single bass source is easier to position in the room for tight bass. Using 6dB/octave also allows a smooth transition from subwoofer to satellite.

You might need to fine-tune the values for C1-3 and R4 to get a smooth response from the crossover. Since I also own a TEAC 2340SX four-channel recorder, the VU meters provided a convenient means of measurement. While the formula T = RC might provide ideal component values, tolerances will vary. Since my signal generator is a SWTP and does not provide perfectly flat output, one of the VUs can be used to verify its output into the crossover. Amplifier outputs can feed two more VUs. As you plot various frequencies surrounding the crossover point, you can adjust the generator's output to provide flat response (as viewed by the VU meter).

CONCLUSIONS. Provided all the meters track similarly, you can get a fairly accurate *simultaneous* measurement. At the crossover point, both outputs should be about 3dB down from "flat." As you get farther away from the crossover point, one amp will become flat, while the other's response drops off. I added 2+4, 1+5, 0+6, and so on. My first attempt had a hump of about 0.5dB, so I changed C1 and 2 from 0.004 (which the formula yielded) to 0.003. As I began, the crossover was 80Hz. Changing C1 and 2 brought it up to 85Hz—both figures different from the original intention. But it works well.

Some may look at the power supply (*Fig. 4*) and scoff at its simplicity. However, *it works*. The project box I had used for the ideas of *Figs. 1* and *2* was too small to house everything, so I mounted the transformer on the outside and crammed everything else inside. You can use a slightly larger box if you start from scratch.

So, how does it work and sound? Using pink noise and a microphone, I tweaked the various levels. For my application, I needed to pad the bass amp with R9. Also, using *Stereophile*'s test CD, I found the overall system response as flat as the room will accommodate.

No stereo separation seems to have been lost. The subwoofer is off to the left

TABLE 2

QUASI-ELECTRONIC CROSSOVER

Resistors	
R1-3	470k
R4	1.8k
R5, 6	47k
R7	50k
R8	100Ω
R9	10k
Capacitors	
C1, 2	0.003
C3	1µF
Miscellaneous	
IC1, 2	741, 318, or any quality op amp

TABLE 3

TIME CONSTANTS

f (Hz)	T (µS)
60	2,700
70	2,300
80	2,000
90	1,800
100	1,600
125	1,300
150	1,100

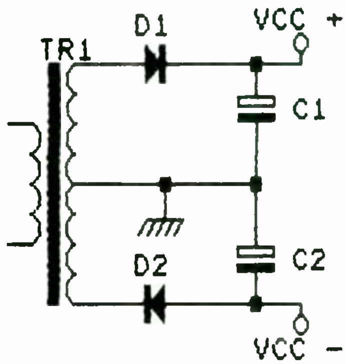


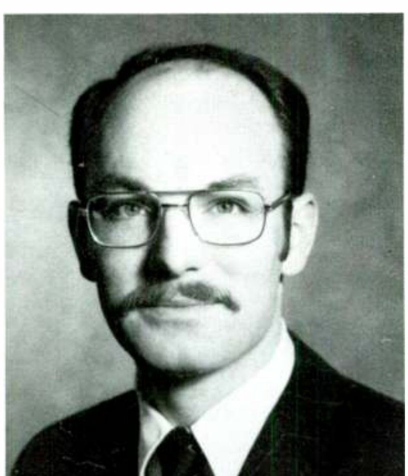
FIGURE 4: Power supply.

TABLE 4

POWER SUPPLY PARTS LIST

C1, 2	2,200μF at 16V or greater
D1, 2	1A, 50 PIV or greater
T1	12 or 18 VCT

of the left satellite in a corner. On a recording of ragtime, the bass drum comes firmly out of the right channel where it seems to be in the recording. On the Sheffield disc "Kodo: Heartbeat Drummer of Japan" the *O Daiko* makes the house *move*. The highs that come from the satellites are about as clean as a set of Magneplanars.



ABOUT THE AUTHOR

Paul Becker has a BA in music. He has professional performing experience with the Vancouver Chamber Choir and the Tudor Singers of Montreal and he has recorded concerts live for three years. His work has included installation of PA systems and five years in the electronics assembly industry. A hi-fi hobbyist for 20 years, the author is a Sign and Banner store co-proprietor with his wife.

THE SPEAKER DESIGNER AS ENTREPRENEUR

AN INTERVIEW WITH KEN KANTOR OF NHT

BY BRUCE EDGAR
Contributing Editor

Ken Kantor (*Photo 1*) and I have always had good dialogs ever since we first met on a visit to AR in 1985. Since that time, I have followed his career in the speaker industry after he left AR and started a consulting business. Those initial dialogs were recorded in Ken's interview that appeared in *SB* 4/86 (p. 20). In the intervening years, I have looked over Ken's shoulder as he and his partner started their own small speaker business, NHT, and began working to establish the company and its unique products.

A company's birth and its struggle to survive is probably the most interesting part of its history, and I hope *SB* readers will gain insight into the process from this interview. By the way, "entrepreneur" means one who assumes the risk and the management of a company. This interview with Ken Kantor took place in April 1990 during my visit to NHT in Benicia, CA with a follow-up interview in January 1991.

Speaker Builder: *The last time we talked, you were working as an independent consultant. But now you are back in the speaker business.*

Ken Kantor: I'm afraid I broke the cardinal rule of consulting and fell in love with a client. I was consulting for a number of companies and was approached by Chris Byrne, who is now my partner. Chris was formerly VP marketing at Pioneer and ran marketing/sales at Akai. He eventually decided to go off on his own and form a speaker company. Chris had the marketing person's view of the world, which is "Gee, I know how to sell the products. It must be easy to make them." I had the engineer's view, "Gee, I know how to make them. It must be easy to sell them." After we worked together, Chris and I decided we were a good match to start a company.

PHOTO 1: Ken Kantor, co-founder of NHT.

SB: *But starting a company is more than saying you have good product ideas and a marketing strategy.*

KK: Well, we had the advantage of coming from big companies and knowing what is required to run a business. That experience can also be somewhat of a disadvantage because many functions in a big company are invisible to you. But we didn't have any illusions. I had worked in a number of startup companies and knew about putting together business plans, and so forth. We had some clear ideas of how we were going to fit in the market, what our products would offer the consumer, and the resources we would need.

SB: *So this effort was not just an off-the-cuff idea of "Let's go start a loudspeaker company."*

KK: Oh no, it was carefully planned. We wrote many business plans, maybe eight in all, and each of those ran for 50 pages.

SB: *What did the business plans cover?*

KK: Everything from the design of the products to the product lines, dealer bases, and pro forma financial outlines. In other words, this is what the company will be at each stage. If things don't go well, this plan is how we will stay alive as a company.

SB: *Were you keeping these plans private or were you showing them to potential investors?*

KK: We showed them to people in the industry who could pass judgement on them. We showed them to people and friends who might contribute money.

SB: *So what were your initial ideas for selling your speakers?*

KK: I must be honest; the last thing we need is another "breakthrough," when numerous other companies spend millions to convince the market they have achieved "breakthroughs." You can't come out and say you have a better speaker because 400 other companies say the same line. And they can usually say it louder than you can. So that eliminates one way of pitching your product. So we thought a great deal about marketing strategy. Initially, our company would be more than a speaker company; it would be a sound company, bringing good sound to realms that needed it, such as TV. Our original idea was to look

Continued on page 24

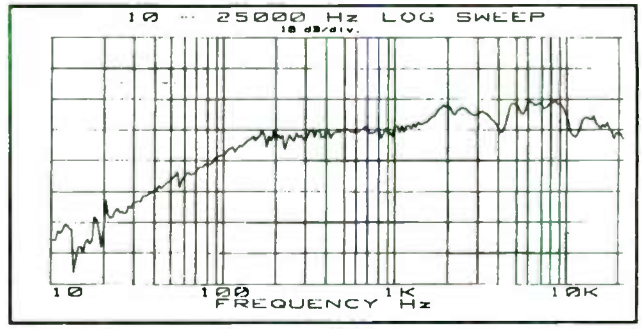
ABOUT THE INTERVIEWER

Bruce Edgar is a project engineer/scientist for a Los Angeles aerospace company. His interests include the history of loudspeakers, horn design, and woodworking.

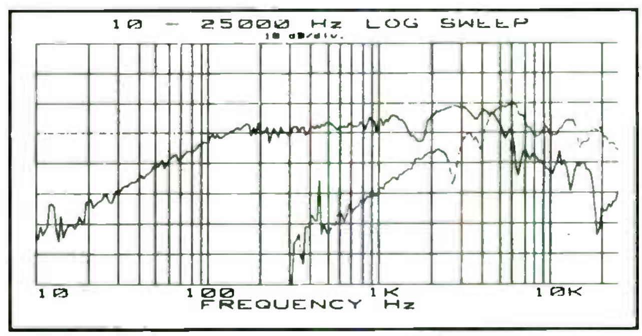
EUROPEAN LOUDSPEAKERS OF AMERICA COAXIAL DRIVERS

Specifications

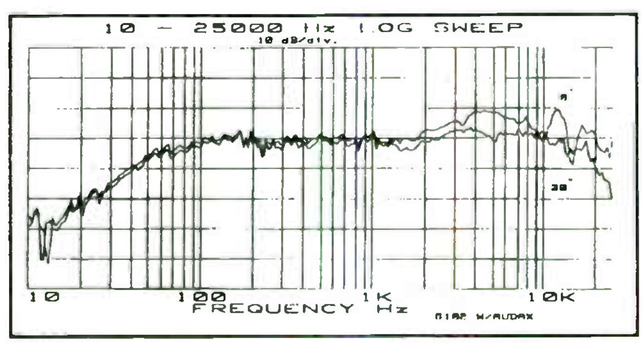
4502/Audax	
Fs	102.0 Hz
Vas	3.8 Liters
Rscc	3.7 Ω
VcL	.18 Mh
Qms	7.77
Qes	.46
Qts	.43
Efficiency	90.5 db 1w/1m
Power	40 Watts
Depth	2¹⁄₁₆"
Cut-out	4"
Price	**$34**



5402/Audax	
Fs	91.0 Hz
Vas	5.42 Liters
Rscc	3.68 Ω
VcL	.22 Mh
Qms	7.84
Qes	.60
Qts	.56
Efficiency	91.8 db 1w/1m
Power	45 Watts
Depth	2.¼"
Cut-out	4⅞"
Price	**$35**



6102/Audax	
Fs	59.0 Hz
Vas	13.1 Liters
Rscc	3.6 Ω
VcL	.25 Mh
Qms	5.6
Qes	.46
Qts	.50
Efficiency	92.5 db 1w/1m
Power	40 Watts
Depth	2¹³⁄₁₆"
Cut-out	5⅞"
Price	**$36**



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Fast Reply #IF112

Continued from page 22
at all products where sound was important. And speakers would be a unifying component.

Our first hard lesson was that nobody will invest in a small US speaker company these days. It just doesn't happen. The investment community is not interested. Now, if you were in some new biotechnology area or something else that is currently hot, the story would be different. They are not interested because they do not see the upside of audio.

SB: What is the upside?

KK: That is a very long conversation, and it's something we are beginning to realize after many years of hard work. But the point is we never had the money to do all our grand schemes. We had to bootstrap our way with products that made us profitable a little at a time.

SB: So what were you doing for the audio market that hadn't been done before?

KK: To be honest, our approach has been done before, but it's not being done now. At one time the "audio" companies were being run by people who loved audio, but many of those companies grew up and changed. I think we always have room for people who love the business and have the intestinal fortitude to say, "I see areas where products can be improved so the customer can get better sound for their dollar."

SB: What part of the audio market are you trying to address?

KK: We are certainly not a high-end company, nor are we a low-end one either. I like to think we bring some high-end sensibility to people who had never thought about buying a high-end product. We care about the sound of our products, but we also care about their affordability.

SB: When did you actually start your company?

KK: The company was incorporated in December 1986, although it seems like a blur to me now. We actually started our development work six months before that. For the 1987 Consumer Electronics Show, we had a little room in a non-CES hotel in Chicago. It was a sad and disappointing experience. Only our best friends who could manage to find us at the end of the day would walk in, look around at our little handmade speakers, and say, "Good luck. When you need a job, give us a call." By the end of the CES, we were really depressed, but we hung in there. We went door-to-door with our speaker and received some press about it with product announcements. Those announcements turned into tentative orders, and tentative orders turned into sell-through. By the next CES, we had sold a few hundred speakers, had orders on the books, and were able to rent an official CES room.

PHOTO 2: A pair of Model 1 loudspeakers, shown with the SW-1 subwoofer in the center.

SB: What was your first speaker?

KK: Our first product was called, creatively enough, the "Model 1" (*Photo 2*). It is a two-way system with a 6" woofer in a funny shaped box.

SB: What does the odd shape of the box do for you?

KK: Speaker boxes are made with square corners for one reason only—it's easy to make them that way. But if you take a fresh look at the sonic and engineering issues involved, a rectilinear shape is not the best for a speaker box. The shape creates standing waves, and in the


PHOTO 3: Sheets of high gloss polymer laminate ready for mounting in the cabinet shop.

typical listening setup, the speaker is pointed away from the listener. The angled front makes the box more rigid. It also breaks up the usual standing wave patterns and avoids the direct reflection back to the woofer cone, with the path length five times longer. You get a cleaner midrange with this shape box. The angled front also directs the primary radiation to the listener and away from the walls and makes a measurable difference in the ratio of direct-to-reflected sound.

SB: So the funny shape does contribute to the sound of the speaker.

KK: It had better make a difference because it's a pain to build, and we wouldn't do it otherwise!

SB: What is the increase in cost over a regular box?

KK: Our costs for a cabinet are two to three times that of our competition. That's a big difference.

SB: How did you develop the drivers for the Model 1?

KK: When I started the company, I thought long and hard about how to improve a loudspeaker's performance without increasing the cost. One idea I had was that I could eliminate most parts of the crossover if I designed into the 6″ woofer the desired rolloff characteristics. I went through almost 30 versions of the woofer before I was able to engineer in the right rolloff and eliminate the little resonant peaks here and there. You gain cost and performance benefits, and no speaker designer on earth says he likes crossovers.

SB: And you have eliminated all that wire in the inductors that the signal must travel through before it gets to the woofer.

KK: That's right. The only parts of the crossover left are a capacitor and series resistor on the tweeter to keep the low frequencies from burning it out. We tried other crossover designs that gave identical on-axis responses, but they didn't sound as good.

SB: Was the Model 1 an immediate success?

KK: There is no such thing as an immediate success if you are a small business. Either you are out of stock of your product or you are out of sales. There is no happy medium. No, it was not an immediate success. Dealers are hesitant to bring in a new product because they don't know whether you will last the year. We had to find dealers who needed a new line and who would trust us.

SB: How did you solve the investment problem?

PHOTO 4: Precut MDF boards with bonded laminate surfaces awaiting assembly.

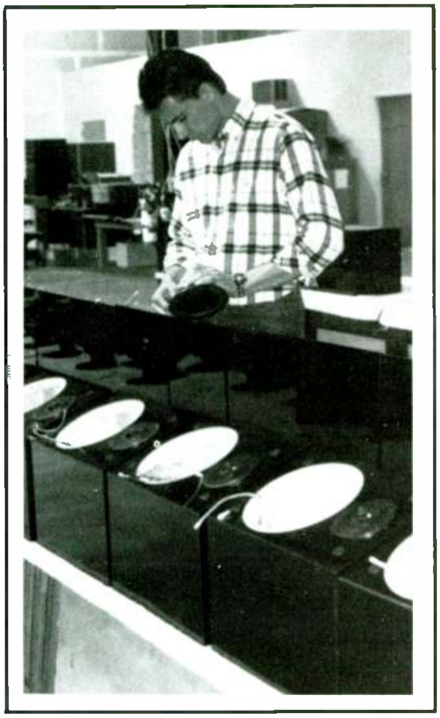

PHOTO 5: Installation of drivers in Model 1 boxes at the NHT plant.

KK: We put up our own money. We signed on the dotted line for many things. After a year, we were able to receive enough orders we could show potential investors. At the point where we had been turned down by everybody and were ready to throw in the towel, an investor read our business plan, liked it, and put in some money. It wasn't enough, but it was sufficient to help the company go forward.

SB: In the beginning, what was your production setup?

KK: At the start, we were a two-city company with Chris in Los Angeles and me in San Francisco. We were buying our boxes from a vendor who builds for many speaker companies. After they built the boxes, we had our own technician in a separate room at the factory do quality control and testing on them. That arrangement worked well when we were shipping 50 pieces a week. But we wished to try new finishes and to go to higher production rates, so we started to look around for other production sources. In a serendipitous situation, we found a guy running a custom kitchen cabinet business. He was great with wood, but he had problems running a business. He had really good innovative ideas about building speakers because he had never built speaker boxes before.

SB: What were the new ideas?

KK: We found ways to use laminate materials that had not been used in mass production before. Nobody had been able to use laminates with our type of edge on a reproducible basis. We built some specialized process equipment and set up a factory that used these new ideas (*Photos 3, 4,* and *5*).

SB: Did the marriage work?

KK: It had its strained moments. The cabinet guy had to learn about production. For example, he made a production run of 200 boxes before we checked whether they really fit properly. We still have a pile of scrap from those early months. But now it's running smoothly as a partnership between the two companies. The box factory is at the stage where it is doing noncompetitive OEM work for other speaker companies, including names you would know. We are both located in the same industrial park in Benicia, CA.

SB: Previously, you talked about going through many samples of 6" woofers in the development of the Model 1. How was that process accomplished?

KK: I was fortunate to have good relationships with many OEM driver manufacturers from my consultant work and previous company affiliations. So they were willing to develop the drivers we needed and, more importantly, to deliver thousands of that final driver design that met my tolerance specifications. That situation is different from what the amateur speaker builder faces. Over the course of many years as a consultant, I've measured many samples of drivers from companies all over the world. And I've found it is dangerous to believe published specs. Furthermore, drivers supplied to the hobbyist market are not faced with issues of consistency. The company knows, as long as the production lot is the same, the amateur who buys a pair

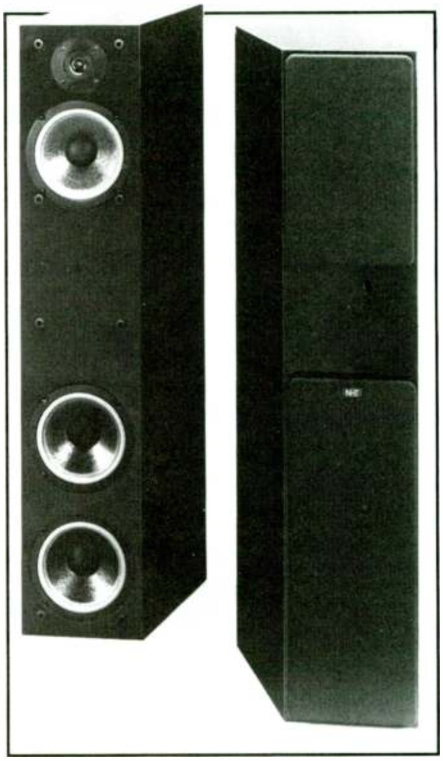

PHOTO 6: The Model 2 couples an upper two-way from the Model 1 with a lower subwoofer module.

of speakers from that lot and fusses with the crossover until it sounds right will be happy until his next upgrade. But the hobbyist never faces the problems a professional company does: the drivers made in January must sound the same as the ones made in December. I can't afford the luxury of tweaking every pad, crossover, and other component to compensate for driver variances.

SB: Where can someone find a NHT speaker?

KK: We have a mix of outlets, including

Continued on page 29

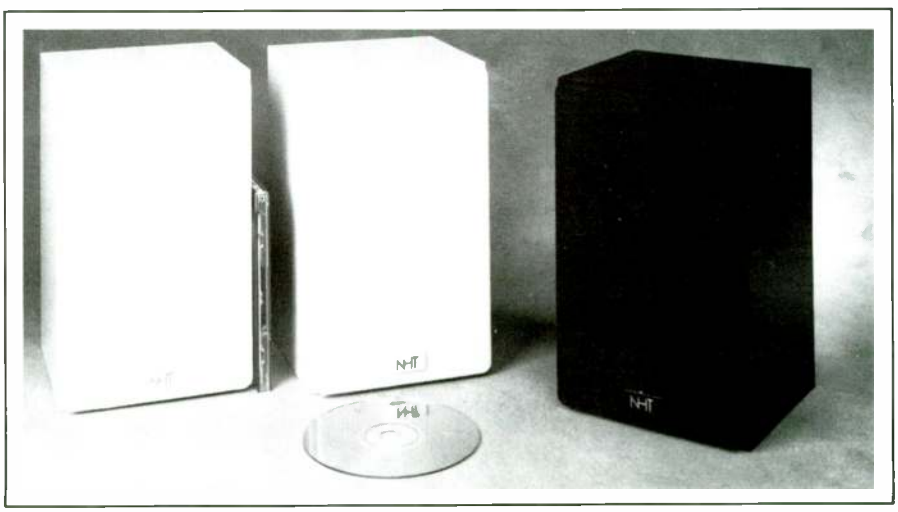

PHOTO 7: The Model Zero as compared to the size of a CD.

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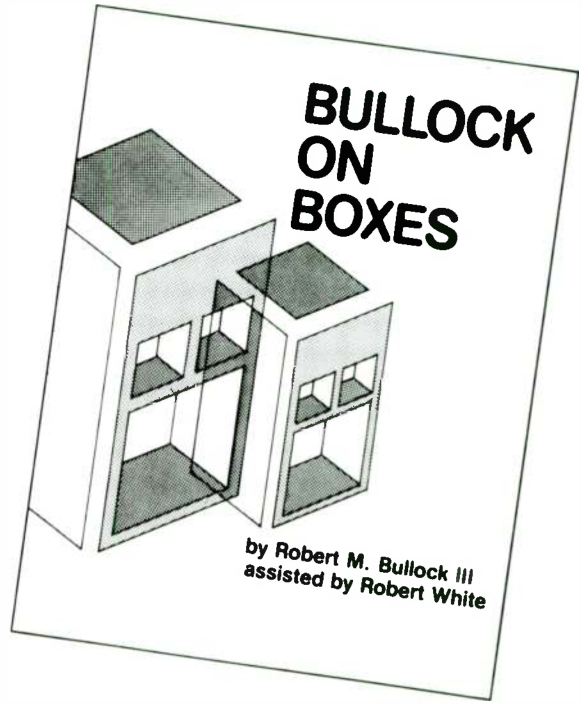

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Continued from page 26
upscale department stores, audio specialty stores, and smaller electronics chains.

SB: How did the name NHT evolve?

KK: NHT stands for Now Hear This. I believe it came from Chris' mother-in-law who was over to dinner at his house one night and saw a giant pair of speakers he was working on. She made the sarcastic remark, "What are they called? Now hear this!" Chris liked the name and thought it might be weird enough to stick in people's minds. We now use NHT, but people still have a laugh when they ask what the initials stand for.

SB: After the Model 1, what was the next product?

KK: We seemed to have used up all of our creative resources after coming up with the company name; so after the Model 1, we came out with the Model 2 (*Photo 6*). You know that two-ways have the advantages in sound stage and imaging, but they don't have the dynamics in the bass. I could have added a midrange and made the woofer bigger, but unfortunately you lose the benefits of a two-way. You have crossover and driver interference problems. So I kept the attributes of the two-way in the Model 1 and added a subwoofer. The crossover point is at 80Hz, which means you have very few wavelength related problems between drivers and no new crossover points in the midrange. We developed a custom 6½" driver with a very low resonance and high cone excursion properties. We put a pair of these woofers in a tower configuration that keeps the same angled baffle shape.

SB: Do you make the subwoofer as a separate speaker?

KK: Yes, we do. It's our SW-1 (*Photo 2*). Originally, we sold it as a subwoofer with the Model 1, but now it is sold mainly with our Model Zero (*Photo 7*).

SB: So you had to back off your numbering scheme for the Model Zero.

KK: We had a Model 1 and a Model 2 and wished to make a smaller loudspeaker, so we came out with the Model Zero, which is small enough that angling the front is no longer an issue. Besides the woofer magnet won't fit an angled cabinet. The Model Zero has applications in a credible stand-alone second system, a surround-sound rear channel setup, or in a college dorm situation.

SB: What about add-ons to a TV?

KK: Absolutely, it is video shielded as are most of our other products.

SB: I noticed you have another subwoofer model.

KK: The SW-2 is a vented subwoofer that has a real bottom end with a cutoff at 22Hz. The driver is a 10" model from Europe with an X_{MAX} of almost 10mm, so it can move some air.

SB: This system is a departure for you.

KK: I've done vented systems before but never for a commercial application. In my opinion, it offers the advantage over a sealed system of a smaller volume for a given cutoff. The big disadvantage of a vented system is the decontrol below resonance. But at cutoffs below 25Hz, it's irrelevant because you are in an inaudible region. When I'm working with 8" woofers or smaller, I don't want a bass reflex system, because inevitably information below the cutoff will modulate the cone. And those effects will be audible.

SB: The enclosure seems to be incredibly small. How big is it?

KK: It's 50 liters in a 16" cube. Normally you would have problems with a cube enclosure, but the box is so small compared to those wavelengths below 80Hz that size is meaningless.

SB: Is the driver a dual or single voice coil?

KK: I was tempted to make a dual voice coil subwoofer, but I observed some odd effects depending on how each voice coil was being driven. So I remained with a single voice coil. It does have a 12dB crossover built in for use with the Model 1 or other speakers, or you can go in directly to the woofer. If you wish to make it into an L+R subwoofer, you must use a separate amplifier.

SB: What are the typical sensitivities for your speakers?

KK: Our published specs are quite accurate and conservative and range from a low of 86dB for the Model Zero to 89dB for the SW-2 and our three-way.

SB: What is your three-way?

KK: That's our Model 100 (*Photo 8*), so called because it is a departure from our other products and was started almost by accident. The little Model 1 has found a home in nearly a hundred recording studios as a near-field monitor. It caught on by word of mouth and its popularity started us talking to record producers. Many studio monitors can play at 120dB day and night until nobody can hear anymore, but they don't really sound good. The near-field monitors sound better but can't play loud. The motivation for the Model 100 is the need for a speaker that can play loud and sound good.

SB: What does the Model 100 use for drivers?

KK: The woofer is the same 10″ as in the SW-2 subwoofer. The midrange was a separate development, and the tweeter is shared by the Model 1.3.

SB: Is the Model 1.3 an upgrade of the Model 1?

KK: Upgrade is the wrong term because the Model 1.3 does not replace the Model 1. The Model 1.3 has better quality drivers and 50% more volume than the Model 1 and still remains a two-way.

SB: The other product I see that is somewhat of a departure is your amplifier.

KK: It has been out since 1989 and was designed to fill a need to which nobody was paying attention. Our model VA-1 (*Photo 9*) is a power booster for home applications similar to power boosters in cars. It has high level balanced inputs so you can hook up a stereo TV's speaker outputs to it. It will take 5–10W from a TV and give 20W/channel into a pair of good quality 8Ω speakers. That's why I call it a power booster. Alternatively, you can also hook up a portable CD player to the line inputs and use it for a second system in the bedroom, dorm, or office or for a rear channel amplifier. It's a no-frills design transparent to most hookups.

SB: What test instrumentation are you using now?

KK: All kinds of testing goes on in a speaker company—testing during building of the product and at the end of the production line to make sure the speaker works before it goes in a box and quality control testing to check the drivers supplied by a vendor. Those are all production-oriented tests. A separate kind of testing goes on in research and development of new drivers and systems. All of these areas have differing instrumentation needs. On the production line, I can spend only one minute testing each speaker for response, phase, and distortion before it goes in a box. In the lab, I need a type of instrumentation that can resolve down to 0.1dB and 1μS.

SB: What instrumentation do you use in the lab to develop speakers?

KK: My own belief is that the first objective to get right is the on-axis frequency response. It must be right at the listening position in a room, not in an anechoic chamber. The goal is to deliver a flat first arrival spectrum response. The only way to measure that response is with an FFT, a MLSSA, or a TEF which can separate what happens to that first arrival signal over time. I use an HP FFT analyzer for my tests. The second job of a speaker is to excite the room with a proper balance of sound energy, the so-called steady-state response or power balance of a speaker. People have fought for the last 50 years over the time response versus the steady-state response. You must get both right to have a good sounding speaker (*Photo 10*).

Continued on page 32

PHOTO 8: The Model 100, a vented studio monitor, features an angled back panel for proper listening orientation. The top mounted midrange has a crossover point at 150Hz and its own sealed enclosure.

Continued from page 30

SB: What do you use for steady-state tests?

KK: I use the Audio Precision system to produce a noise band sweep with a ⅓-octave bandwidth. That test signal tends to smooth out all the little wiggles you find.

SB: Many people say that the ⅓-octave test duplicates how the ear hears.

KK: I don't agree. The ear is more complicated in how it interprets sound. When you listen to an orchestra and say it sounds too bright or dull, those observations are dominated by the steady state. But the ear also has the ability to determine the spectrum of a transient, which is approximated by the first arrival response. For example, the timbre of a drum or of a piano is much more affected by the behavior of the first arrival sound than by the steady-state room response. On the other hand, a group of strings playing at a steady rate will be more affected by changing the relative driver outputs than by changing the properties that could affect the first arrival. The room response is irrelevant to localization. The first arrival is fairly irrelevant to the tonal balance. But again my job as a speaker designer is to get the steady state and first arrival responses right.

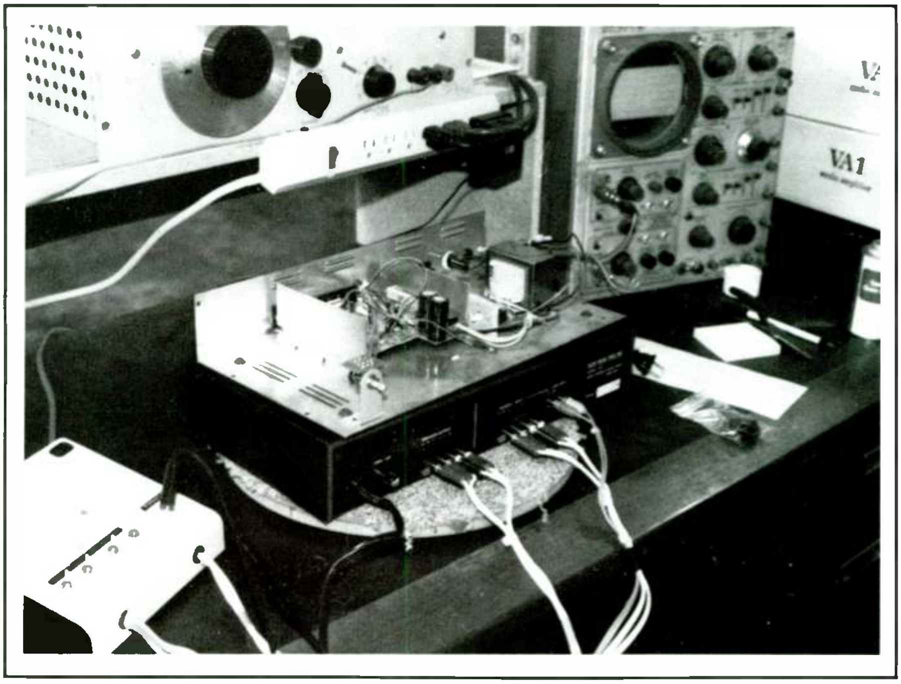


PHOTO 9: The VA-1 utility amp undergoing checkout.

SB: Do you test for the Thiele/Small parameters of your drivers?

KK: Yes, I do. The essence of T/S parameter testing is derived from impedance curve measurements. My initial efforts were like everyone else using a generator, resistor, and so on. But that process is tedious work when you receive a shipment of 500 drivers and must test 5% of them before a production run can begin. So I began to look for ways that the Audio Precision could be used for automated testing of drivers.

SB: What did you find?

KK: The old generator/resistor method was not accurate enough because it wasn't a true constant current source. The excitation level was low enough that I picked up noise and vibration from other sources. And the third problem is that the resonance of the drivers was always shifting. I needed to develop a precision current source that could be driven from the voltage output of the Audio Precision.

SB: What is a precision current source?

KK: It's an amplifier with three op amps, two transistors, a power supply, and 0.1% resistors. It yields more accurate and repeatable data. I can measure the terminal voltage at the speaker, which is directly proportional to impedance including phase and amplitude. And in a high resolution mode, the Audio Precision and the PC/AT will make a complete measurement of the Thiele-Small parameters in less than a minute.

PHOTO 10: Ken Kantor's driver test setup, including a PC/AT control computer, an Audio Precision system, and an HP FFT analyzer.

SB: How do you see NHT's future?

KK: The company has gone through two phases of what is a three-phase process. Phase one is getting a product out the door. It's one thing to have ideas, but it's another to sell a product. Talk is cheap, and a dealer is always going to tell you he likes your ideas. The first time you prove something is when a dealer writes you a check. Phase one is proving you can deliver a product and people will buy it.

Phase two is to make yourself profitable. You must build a company and hire a sales manager, accountant, production supervisor, and other personnel. You need to let people know you exist through product reviews, publicity, and advertising. That's the phase NHT has been in for the last few years.

Phase three is when you establish yourself on the landscape. You become a brand name. Because NHT has done some things right, people are tempted to say we are an established brand. I wouldn't say that. At some point the company must outlive its founders and have a life of its own. At present Chris and I are the heart and soul of NHT, which would have a hard time continuing if either of us were run over by a bus. So we are not at phase three yet.

SB: What do you say to someone who wishes to start his or her own speaker company?

KK: I think it's a personal decision if someone wishes to turn pro. Most people who start speaker companies are not finance guys or investors. The only people who start speaker companies are the ones who love speakers. I think different people have different goals. Some wish to make the best speaker possible. Others have the goal of selling the most boxes. Some people start speaker companies by accident. They develop speakers to please themselves according to their own ideas and begin to develop a following. My word of advice is that I hope you have plenty of money in your bank account to live on for 12–18 months because you won't have any money coming in so fast.

The first rule of a business is to make real revenue out of a product. I mean making a profit and not selling at a loss. Profit is directly related to pricing of a product, and most amateurs without business backgrounds have no idea how it works. Material cost is one portion, but you have your time, overhead, advertising, and other factors. The bottom line is that if you are making a profit in a year, you are doing amazingly well.

Editor's note: In May 1990, International Jensen bought NHT. At the 1991 CES in Las Vegas, I talked to Ken again to update this interview.

SB: How did International Jensen approach you about buying NHT?

KK: A Jensen VP who was a friend of Chris saw some of our products. The Jensen management decided to visit us at the 1990 Las Vegas show. We had a potential fit with Jensen, and we all seemed to get along with each other. After five months of negotiation, Jensen bought NHT.

SB: Has NHT changed because of International Jensen's buyout?

KK: Many subtle changes have occurred, but if you walked in the door today, you would probably not see the differences. I now have an engineer working for me, and Chris has a sales manager. We have a new computer tied into Chicago to help us with finance and order processing. But operationally, it's been a hands-off affair. Jensen has not told me what

PHOTO 11: The VT-1 video tower loudspeakers.

I should be making, how the speakers should sound, or what we should sell our products for. The attitude has been that we submit our goals and requirements, and as long as we are reasonably on track with them, Jensen has nothing to say. It's been a very nice relationship.

SB: How big is International Jensen now?

KK: Jensen is bigger than most people think. In addition to the well-known brands they own like AR, Advent, Phase Linear, and Jensen, they are a large supplier to Detroit. They own the largest driver plant in North America, located in North Carolina. They have large magnet and cone plants. International Jensen is among the top three players in the American loudspeaker market.

SB: What does Jensen do for NHT?

KK: They give us the ability to make long-term decisions and plans. They provide the capital investment necessary to upgrade production facilities, get the drivers we need, and provide advertising—all the items you need to make a company run well.

SB: Have there been any changes to your product line since we last talked?

KK: We have upgraded the Model 1 after four years on the market. The Model 1A has a subtle change to the crossover that takes a little time to hear, but it's there. The Model 2 has been upgraded to the 2.3.

SB: Do you have any new designs?

KK: The new product is the video tower speaker, the VT-1 (*Photo 11*). It's the first speaker to be optimized for the requirements of video, which means a subtly differing sonic approach. The crossover is set to give a spacious sound rather than a localized sound. In a video setup, you are not so concerned by the exact positioning of the sound sources.

SB: Is the VT-1 optimized for voice?

KK: It's more complex. It's optimized for the mix of voice and music in a video soundtrack. It trades off the reverberant steady-state energy with the direct energy in a different way than what I would choose for an audio speaker. This choice leads to more clarity and removes some of the harshness you hear on video soundtracks with the reverberant soundfields that typical hi-fi speakers give.

SB: The tall configuration is a bit unusual.

KK: Yes, the narrow profile does some tricks with tuned column loading of the bass. Originally, we tried to design the VT-1 with a small footprint and ran into all these interesting organ pipe effects. Over six months, we built 43 versions and tried to use the pipe resonance to our advantage. We vent one end to the room such that it equalizes the system at the low end. The net result is a system with the efficiency of a vented box but without the unloading problems below resonance. The small 4½" woofers go down to 70Hz with a 91dB sensitivity.

SB: Where do you see NHT going in the future?

KK: I don't see much change. We went with Jensen to do the things we always wished to do. One of the pitfalls we wished to avoid in starting NHT was selling down to the customer. If you do that, you develop more and more customers who don't care anymore. This has happened to the mainstream audio business, who said features are important because everything sounds alike. That attitude created a mass of consumers who believed it, and then the audio business was caught in their own trap because they could not sell against competitors anymore.

And NHT believes people can hear the difference. Maybe they don't understand the buzz words, and maybe they must be taught by the dealer. But we believe that ultimately good sound and performance will benefit the consumer.

SB: Are you at phase three yet as a company?

KK: No, we are still ending phase two, that is being taken seriously. We have a lot of work to do yet. But I don't have ambitions to have NHT grow to mammoth proportions, and neither does International Jensen. They already have large companies. But we wish to reach our potential in our niche, bringing high performance to affordable audio systems. I'm gratified people are now taking us seriously as a sonically good brand.

SB: Thank you, Ken, for your insight into your company and the industry.

THE DOALS PARTY SPEAKER

BY JOSEPH R. DEMERS

A party doesn't seem like a party without music. Although some readers may hire professional musicians, many of us depend on a powerful stereo. I was recently at a party in which the host was entertaining his hundred or so guests with loud music. The music emanating from the speakers wasn't unusually loud when I was in the back of the room, but as I moved closer to the stereo, conversation became impossible. Since I couldn't talk to anyone and since it wasn't an entertaining party, I tried to figure out ways to spread the speakers' output, but retain their tremendous resolution.

I continued my pondering for the next six months or so, and I came up with a number of ways to prevent the one-side-of-the-room stereo effect encountered at countless gatherings. All the solutions depended on separating the speakers and placing them more toward the center of the room. But, unless the output could be spread more uniformly, people would simply move from the middle of the room into the corners.

A simple solution would be a speaker with three tweeters, three midranges, and a couple of woofers all pointing in different directions. Another solution might be ceiling speakers scattered throughout the room. In either case multiple drivers are needed. (A clarification is necessary here. When I write about multiple-driver systems, I mean multiple drivers used in one frequency range, and when I speak of single-driver designs, I mean ones in which a single driver is used for a range of frequencies.)

The most noticeable problem with multiple-driver systems is the reduction of resolution, which occurs when two drivers radiating the same channel in synchronicity are separated by some distance. The separation of the drivers distorts the timing/phasing which enables the listener to pinpoint, from the music, where the instruments are located. For example, with common directional speakers, you can determine that the horns were a little to the left, the violins were to the right of center, while the singer was directly in the center during the recording. If, however, a second tweeter is hooked up to each speaker and is pointed in a slightly different direction, it is impossible to tell where the horns, violins, and singer were.

Considering the problem further, I contemplated spreading the output of single-driver systems with a lens. I imagined a stacked type of assembly with a lens for the midrange and a lens for the tweeter, but none for the woofer because lower frequencies are naturally omnidirectional.

What was finally created was a vertically phase coherent, vertically diffractive, semicircular omnidirecting speaker. Whew! Since that description was on the windy side, I shortened it to DOALS: Demers Omnidirecting Acoustic Lens Speaker. I had to include my name to get such a catchy title. And, as such, a patent was applied for on the unit.[1]

DESIGNING THE LENS. At first, I wondered if it were possible to design a device that would redirect the drivers' output without destroying the acoustic quality or the overall resolution of the speakers.

My first idea was a simple reflective device, a cone mounted under a driver in such a way that when the wavefront hit the nonresonating surface it was reflected 360° in the horizontal plane. Crude, but workable. Even if I could design an acceptable lens, it could only be placed in the center of the room. That means cords for people to trip over, a convenient place to set one's drink, possibly a loss of stereo, and a simply unaesthetic approach. If only the speakers could be placed against a wall, or better yet, hung, unobtrusively, from a wall. Such an arrangement would require that I create a speaker with a 180° output. How could I do this?

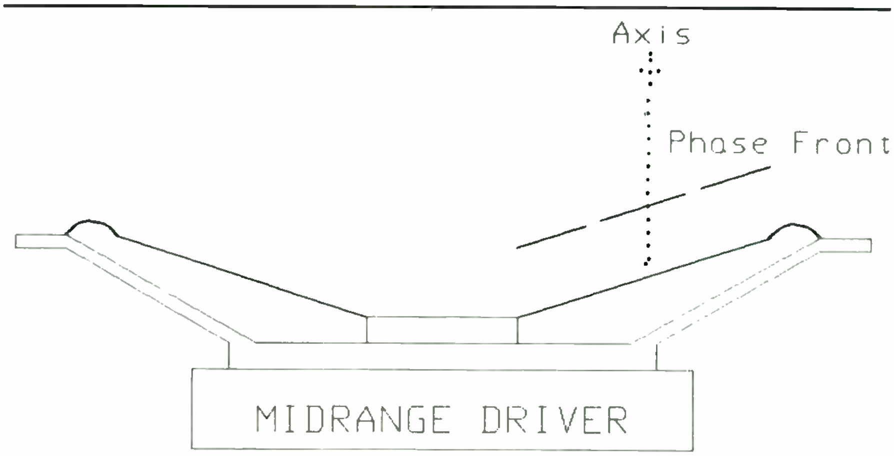


FIGURE 1: The phase front at an angle to and moving down the driver's axis.

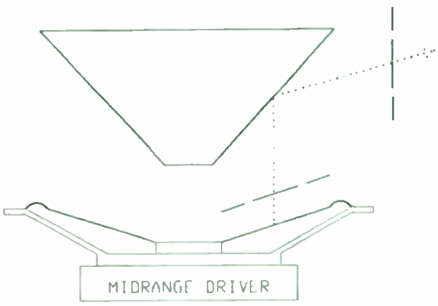


FIGURE 2: A vertically aligned phase front.

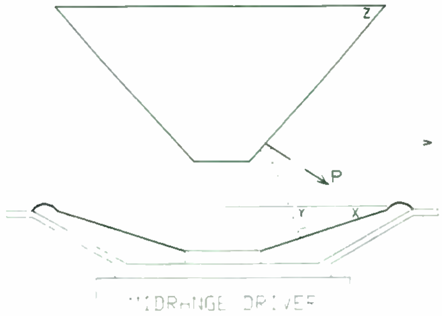


FIGURE 3: The lens angles.

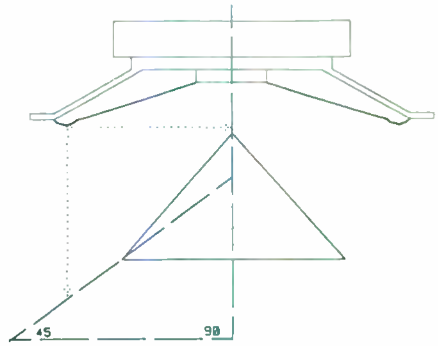


FIGURE 4: The driver radiates like a point source at the outer edge.

My thoughts strayed toward simply damping the undesired output off the cone with an absorbent material. I quickly abandoned this idea because it was wasteful. I tried to imagine a cone that would not only radiate the anterior portion of the output into the room, but would also redirect the posterior portion away from the wall and toward the extremities of the room.

Other factors would also decide the final shape of the lens: it should radiate without any phase cancellation and the surface had to be intricately designed to match the driver. Should the lens be designed around the reflecting wave front or the phase front, and from what could the lens be constructed?

I tried to design a mathematical model for the shape. It would be a simple matter of applying Huygens' Principle to every point on the driver cone, transforming the wave vectors at the lens, and deriving the final shape from this information. Needless to say it turned out to be quite a project. I am still working on it and probably will be for some time.

While I was laying the foundation for a mathematical model, however, an intuitive model started to take shape. Even so, I was still deciding how to meet the two constraints: radiating without phase cancellation and redirecting the posterior portion of the drivers' output. I examined a typical driver more closely for a solution and discovered the wave front and phase front are at an angle to each other.

Why a difference exists between them has to do with how the driver is oscillating. In most cases, the driver is moving in and out, sending the wavefront down the speaker's axis. Since the sound is created at the same instant at the back, inside portion of the driver surface as it is at the outer portion, and since both waves are moving at the same speed, they stay the same distance from one another. Therefore, the phase front is at an angle to the driver's axis, but is moving down that axis (*Fig. 1*).

For calculations, it is easier to use the normals for the fronts. These normals are perpendicular to the fronts and, from the forementioned information, you know that the phase vector is at a constant angle to the wave vector.

Since phasing causes interference patterns, I chose to design the lens around the phase vector. If the phase of the output were controlled and coherent, would this minimize destructive interference? I believed if the phase front were vertically aligned as it left the driver-lens combination, it would help keep interference to a minimum. What "vertically phase coherent" means is that if a vertical line is drawn next to the driver-lens combination, all output on that line was created by the driver at the same moment in time. This can be seen in *Fig. 2*.

CALCULATING SLOPES. I calculated the lens slope from the characteristics of the driver surface. The simplest driver to use is one with a flat cone, mostly found in midrange or bass drivers. Before calculating the slope of the lens, it is necessary to find the slope of the driver surface. I did this by measuring the angle between the surface of the cone and the horizontal. An angle of 30° is common for a midrange driver, so I will use that to illustrate my calculations. If the driver surface is at 30° with the horizon then, from simple trigonometry, the vector for the phase front has an angle of:

$$180° - (30° + 90°) = 60°$$

Since the phase front will strike the lens and reflect at the angle of incidence, the normal to the lens surface must bisect the angle of the phase front (60°), while also reflecting the phase front parallel to the ground.

The lens angle calculations are as follows. From *Fig. 3*, X = 30° and Y = 60°. Since the phase vector reflected from the lens is parallel to the horizontal, the angle between incidence and reflection is also 60°. The bisector of this angle must be perpendicular to the lens surface, such that the calculation for the lens surface is:

$$Z = 90° + Y/2 = 120°$$

These calculations ignore Huygens' Principle to some extent. Since the face of the driver is flat and oscillating uniformly across its surface, the outer edge is the

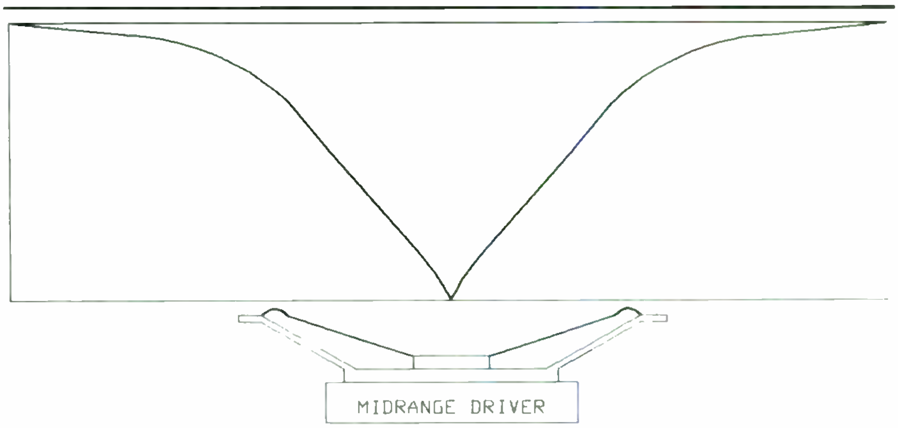


FIGURE 5: Approximating a lens curve from the calculated slope, the 90° slope, and the 45° slope.

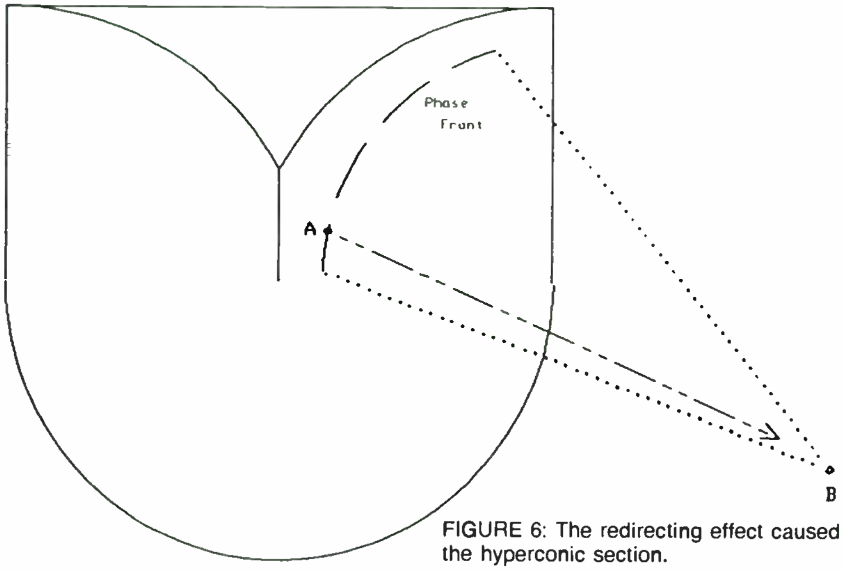


FIGURE 6: The redirecting effect caused by the hyperconic section.

only area where it is necessary to apply the principle.

At the outer edge, the driver is radiating like a point source (*Fig. 4*). It is easy to adapt the lens to two of the wave vectors that originate from this point source: the horizontal vector and the vertical vector. The horizontal wave vector must strike the lens at 90°—that is, straight on. Therefore, I simply approximated the cone shape at the top of the lens and brought the lens to a point with an almost vertical slope. The vertical wave vector, however, must strike the lens at 45°. This means the portion of the lens directly below the driver edge must have a slope that creates an angle of 45° with the driver axis.

At this point in the description, the lens is characterized by three features: the calculated slope, the 90° slope near the top, and the 45° slope near the bottom. From these three features, it is possible to approximate a lens curve that will connect all the points smoothly (*Fig. 5*).

More common, nondome tweeters (like the Polydax TW60s I used), however, do not have a flat radiating surface, but a curved one. For a case like this, it is necessary to derive the lens surface through an integration. It is helpful to imagine an infinite number of flat surfaces arranged around a single point all with slightly different slopes. Each surface is treated as the driver's actual surface and a lens calculation is done with each of the slightly differently angled pieces. Whichever driver and method you use, the calculation produces the shape of a regular cone.

REDIRECTING OUTPUT. As I stated earlier, I didn't want a simple cone, but a lens specifically designed to reflect the drivers' output away from the wall and to spread it uniformly throughout the room. To accomplish this, I slightly altered the rear portion of the lens. Since the phase vectors striking this area have the same angle associated with them as the front portion, the lens slope is unaltered. I built the rear portion, however, so it will deflect the drivers' output into the extremities of the room. This required that I change the horizontal components of the reflected wave vector.

The output from the rear of the driver must strike a lens that not only renders it vertically phase coherent, but also focuses it. For this reason, the back of the lens is hyperconic; it still possesses the same slope, but is the opposite of a cone. In *Fig. 6*, point A denotes the bottom of the wave vector (V), more easily seen in *Fig. 7*. This wave vector moves parallel to the driver's axis and has the phase front associated with it. Because of this, the lens must have the previously calculated slope. If the lens curvature (with respect to the X-Y axis) is altered slightly, however, the wave vector will be directed into the room and away from the wall.

Figure 6 displays this redirecting effect caused by the hyperconic section. B appears to be a focal point, but the focal point actually changes as you move around the driver. If you follow different wave vectors produced by different areas of the driver, you find that the focal point (B) moves. For this reason, the lens doesn't focus the output, but radiates it through 180° in the horizontal.

Huygen's Principle suggests that from a single point on the driver a spherical shell of wave vectors should arise. This gives rise to the following type of interference patterns: a spherical wave is created at point A (*Fig. 8*) on the driver; one wave vector (V1) strikes the lens and travels to point B, another (V2) strikes a different portion of the lens and travels to point B. These wave vectors have traveled different distances. Since they have a phase associated with them, they will interfere with one another. Fortunately, because we are not using a point source but a flat plane, the only wave vector that exists is the one perpendicular to the driver's surface, or V1. All others destructively interfere with the nonperpendicular wave vectors from nearby points. Interference occurs only due to the driver's edge effects.

Actually, the rear portion of the lens is used mostly to redirect the waves created at the edge of the driver. This can be seen in *Fig. 9*, where the dashed line indicates the driver's edge. It seems this hyperconic section need not be so large if only the inside portion redirects the main output of the driver. The extra size, however, is there for the same reason the lens is so much larger than the driver: to redirect as much of the output as possible.

From *Fig. 9* also notice the lens blade.

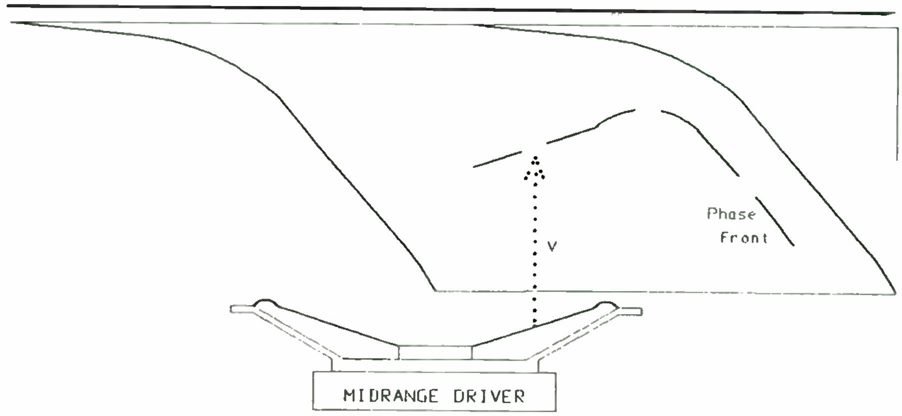


FIGURE 7: The bottom of the wave vector is moving parallel to the driver's axis and has the phase front associated with it.

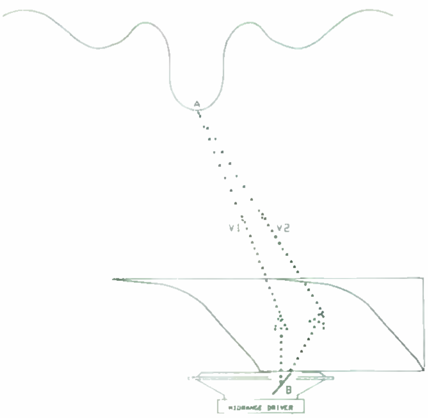


FIGURE 8: A spherical wave is created at point A on the driver.

This is the portion at the top which is necessary to split the driver's output into two equal areas. It was not part of the original design, but the need for it became obvious when I placed the driver over the lens.

Photo 1 shows the heavy, construction paper understructure and *Photo 2* is the final lens. If you view any one of the pieces in *Photo 1* more closely, you will see it is a triangle, except that the hypotenuse is slightly curved, reflecting the shape of the lens slope. This reinforces the concept of a constant slope since the pieces are the same for the hyperconic section as for the conic section.

I filled this understructure with plaster of paris mixed with sawdust to create a dense, nonresonating surface. I later touched up the surface with bondo and glazing putty. After I sanded the unit, I sprayed it with a sandable primer and resanded it until the finish was smooth and even. This created a regular surface that would not color the output (*Photo 2*).

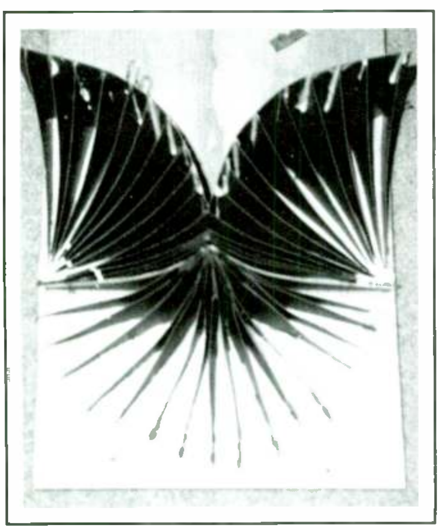


PHOTO 1: The heavy construction paper understructure.

THE SPEAKER SYSTEM. Before I could design a speaker system around the lens, however, I needed a lens for each tweeter. Since the high-frequency lens was considerably smaller, I simply carved it out of wood. The tweeter's lens was also cleaned with bondo, glazing putty, and sandable primer until a smooth, polished surface resulted.

When the lenses were finished, I started to design a speaker enclosure that would effectively use them. I discovered that with the midrange and tweeter mounted over their respective lenses, an aperture is formed between the driver and the lens. The width of this aperture was on the order of the wavelength of the frequencies being produced by the respective drivers. The aperture of the midrange is approximately 9cm corresponding to about 3.8kHz, and for the tweeter, 2cm or 17kHz. Not only is the lens directing the output over 100° in the horizontal, the lens and driver unit is causing diffraction in the vertical plane as well!

Another factor affecting the output of the driver/lens combination is something I discovered when I first designed the lens. Although the phase vector is aligned parallel to the horizontal, the wave vector (the path down which the phase front travels) is at an angle to the horizontal. This means the driver/lens must be located high or turned upside down and placed low so the output will travel to ear level.

With everything previously mentioned in mind, it was not difficult to design a speaker. To make it simple and keep the variables to a minimum, I tried to secure the same drivers and crossover I had used in a previous set of great-sounding speakers. I couldn't find them, but I wasn't concerned about how the drivers sounded. At this point in the project, I simply wished to test the redirecting effect of the lens. All the components I employed are used because of cost and availability. You can substitute different drivers if they meet the constraints mentioned, or you may construct custom lenses.

The DOALS speaker centers around

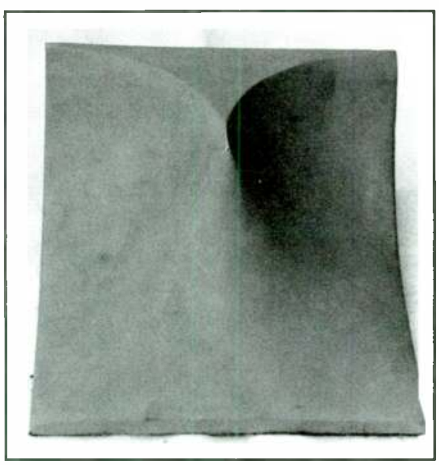


PHOTO 2: The completed midrange lens.

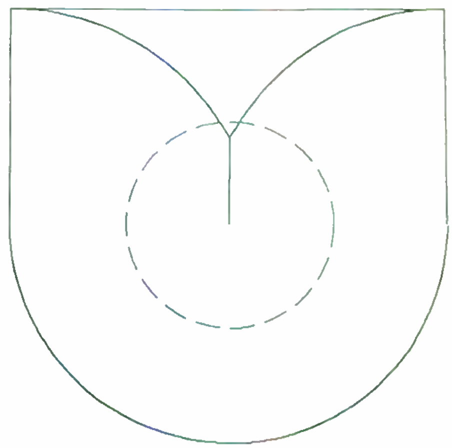


FIGURE 9: The rear portion of the lens redirects the waves created at the driver's edge (the dashed line). The blade portion of the lens (at the top) splits the driver's output into two equal parts.

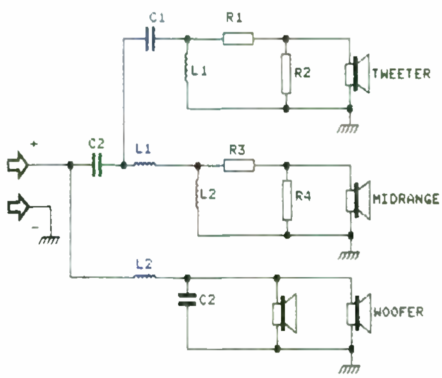


FIGURE 10: The crossover schematic.

TABLE 1

DOALS PARTS LIST

Attenuation resistors (15W)	
R1	0.5Ω tweeter, series
R2	37Ω tweeter, parallel
R3	2Ω midrange, series
R4	12Ω midrange, parallel
Capacitors (nonpolar and 50W+)	
C1	5.6μF tweeter-midrange
C2	33μF midrange-woofer
Inductors (air core and 50W+)	
L1	0.15mH midrange-tweeter
L2	1.25mH woofer-midrange
Drivers	
tweeter	Polydax TW60A, piezo
midrange	Polydax MHD12P25 FSM, paper cone
woofer	6.5″ heavy-duty polypropylene in an Isobarik configuration

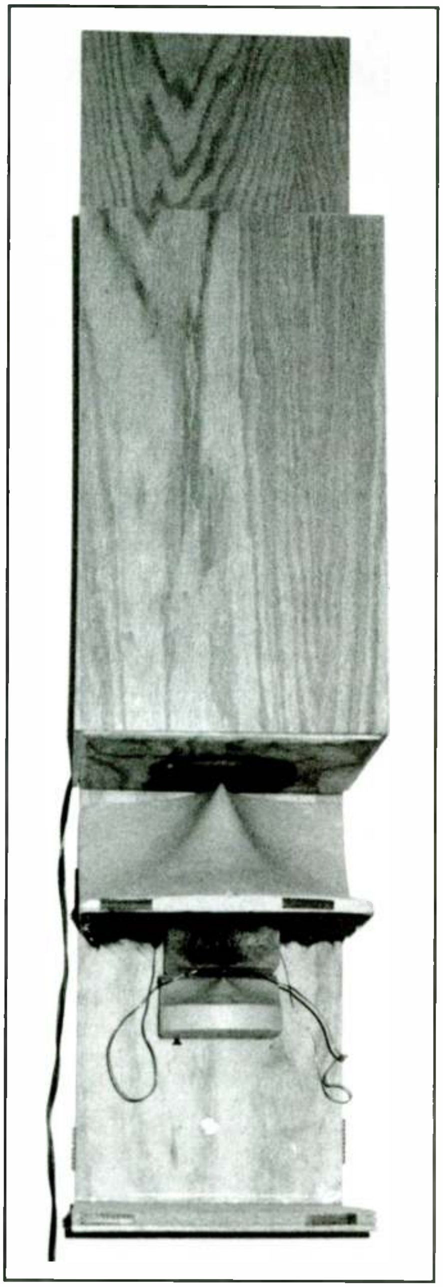


PHOTO 3: The speaker mounted on the wall.

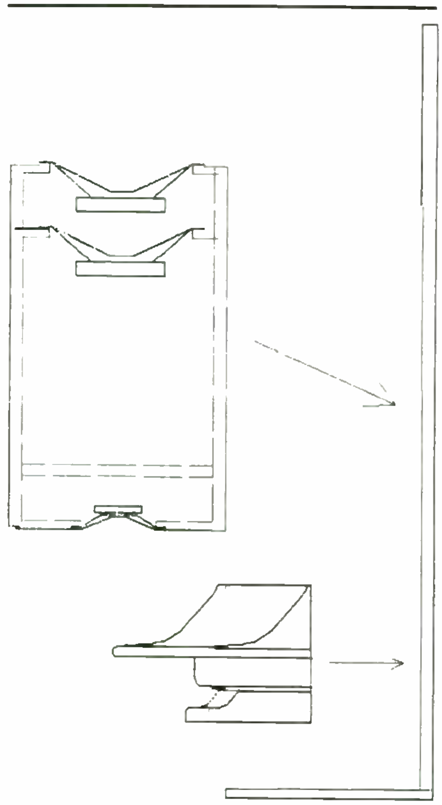


FIGURE 11: The cabinet modules.

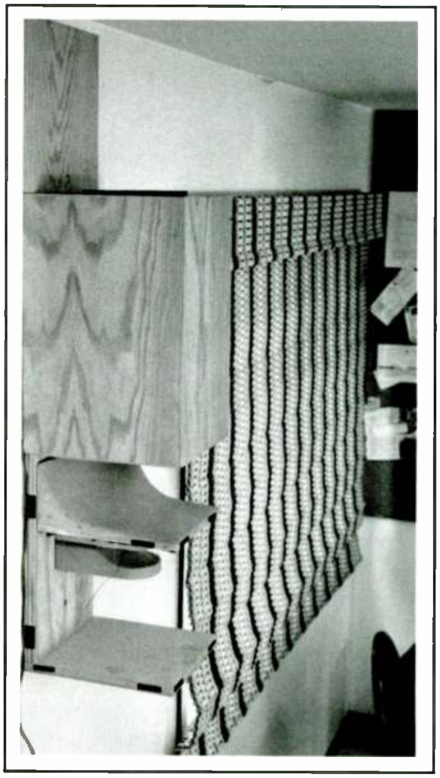


PHOTO 4: A side view of the speaker.

two 6½″ woofers in an Isobarik alignment. The amount of flawless bass that can be produced in this alignment always amazes me. I've also discovered that the Isobarik design is forgiving about the woofer quality. As long as the driver with a Q_{TS} of about 0.70 and a power rating between 30 and 50W is used, you will have no problems. Woofers with these characteristics are fairly common and can be obtained as generic products from any electronics mail-order catalog. Unfortunately, using two 8Ω drivers in parallel means a 4Ω system.

I used a Polydax MHD12P25 FSM rated at 50W and with a 4Ω resistance for the midrange.[2] For experimentation, it's cost-effective, as is the tweeter I employ, a 4Ω Polydax TW60A.

A three-way, 12dB slope crossover, which crosses at 800Hz and 5kHz is used for the passive filtering. (Unfortunately, I have never been able to find the article, or book, from which I obtained it.) *Figure 10*, the schematic, displays the attenuation circuit necessary not only because of different driver efficiencies and the room environment, but also because of the drivers' orientation.

The enclosure uses an upward firing woofer. This resulted in an attenuation of 3.5dB for the midrange driver (R4 = 12Ω, R3 = 2Ω) and 1dB for the tweeter (R2 = 37Ω, R1 = 0.5Ω). A simple attenuation circuit taken from *The Loudspeaker Design Cookbook* (p. 50) took care of this.[3]

I made an axially aligned system with the centers of all the drivers on the same vertical axis and kept the lens right side up. An upright lens necessitates mounting the speaker on the wall near the ceiling, with the woofer facing the ceiling and the tweeter and the midrange driver pointing toward the floor. As you look at the speakers on the wall, the lowest part is the lens for the tweeter, followed by the tweeter mounted in a spacer, which itself is attached to the midrange lens (*Photo 3*). A spacer is necessary to prevent reflections of the tweeter's output off the back of the midrange lens. The midrange is placed over its respective lens and attached to the bottom of the woofer enclosure. The midrange has a compartment separate from the woofer, which it shares with the crossover. The top of the speaker is the 0.52 ft³ enclosure for the Isobarik woofer (*Photo 4*).

THE ENCLOSURE. The cabinet is made from ⅜″ particle board covered with ¼″ oak plywood. I made no attempt to construct a heavy nonresonating enclosure as it makes frequent removal from the wall for adjustments difficult. The cabinet is made of modules as can be seen in *Fig. 11*.

As stated earlier, the two lenses were screwed together with the tweeter and spacer assembly sandwiched in between. This was then fastened with screws to the speaker backbone (¾″ plywood). The woofer and midrange enclosure is also a separate unit (albeit lacking a back), attached to the speaker backbone.

At the top of the speaker (near the woofer), the plywood was covered with the oak plywood while at the bottom (near the tweeters), a square piece of oak-covered particle board was added (*Fig. 12*). This square portion is smaller than the woofer and midrange enclosure by ¼″ so the foam that hides the lens assembly fits snugly against the speaker. The foam is held securely with hook and latch adhesive strips.

This enclosure design has some drawbacks, the main one being accessibility to the crossover. To make adjustments,

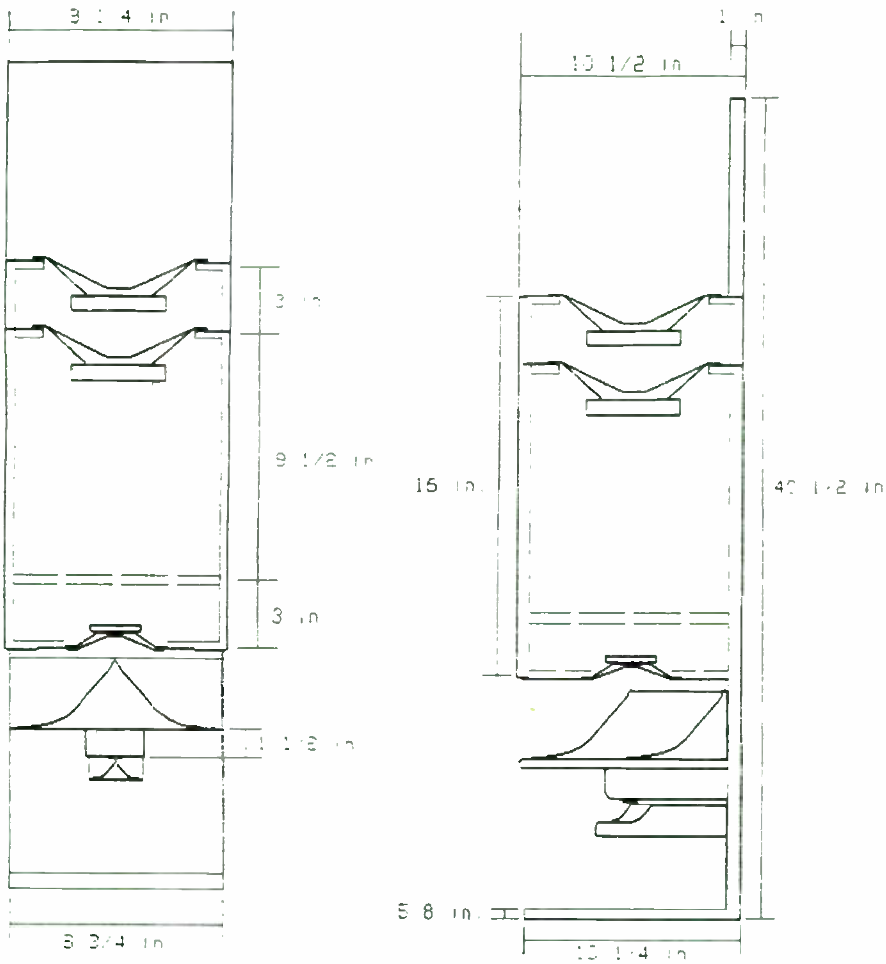


FIGURE 12: The cabinet dimensions.

you must remove the speaker from the wall and disassemble it. As a whole, many improvements can be made on the enclosure, but in my case it served its purpose—to test the lenses.

THE BIG TEST. The DOALS are mounted to radiate into three rooms: a family room, a dinette, and a large kitchen. All rooms are open to one another. In *Photo 5* you can see a portion of the dinette and the family room as well as one of the wall-mounted DOALS. From the wall of the kitchen to the wall of the family room is 38 feet, while the width of all the connected rooms is roughly 13 feet. A perfect environment for the DOALS system.

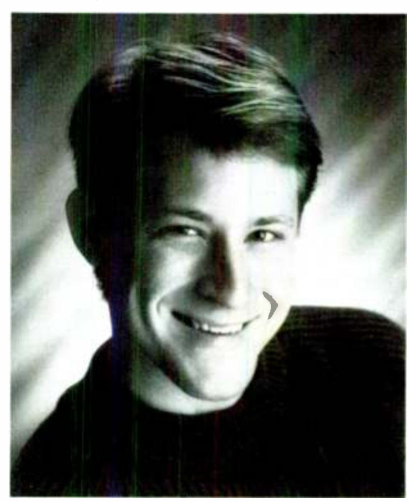
The first characteristic of the speaker I noticed (after powering up) was that they seemed inefficient. As I increased the volume, the output didn't appear as great as the efficiencies of the drivers implied. After a moment, I realized this was because I was hearing only a portion of the 180° output. Did this mean they worked? I switched the balance to only one speaker and walked in a half circle around it. The volume didn't change.

Returning the balance to normal, I started from the wall of the kitchen, walked between the speakers to the far wall—38 feet. The volume and resolution hardly changed at all. I continued to walk around; the stereo was prominent wherever I stood and the output was spread uniformly throughout the three rooms. They worked! The only problem was near the walls on which they were hung. As I neared either, the music became slightly distorted. I believe the distortion indicated interference effects created by reflections from the walls.

Surprisingly, even though availability and cost governed the choice of components, the speakers sound good. Now that I know the lenses work, I can use well-known driver and crossover combinations, maybe even dissect and rebuild a set of commercial speakers. Because the lenses are inexpensive to construct, the next step will be to experiment with different lens coatings and alignments. For instance, the volume increased slightly directly between the speakers. Could this be remedied by removing a vertical slice from the middle of the lens?

Continued on page 86

PHOTO 5: A portion of the listening area for the DOALS.



ABOUT THE AUTHOR

Joseph R. Demers is a fifth year physics and biology student at the University of Dayton. He first became interested in loudspeaker design four years ago when his father described how an Isobarik system worked. Since that time, it has become his primary hobby.

A LOUDSPEAKER CABLE

BY JOHN D. JACKSON

What I'm proposing here is a high-quality loudspeaker cable construction you can modify to suit your applications. The largest diameter wire I could source was 20AWG. As it is very thin, you can bundle cables and put them in a shrink tubing sleeve or lay them out in strips and place them under a carpet.

The cable is constructed with two strands of silver-plated, solid-core, teflon (PTFE) insulated 20AWG wire. You'll also need 1mm Ø thread or nylon fishing line. Since this cable is quite thin, you can make up pairs to decrease resistance and/or increase current-carrying capacity.

How little resistance and how much current-carrying capacity should a cable have?

Regarding current-carrying capacity, I don't know what quantity of current density *sounds* best, especially since music does not usually consist of a constant AC signal. According to a textbook through which I was browsing, 15A appliances require a minimum (for safety reasons) of 14AWG wire, which has a diameter of 0.064″ or 1.6256mm Ø, representing an area of 2.075mm². Going by that, each 1mm² using copper can carry 7.23A. To use the suggested example, 20AWG has an area of 0.52mm², therefore a current capacity of 3.76A. The 20AWG as used above has a resistance of approximately 0.032Ω/m. Considering that some tube power amps have an output impedance as high as 2Ω, you wonder what's the point in spending lots of money on super conductors.

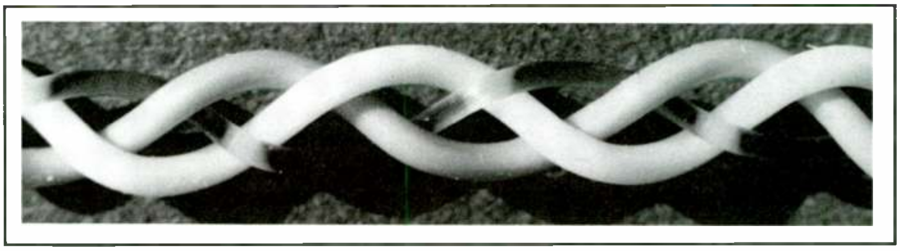


PHOTO 1: Plaited cables (photo by Ulrich Otte).

According to Martin Colloms,[1] a cable with a resistance of 0.4Ω would represent an audible 1dB drop with a 4Ω speaker (0.5dB with 8Ω). This should be okay if you can keep the cable resistance down to 0.2Ω. James Moir[2] suggested the same. The beauty of the proposed scheme is that you can add cable pairs until no audible improvement is wrought. In most cases, the limiting factor will be the loudspeaker cable resistance as opposed to current-carrying capacity. From the above, it stands to reason why bi-wiring evokes an audible improvement.

Even if commercial cables have less resistance per unit length, their inductance is invariably higher. For example, Naim NAC 5 R = 0.048/m; L = 4.9μH/m (5μH represents 0.63Ω at 20kHz). My cable construction has little inductance, virtually no self-inductance, and is phase-coherent up to a frequency of 26.6kHz.

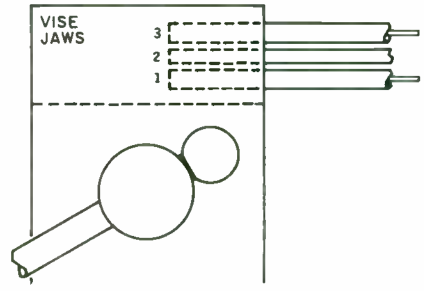


FIGURE 1: Ends of the cables placed in a vise jaw.

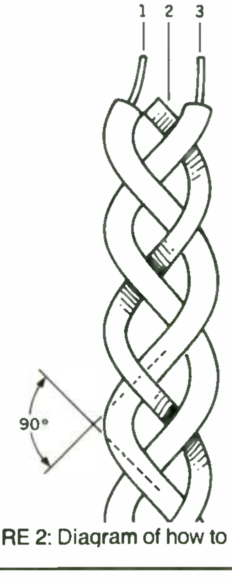


FIGURE 2: Diagram of how to plait the cables.

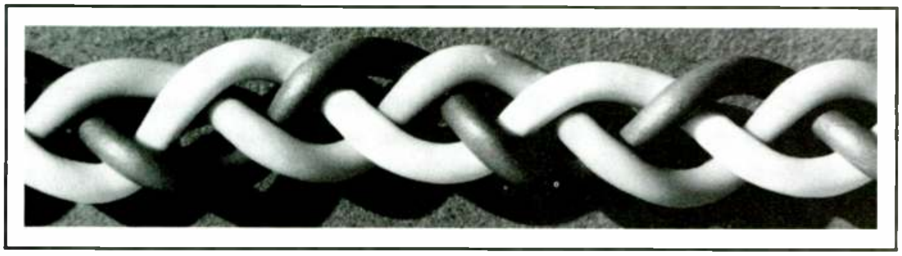


PHOTO 2: Cable constructed from PVC insulated wires (photo by Ulrich Otte).

TABLE 1					
APPLICATION VALUES					
SINGLE RUN (m)	NOMINAL LENGTH (Ω)	FINISHED LENGTH (Ω)	X2	X3	X4
1	0.064	0.08	0.04	0.027	0.02
2	0.128	0.16	0.08	0.053	0.04
3	0.192	0.24	0.12	0.080	0.06
4	0.256	0.32	0.16	0.110	0.08
5	0.320	0.40	0.20	0.130	0.10
6	0.384	0.48	0.24	0.160	0.12

TABLE 2	
Z VALUES	
Z (Ω)	5% (Ω)
8	0.4
4	0.2
2	0.1

To construct cables, decide on the required finished length, allowing for the 25% braiding loss in length. Use two different color strips of wire and one length of nylon fishing line of approximately the same diameter. Place the ends of the cable and nylon in a vise (*Fig. 1*) after stripping 5mm off the insulation.

Now start plaiting the strips; place wire 1 over wire 2, 3 over 1, 3 under 2, 1 under 3, and continue in this manner (*Fig. 2* and *Photos*). Whoever does this work (and it *is* time-consuming) must try to get the wires to cross over at 90° to cancel any magnetic fields caused by current flow.

Once you've decided on the number of bundles, place them in neoprene or heat-shrink tubing. If you are looking for a high-quality loudspeaker connector, I recommend Edison Price's Music Post.

To help you decide on the number of bundles you'll need for your application, I have drawn up *Table 1*. James Moir recommends you keep DC cable resistance at or below 5% of the nominal loudspeaker impedance (*Table 2*).

[See the author's related article, "A Musical Link," in TAA *3/91, p. 36.— Ed.]*

REFERENCES

1. Colloms, Martin, "A Cable Survey: Speaker Wires," *Hi-Fi News & Record Review*, July 1990, p. 51.

2. Moir, James, "Loudspeaker Cables," *Hi-Fi News & Record Review*, May 1979, p. 67.

SOURCES

Music Link
Roentgenring 9
D-403 Ratingen
W. Germany
20AWG teflon-insulated, silver-plated, solid-core wire

The Tweak Shop
2151 Riesling Way
Santa Rosa, CA 95403
Edison Price Music Post

ABOUT THE AUTHOR

John Jackson is 43 years young. He's been interested in music and audio since his mid teens. He studied mechanical engineering at Slough Technical College in England. Besides DIY audio, his hobbies are squash racquets, off-road cycling, wind surfing, and gourmet cooking. He also collects LPs. He has been working as a flight attendant for the past 13 years.

PREVIEW
Elektor Electronics USA

November 1991

- Class A amplifier
- Digital compact cassette
- Function generator, Pt. 2
- Audio Spectrum shift techniques
- 24-bit full-color video digitizer
- Four-terminal networks, Pt. 2
- Computer-aided electronics design

SPEAKER-TO-EAR INTERFACE

BY DONALD F. SCOTT

It seems appropriate in these days of emphasis on interconnects to give the most crucial one more consideration than it has been receiving. In any speaker listening situation, two kinds of sound reach your ears—direct and reflected—and two types of listeners are on the receiving end—casual and definitive. Probably the most typical environmental listening space is similar to that of Charles E. Ulrich's (*SB* 1/87, p. 32) in which he expertly researched and documented various geometric listening schemes to devise the most desirable one.

To get this on a scale for comparative purposes, let's say he came up with a mixture of 70% direct to 30% reflected sound reaching the ears—30% being the voice of the room. Three other widely used arrangements in similar environments are:

1. Speaker diaphragms specially arranged to give a 30% direct to 70% reflected mixture, sometimes called "stereo everywhere."
2. Similar to the above arrangement, but with diaphragmic output aimed to give an approximate 10% direct to 90% reflected mix, most often called "concert hall sound."
3. Use of the electronic synthesis to simulate any of the above and beyond even to placing the listener in the Astrodome, if desired.

A seldom used approach is to attempt to make a sonic copy of diaphragmic output and rely on the recordings to make the musical statements.

This has been quite successful using the approach outlined in *SB* 3/89 (p. 61). Using the above scale, it is judged I'm listening to an acoustic mix of about 90% direct to 10% reflective. Adding another 90 ft² of acoustic foam scarcely made an audible difference, leading me to believe that 100% direct, as in an anechoic chamber, is unnecessary or possibly even undesirable.

In the almost five years I've been enjoying and reassessing my musical and sound library, I've made only one upgrade: that of getting a handle on speaker box diffraction distortion. Presumably many have read Robert C. Kral's exposé on diffraction (*SB* 1/80, p. 28). I almost completely ignored this on the basis of his admonition that in an ordinary room this type of disturbance was usually covered up by reflective and diffractive room distortion. But now, in a controlled room, it is an entirely different situation.

Pertinent facts gleaned from Mr. Kral's article are that rectangular shapes, such as my satellites, are good generators of diffractive disturbances; box dimensions determine how low in frequency disturbances go (my 21 liter boxes go down to about 950Hz); and box surfaces also act as reflectors, just like any other hard room surface.

Obviously, tweeter rings would be only a start, so I covered the front, the two sides, and the top with 2" foam left over from room activities. I flared driver cutouts to prevent restriction of the sound window. I stuck on everything with mounting tape for easy removal. Only one removal was needed to convince me to stick it on permanently.

You would think elimination of extraneous sound would result primarily in smooth response, but of equal importance is preservation of transient behavior, the lifeblood of sound. Diffraction control puts the icing on the cake.

Speaker builders prefer to have something more substantive than subjective opinions. With this in mind, I include a

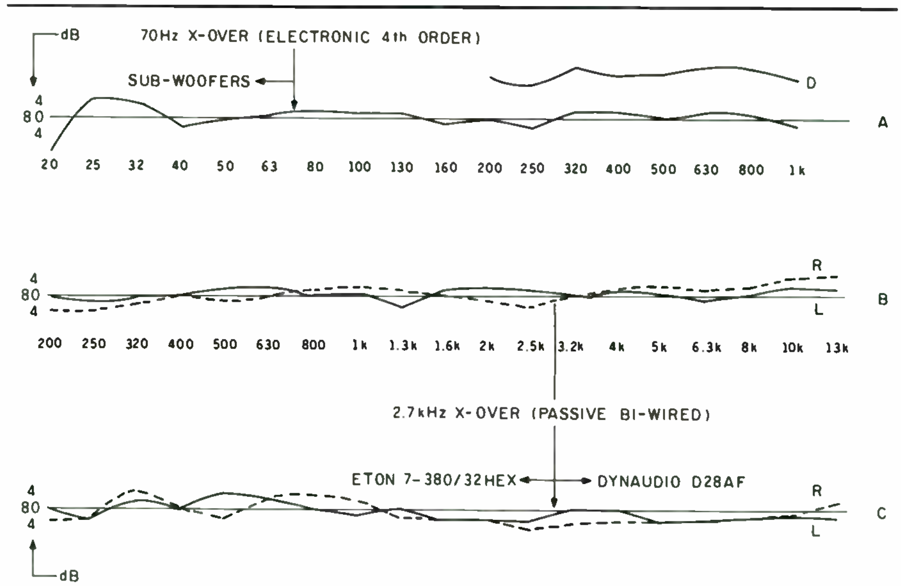


FIGURE 1: Response curves. A: L + R (mono) listening position 8′ from speakers; B: R and L close (1m); C: R and L listening position 8′ from speakers; D: sum of R and L from C. Signal generator: Old Colony Warbler KK-3; mike amp: Old Colony KP-6; readout: Heath IM-22 audio analyzer. *Stereophile* Test CD uses the Warbler for test tones—bands 20–31.

family of response curves (*Fig. 1*). Old Colony's KK-3 warbler was the signal source, and the KP-6 preamp brought the signal to where the Heath IM-22 Audio Analyzer could read it. I used three reputably flat (±2dB) omni-mikes, one at a time, and averaged the responses. Mike location was at ear height in a normal listening position.

Curve B at 1 meter establishes a reference for judgement of room control effectiveness. Curve C compares favorably with B. The slight additional roughness is attributed to residual reflected energy showing through the cracks. Curve A (mono) is what you might expect at center stage and includes fundamental voice ranges from Basso through Coloraturo. Curve D is a summation derived from L+R in C and compares well with the acoustic sum represented by A.

Directionality completely disappears at 90Hz in my room. This justifies a common subwoofer system crossing over at 70Hz. A 24dB/octave slope relieves the mid-woofer of stress even at high levels.

If you desire a 3-D soundstage, completely unrelated to speakers, don't overlook the importance of optimizing listening geometry. You need go no further than David Davenport's Focal Egg II report (*SB* 6/89, p. 49) for an excellent rundown on this. Dramatic improvements are possible with only seemingly minor adjustments.

ABOUT THE AUTHOR

Donald Scott was a musician for four years in the 1930s. He spent 40 years with The Marley Co. in Mission, KS as a construction supervisor and service engineer (troubleshooter). Audio has been his hobby since 1925.

Now for a blatant display of one-upmanship on how it sounds. The most frequently used "show off" selection represents the best in realism my library has to offer—*Jazz at the Pawnshop*, Proprius CDP 7778/9. To recognize realism, it helps to be familiar with it and I spent about two years of my short musical career playing in such a group. A musician friend said, "Holy cow, sounds like being at a front row table."

As a "bass freak," I like to display Cesar Franck's *Fantaisie in A* on Telarc's CD 80096 which displays a plethora of organ pedal tones. At exactly 5:12 he steps on Low F (21.85Hz) and holds it for 12 seconds.

Percussion devotees and everyone else are brought to attention by *Fanfare for the Common Man*, Telarc CD-80078. Others who prefer their percussion from the more normal back row are quite taken with Stravinsky's *The Rite of Spring*, London 400084-2. This plumbs the depths and still retains the "whack" of stretched membranes.

The biggest plus of all, to me, is to be free of that undercurrent of room sound that used to pervade all programs. ▶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Continued from page 46
also provided instructions for operation with their Little V subwoofer. Since the manual was written before the Sub-1s were available, they are not covered. This isn't of any consequence since the Sub-1 manual contains all necessary instructions for connecting them to satellite systems.

The manual I received didn't contain cabinet plans since the parts kit wasn't available at that time. Audio Concepts' president Mike Dzurko says he is in the process of drawing cabinet plans for inclusion in the parts kits and they should be available by the time you read this. If you purchase the parts kit, make sure you have the necessary woodworking skills and tools to properly cut and match the enclosure's odd angles. The Little V does not have the elaborate internal bracing of the Sapphire II as it is considerably smaller and doesn't need it. But, the sloping front panel will present difficulties for those used to working with rectangular boxes.

Assembly instructions for the full kit are straightforward and uncomplicated. I'll add a couple helpful comments.

When you glue the woofer crossover coil in place, position it in the right rear corner of the enclosure as you face the front panel. Mount the coil on the bottom of the enclosure. Allow 12 hours for the silicon glue to cure before proceeding to the next step. You don't want the coil to come loose after you've assembled the system.

As I've indicated in the Sapphire II and Sub-1 reviews, I prefer to solder the leads to the drivers. Doing this voids your warranty, however. Since Audio Concepts has a 15-day money-back guarantee, I recommend you first use the slip-on connector they supply. After the 15-day period has lapsed, you can carefully remove the drivers and replace the slip-on connectors with solder connections. Any damage to the drivers due to incorrect soldering technique is your problem. Audio Concepts won't replace soldered drivers.

For the best performance in terms of soundstaging and tonal balance, the AC Mini Stands are a necessary part of the Little V system. They screw into the bottom of the loudspeaker enclosures, and spikes on the bottom of the stand ensure rigid coupling to the floor. If you will be using the Little Vs in a video installation, wall or bookshelf mounting is acceptable, although the stands are still preferable. Audio Concepts sells the BT1 wall brackets for $24.99/pair. If you place the speakers on a bookshelf, avoid mounting them sideways. This defeats the time-alignment designed into the system. Also avoid placing them on a shelf higher than 30–35″ off the floor.

The 15-Day Guarantee

If, for any reason, you are not completely satisfied with your kit, Audio Concepts will refund your money in full. Since you can't audition their speakers in a dealer showroom, they offer you a risk-free way to purchase their kits through the mail.

This is an extremely generous guarantee, and I hope you understand that abuse of

Manufacturer's Specifications

Tweeter: Vifa D25A aluminum dome
Woofer: Audio Concepts AC-5, supplied in matched pairs
Frequency response: 70Hz–20kHz ±3dB
Crossover: 6dB/octave minimum phase
Crossover frequency: approximately 3.5kHz, broad overlap
Sensitivity: 89dB, 1W/1m
Impedance: 8Ω nominal, 6Ω minimum
Recommended amplifier power: 20–100W/channel
Dimensions: 11″ by 7″ by 10.5″
Height on stands: 36″
Price: full kit, $299/pair plus $20 shipping; parts kit (includes drivers, crossover, damping, and AC Cup with binding posts), $199/pair plus $12 shipping; assembled, add $50 to full kit price; optional AC Mini Stand. $99/pair plus $20 shipping
Warranty: five years on parts

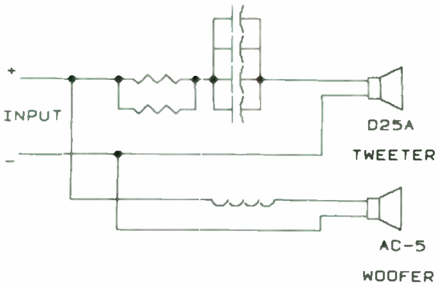


FIGURE 1: The Little V crossover. The network uses simple first-order filters for a minimum phase characteristic.

this privilege is not fair to the manufacturer. Mike Dzurko has mentioned that people occasionally order loudspeakers with no intention of keeping them. They're curious to see how they sound, knowing they can send them back when they're through listening. (Some "customers" actually admit this when they return the loudspeakers.)

The purpose of the 15-day guarantee is to prevent you from being stuck with a disappointment if the loudspeakers were not all you thought they should be (highly unlikely with the Audio Concepts systems I've reviewed). Let's treat manufacturers as fairly as they treat us. Don't place an order unless you're seriously interested in owning the product.

Measurements

The impedance curves for the two samples I reviewed are shown in *Fig. 2*. System resonance was 90Hz for both. The Little V should be an easy load to drive since the impedance never drops below 6Ω. Even inexpensive integrated amps or receivers (a distinct possibility in a video system) shouldn't have any trouble with these loudspeakers.

The ⅓-octave frequency response is shown in *Fig. 3*. I made the measurements with my Josephson Engineering C-550 measurement microphone and Old Colony Warble Tone Generator. From 400Hz–20kHz, the microphone was at a distance of 1m, positioned midway between the woofer and tweeter. Below 400Hz, I used a near-field measurement. The speakers were driven with 1W for all measurements. Little activity occurs above "0dB" (my arbitrary 1kHz reference). Audio Concepts claims a response of 70Hz–20kHz ±3dB. This doesn't mean 70Hz is the −3dB point relative to 1kHz. What it indicates is that the window of variation between these two frequencies is no more than 6dB.

My measurements come within 1dB of their claim, close enough to be considered within the normal margin for error of the measurement procedure. At 20kHz, however, a noticeable droop occurs in the response. I made a similar measurement on the Sapphire II. I have received information recently that indicates this rolloff is due to a problem with the Josephson microphone and not the loudspeaker. It seems the Josephson C-550s are 3–5dB down at 20kHz, despite the curves supplied with the mike indicating response to 20kHz ±1dB. The accuracy of my 20kHz measurement is, therefore, suspect, and I have no reason to believe the Sapphire IIs and the Little V are not operating as specified.[1]

Listening Evaluations

Before listening seriously to the Little Vs, I followed Audio Concepts' advice and gave them more than 40 hours of break-in with pink noise at a low volume level. My associated equipment was what I used for the Sub-1 review so I won't repeat the list here. I first used the Little Vs as stand-alone loudspeakers and was impressed with their performance. Although the drivers used are from different manufacturers than the Sapphire IIs, they have a definite family resemblance. These speakers sound much like miniature Sapphire IIs.

Audio Concepts describes the Little Vs as "gutsy pint-sized speakers"—an accurate description. They put out an impressive amount of sound for their size, but more than volume is at work here.

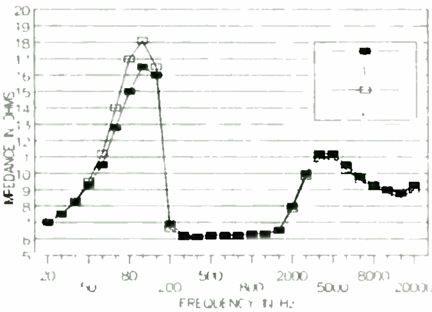


FIGURE 2: Impedance curves of both Little V samples.

to the newer stereo films which have holographic surround encoded into the two hi-fi/stereo VHS channels. Among my favorites are the two James Bond films with Timothy Dalton and *The Untouchables*. These have impressive surround effects, realistically reproduced by the Little Vs.

To hear these effects properly, you must position your video loudspeakers with as much care and attention as required for listening to music. If you insist on placing the speakers right next to the TV monitor, to get them "out of the way," you'll lose the effect. You'll probably ruin the purity of your picture as well, due to the drivers' magnetic fields. Mike Dzurko says he's working on a special video version of the Little V, using drivers with shielded magnets. If you must place your speakers close to the television, the shielded drivers will be extremely helpful. In a properly installed video sound system, the speakers should be far enough from the TV to make shielding unnecessary.

Many movies available in VHS hi-fi format contain impressive low-frequency effects which, of course, the Little Vs aren't capable of reproducing. If your video system requires low bass, it is easy to add a subwoofer. Although I haven't heard the Little V subwoofer, it appears to be a cost-effective way to add bass to your video system. You can buy a single Little V Subwoofer as a full kit for $299. To get true stereo bass, purchase two.

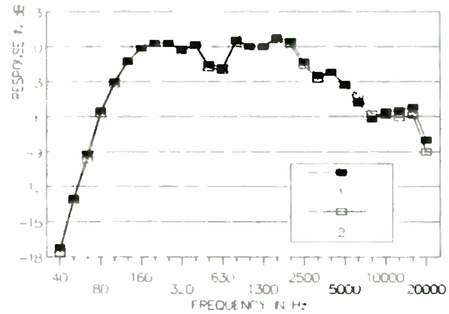


FIGURE 3: The ⅓-octave warble tone response. A near-field measurement was used below 400Hz.

Part of the family resemblance is this system's refinement. It is a clean loudspeaker, with clarity and inner detail I expect to find in more expensive products. The AC-5 doesn't offer the pristine midrange clarity of the Focal driver used in the Sapphire II. Compared to similarly priced loudspeakers from Boston Acoustics or Polk, however, the Little Vs puts the competition to shame.

If you've looked inside a pair of Boston Acoustics speakers, you can understand what I've said. A friend of mine is head of the service department at a local hi-fi/music/video store that is a Boston Acoustics dealer. I've seen these loudspeakers opened up by his service technicians. They use surprisingly flimsy drivers with tiny magnets, resonant enclosures, combined with cheap electrolytic crossover capacitors and thin, garden variety hookup wire. Is it any surprise the Little Vs, with their solid cabinets, robust drivers, polypropylene capacitors, and oxygen-free AudioQuest wiring sound better?

The minimum phase philosophy forming the basis of the Sapphire II design has left a strong imprint on these speakers as well. The result is a small speaker that produces a fine stereo image, with a realistic sense of depth. Compared to the Sapphire IIs, they reproduce a somewhat smaller soundstage, but they're superior to any other speakers I've heard for the price. As expected, the Little Vs don't have much to offer in the bass region, but they weren't designed to rattle the floors at low frequencies, and no one expects a small loudspeaker to do so. If you want low bass along with the virtues the Little Vs offer, add a subwoofer (more on that below). Where size or price is most important, however, the Little Vs perform impressively.

I have the stereo audio outputs on my VCRs connected to my stereo system. During my listening sessions, I watched several movies with the Little Vs connected to it. They make an ideal video system loudspeaker. The stereo loudspeaker systems supplied with any stereo TVs I've heard don't come close to the clarity and accuracy of the Little Vs. Their excellent imaging is extremely beneficial

System Comparisons

I spent a substantial portion of my music listening using the Little Vs with the Sub-1. This complete system, including stands, can be purchased as full kits for around $1,000. Since the highly regarded Vandersteen 2Ci loudspeakers sell for about $1,300/pair including stands, I thought this would be a reasonable comparison. The Vandersteens are fine loudspeakers, having earned a reputation as one of the best speakers in their price range; they are particularly impressive in the area of soundstage reproduction. I hear the Vandersteens often, since my listening partner, Lorelei Murdie, owns a pair. You may recall she assisted me in my evaluations of the Sapphire II and Sub-1.

Connecting the Little Vs to the Sub-1s improves midrange clarity and detail considerably. Removing the low frequencies from the mid-bass driver will invariably improve any small loudspeaker in this respect. One problematic characteristic of the Vandersteen 2Cis is the harshness and grittiness in the upper midrange/lower treble region. Lorelei observed that the Little V/Sub-1 system sounded cleaner than the Vandersteens. This was particularly noticeable in choral music, such as disc 1, track 5 of Karajan's 1959 London recording of *Aida*. The choral passages are quite harsh on the Vandersteens, but are considerably smoother on the Audio Concepts system. Massed strings also fare better on the Little V/Sub-1 system.

I believe the 4½" midrange driver used in the 2Ci may be breaking up around the 5kHz crossover frequency. Without a MLSSA analysis of the loudspeaker, it is difficult to pinpoint the cause of the harshness, but breakup is a reasonable guess. No 5¼" driver can be completely free of breakup in this region either, but the AC-5 is audibly better controlled than Vandersteen's midrange driver. When combined with a crossover frequency considerably lower, the AC driver sounds smoother, more detailed, and more "musical."

I also prefer the Audio Concepts tweeter. It is slightly more detailed than the Vandersteen and sounds a bit sweeter on massed string passages. Cymbals sound crisp, but never hard. The Vifa tweeter is, indeed, free of the problems plaguing so many aluminum dome tweeters. Like the Sapphire IIs, integration between the mid-bass driver and the tweeter is excellent.

Soundstaging is the Vandersteen's strongest area of performance—they actually rival many dipole systems. In this regard, I consider them slightly better than Audio Concepts' system. Antal Dorati's impeccable performance *and* recording of Schoenberg's *Five Pieces for Orchestra* on Mercury is a soundstage *tour de force*. The Little V/Sub-1 combination reproduces this recording with very good localization, both left-to-right and front-to-back. But the 2Cis render a slightly larger soundstage with an even greater sense of depth.

I've already commented on the superb bass performance of the Sub-1 system. The Vandersteens are still no match for the Sub-1s. Subjectively, the Audio Concepts subwoofers descend nearly an octave lower than Vandersteens, with superior definition and control. The Telarc/Monster Music demonstration CD *Signatures* (referring to the use of Monster Cable during the recording sessions) contains an impressive recording of part of

Copland's Third Symphony. This is the original version of what became *Fanfare for the Common Man*. It's not my favorite music, but the recording of the bass drum is perhaps the most impressive that Telarc has done. The Little V/Sub-1 system lets you feel the extreme low end. The Vandersteens never give you the physical sensation that accompanies the sound.

London's CD of *Siegfried's Funeral Music*, in the complete Solti recording of Wagner's *Gotterdammerung* is a regular reference disc during our sessions. The power and weight of a full symphony orchestra have rarely been captured as realistically as on this 1963 recording. Lorelei believed the Little V/Sub-1 system was a little thin in the upper bass/lower midrange region on this, and some other orchestral recordings. I agree that the Sub-1s used with the Sapphire IIs deliver considerably more weight in this region than the Little V/Sub-1 combination. But, I don't consider the Little V/Sub-1 system inferior to the Vandersteens in this regard.

Conclusions

The Little V loudspeaker is a fine performer. Used as a stand-alone loudspeaker, it delivers truly impressive performance for its modest size and price. Used with the Sub-1s, it forms a full range system unequaled for the price. Comparisons to the Vandersteen 2Cis revealed at least one trade-off, with the Vandersteens having the edge in soundstage reproduction. The Audio Concept system, however, outperforms the Vandersteens in nearly every other aspect of performance. It is free of the annoying upper midrange/lower treble harshness of the Vandersteens.

The best news is that a novice can build these loudspeakers with little or no difficulty. They are an excellent choice for your first loudspeaker project. Even fully assembled, they are a great buy.

Mike Dzurko comments:

Thanks to Gary Galo and the staff of Speaker Builder. *The review of our "LV" is an excellent example of thorough and accurate audio journalism done by a man who loves and knows music. Thanks again.*

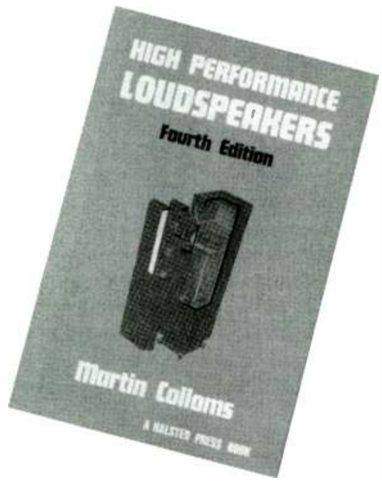
REFERENCE

1. Since I completed the Little V measurements, I have spoken to David Josephson on the phone. He has confirmed the microphone problem. He said that a number of the electret capsules used in the C-550s developed a high-end rolloff after a period of time. Since all Josephson mikes are backed by a five-year warranty, David has offered to replace my capsule and recalibrate the microphone. If you own a Josephson C-550 and suspect a problem, call him at (408) 238-6062 for instructions on returning your microphone for repair.

Book Report

Loudspeaker Techniques

Reviewed by David W. Davenport



High Performance Loudspeakers, Fourth Edition, **by Martin Colloms.** Available from Old Colony Sound Lab, PO Box 243, Dept. B91, Peterborough, NH 03458-0243, (603) 924-6371, $44.95 plus $2 S/H.

Perhaps you are familiar with Martin Colloms' work through his many articles in *Stereophile* or *Hi-Fi News & Record Review*. Or maybe you are acquainted with his book *High Performance Loudspeakers*, which has just been released in its fourth edition. It is just what the doctor ordered for the advanced amateur—not a how-to book or a compilation of journal papers, but rather, a solid introduction to the broad spectrum covering loudspeakers. This book is comprehensive enough to provide a good foundation without having to wade through the details that a professional engineer would require to practice the art. That is not to say it isn't a good starting point to acquire an in-depth knowledge of a particular aspect of speaker building—its references and bibliography are extensive.

Colloms starts the journey with a somewhat superficial general review before delving into the real meat of the subject. I would imagine the second chapter was the hardest to write. Whole books have been written on theoretical aspects of diaphragm radiators. How do you boil it down to 24 pages? The section on equivalent circuits nicely ties the significance of different system components to their effect on the audio spectrum. This type of analysis is important because it helps formulate your thoughts when analyzing a system.

The next chapter, "Transducers, Diaphragms and Technology," provides an overview of different kinds of speakers and insight into the types of trade-offs designers make in the development of a driver, such as the shape of the cone or the structure of the magnet.

The fourth chapter, "Low Frequency System Analysis," is a reasonable introduction and synopsis, but you should read something like the *Loudspeaker Design Cookbook* to realize a design. It does, however, provide good coverage of different types of systems and factors that influence them, as well as effective use of equivalent circuits. This chapter does a nice job of illustrating the interaction of enclosure volume, efficiency, and bass response. Also, fundamental performance limitations are introduced—no snake oil or attempts at fooling Mother Nature here. It is must reading for anyone who believes he has invented something that rewrites the physics books.

The chapter titled "Moving Coil Direct Radiator Drivers" is my favorite; it dissects the driver and discusses each of its constituent parts, covers spectrum from LF through MF to HF, and touches on every aspect with a clear and concise description. For example, did you know that the Neoprene used for surrounds in woofers exhibits hysteresis resulting in stiffness at higher frequencies that renders it inappropriate for use in midrange or tweeter drivers? Not the type of information you are likely to find in a popular magazine.

By far the weakest part of the book is the chapter on systems and crossovers. The section on digital signal processing is particularly disappointing. Colloms does such a good job elsewhere of explaining complex technical matters that the contrast here is glaring. He focuses on the possibilities of DSP without even attempting an explanation of the underlying principles.

In comparison, the analog material is better—a knowledgeable introduction without in-depth analysis. It provides an introduction to basic crossovers, covers the importance, and effect, of driver impedance on the result, and discusses the amplifier-loudspeaker interface. It's no secret I'm a fan of active crossovers. This chapter provides a good argument for, and introduction to, this subject.

"The Enclosure" is not about alignments—that is covered in the chapter on low-frequency analysis. It does address problems such as mechanical resonances and their solutions. It talks about enclosure materials, cabinet construction techniques, and cabinet shapes. It also contains a section on that most neglected of all components: the loudspeaker stand.

I am pleased to see Mr. Colloms chose to devote a large chapter to component evaluation—subjective as well as objective. For the objective, he has a short introduction to each test. These tests cover driver parameters like suspension compliance and coil inductance, as well as complete loudspeaker system tests like Doppler distortion and phase response. The chapter also has a good introduction to computerized testing such as MLSSA and the Audio Precision test set, as well as discussing high-tech stuff like laser measurements.

The discourse on subjective testing is basic material—covering environment, positioning, and acoustics, as well as speaker factors. It provides a consolidated reference for the experienced listener or an introduction to the subject for someone getting acquainted with subjective evaluation.

Although international standards are addressed, only a single page is devoted to this subject, basically a listing. I was left wanting more—perhaps a paragraph explaining each.

Comparison to Earlier Editions

I'm sure many of you already own a previous edition of *High Performance Loudspeakers* and are wondering what's new in the fourth edition. I'm not one to rate a book by its cover or number of pages, but this edition is about 30 percent larger than the previous one, which in turn was 30 percent larger than the second edition. Most of these pages are due to expansion of existing topics, although the book contains plenty of new material.

One hot topic these days is bandpass enclosures. A new, large section on this topic is included, as is a section on large film transducers, and another on ribbon speakers. Digital technology makes a debut with a mention of a hybrid digital loudspeaker patent, in addition to material on digital filters. The book mentions the Isobarik enclosure, resistive chamber coupling, line sources, and the moving coil spaced dipole, as well as an expanded discourse on full-range units. The section on amplifier-loudspeaker interface is expanded, as are those on cutting down resonances, loudspeaker stands, and testing.

And, of course, we can't forget the new appendix. *SB* readers will be familiar with the material covered in the appendix on CAD software; it was compiled with help from Old Colony and covers all the programs advertised in these pages.

Conclusions

I have never been one to plug numbers into a canned formula to obtain results I don't understand. I don't even like to rest on an intellectual understanding of a subject; rather, I am comfortable only when I gain an intuitive understanding, that gut feel of what it's all about. Only then can I let my creative juices take over and not be afraid they will lead me to some strange conclusion where the laws of physics are broken. For me, *High Performance Loudspeakers* fills the bill, providing a knowledgeable introduction for the knowledgeable amateur.

All in all, the book was devoid of factual errors, and the few typos that crept in and misreferences to figures were only a minor distraction. *High Performance Loudspeakers* is a must addition to the library of anyone who considers him- or herself, or strives to be, a serious audio amateur.

When you write for HELP!!!

1. Please enclose a self-addressed stamped envelope (SASE).

2. Leave room for replies on your sheet of questions.

3. Don't ask for authors' addresses. They like their privacy too. We will forward your letter if, and only if it contains a reply envelope. If the author lives abroad, enclose an addressed unstamped air-mail envelope and one international postal reply coupon, available at your post office.

4. Please resist the temptation to use us as design consultants, research assistants, product index clerks, project troubleshooters or equipment evaluators. Doing so slows up our editing.

5. If you have a problem with one of our circuits, fully describe the problem and what you have done about it. Include all significant voltage and/or current readings, and don't forget that SASE!

Craftsman's Corner

Semi-Horn Loaded Loudspeaker

When I started building speakers, because high-powered amplifiers were too expensive, I began looking at high-efficiency designs with wide dynamic range. The type of loudspeakers that always gave me the most bang for the buck were horn-loaded designs. If you need a simple, high-efficiency loudspeaker for home, studio, or club use, I offer my semi-horn loaded design based on Klipsch's venerable "La Scala." It incorporates several refinements and improvements I believe make it superior to the original.

Design Goals

The original La Scala is a good horn-loaded design with a curable ill. Low-frequency notes cause the cabinet's sides to resonate, coloring the sound. If you own an original pair of La Scalas, you can modify them by adding the auxiliary side panels (¾" veneered plywood) to increase the side panel thickness to a total of 1½." This damps almost all of the resonance since this mod raises the resonant frequency to a point above the bass horn's cutoff point.

Driver Selection

I used a 15" woofer in a simple folded bass horn, a professional 6½" hybrid midrange driver, and the Electro-Voice T35-A horn tweeter. The T35-A is the tweeter Klipsch uses in their top three fully horn-loaded loudspeaker systems. Since no suitable midrange horn/driver combination that goes down to 400–500Hz fits my enclosure and costs less than the rest of the speaker, I settled for a new driver from Audax (Polydax in the US), who manufacture four exceptionally high-efficiency drivers: PR17HR70, PR17HR90, PR17HR100, and the new PR17TX100.

The PR17TX100 has a good price/performance ratio, high power handling, excellent dynamic range, and high sensitiv-

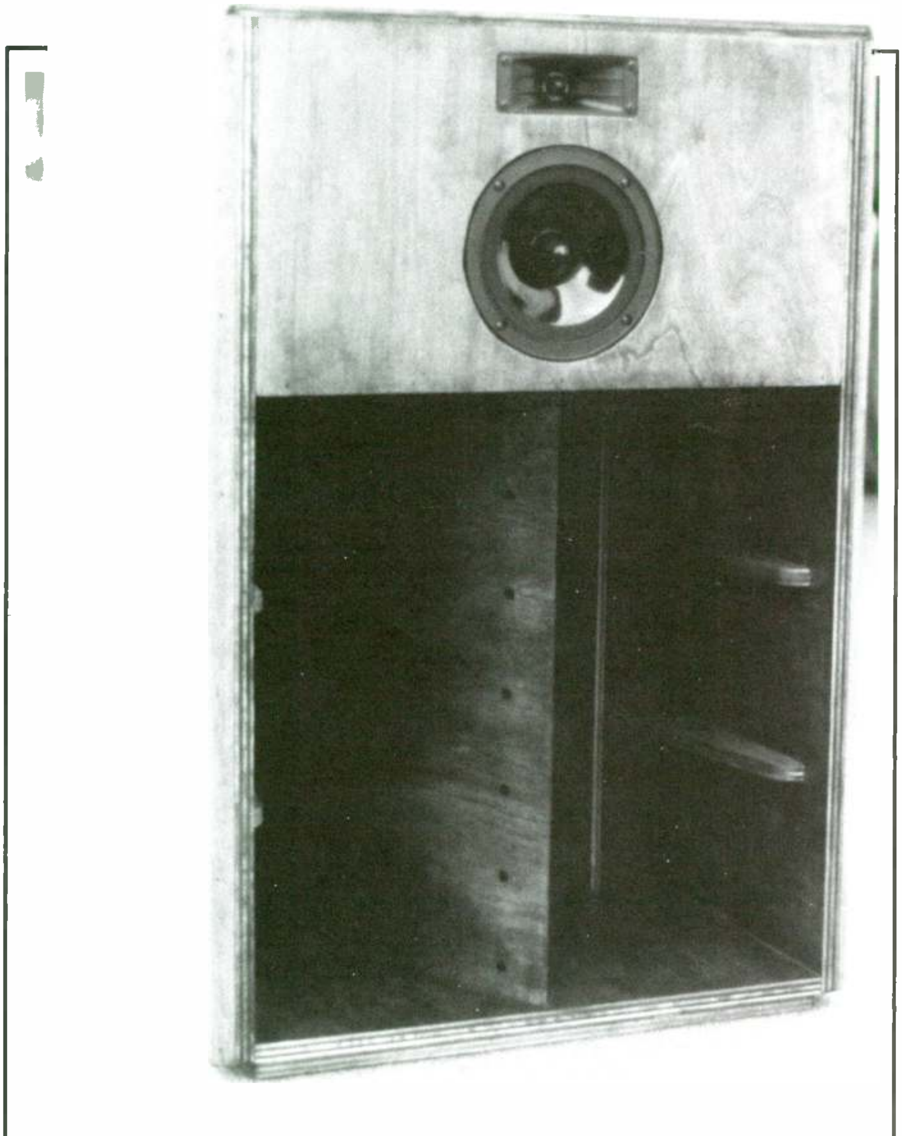


PHOTO 1: The semi-horn loaded loudspeaker.

PHOTO 2: The PR17TX100 driver about to be inserted into the speaker.

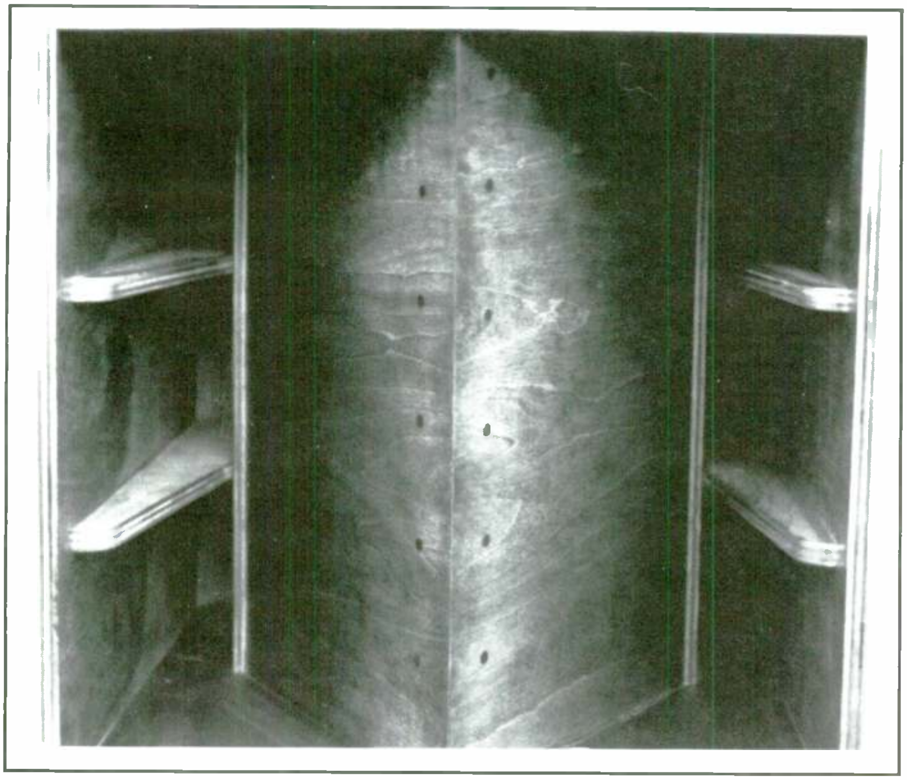


PHOTO 3: A closeup view of the interior.

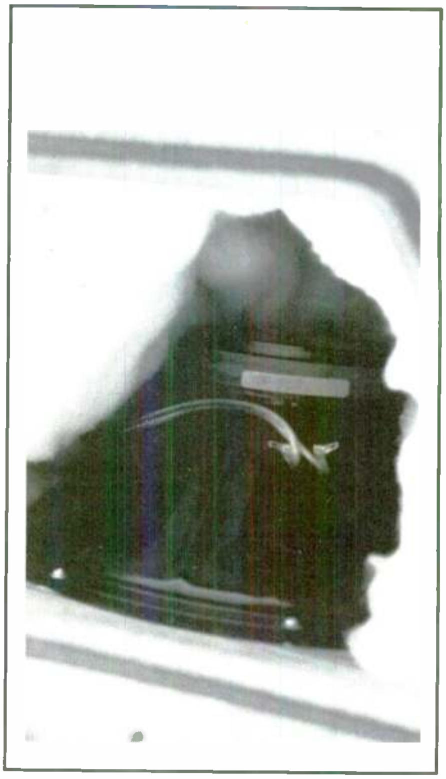


PHOTO 4: The driver in place.

ity. The Polydax literature convinced me that their proprietary TPX cone material produced the least colored sound compared to treated paper or polypropylene cones. It combines the impact of a compression horn driver with the smooth response and quality of nonhorn loaded drivers.

Few woofers today are suitable for use in bass horns. I wished to use a 15″ woofer in my design. The JBL 2220H hadn't an optimum throat area. After a long search, I chose the Electro-Voice EVM-15L (not the EVM-15B), which costs about $190. Its optimum throat area is 74.4 in², very close to my 78″ size. You could also use a woofer suggested in Bruce Edgar's "Solving the Klipschorn Throat Riddle (*SB* 4/90, p. 28). I am now using an Eminence 15″ #15587 with good results. Since two of these cost less than half of a single EVM-15L, they are even more attractive.

Bass Horn Design

The throat on the bass horn is divided in half, due to its folding geometry. Two 3″ by 13″ sound paths form the total 6″ by 13″, 78 in.² throat. My folded horn uses a version of the "Rubber Throat" designed by Paul Klipsch. Together, they give the woofer an initial flare rate of about double the rest of the horn, producing more output (SPL) in the bass horns' upper octave region of 200–400Hz.

Peter J. Groth
Clinton Corners, NY 12514

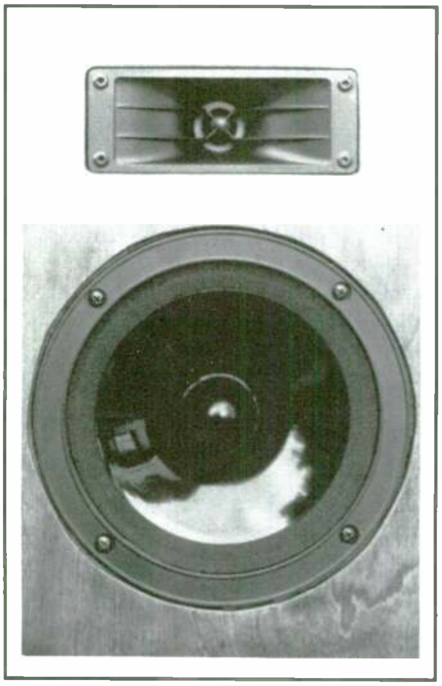


PHOTO 5: Closeup of the tweeter and driver.

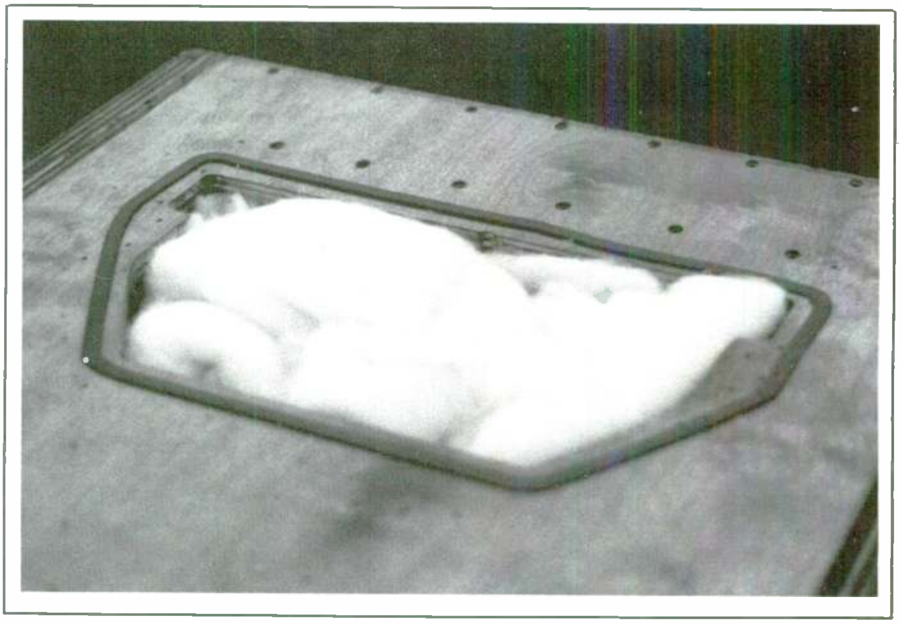


PHOTO 6: A view of the author's completed modified design.

A PC FREQUENCY GENERATOR

After having relied on inconsistent loudspeaker manufacturer's datasheets for several years to tell me driver f_s and Q, I determined them myself using the methods outlined in the *Loudspeaker Design Cookbook*. The only problem was the high cost of a function generator and a frequency counter.

In my youth, I had tried to make music on my IBM PC, but had never been able to create anything more complicated than "Mary Had a Little Lamb" using the pure tones generated by BASIC's Sound command. Suddenly, I had an idea. Why not use my computer to generate the frequencies I needed to determine the driver f_s? It sounded too simple to be possible, but was worth the try.

After loading BASIC and experimenting with the Sound command, I ran into my first problem. BASIC refuses to generate any pure tone less than 37Hz. I had wished to use the function generator to measure a 15″ cast steel frame woofer I had inherited from an eccentric uncle. To do so would probably require frequencies as low as 20Hz, since the driver's gauze surround had become loose with age. I was disappointed to discover that Microsoft's BASIC was attempting to protect the PC's 2″ paper speaker from excessive low-frequency induced cone excursion. To make a long story shorter, I wrote a low-level program using C that let me generate any frequency and duration.

I opened my PC's chassis and cut the two wires running to the 2″ speaker. I then drilled four holes in the plastic expansion drive slot covers and mounted two five-way binding posts (to connect a new external computer speaker), an RCA jack, and a 10kΩ potentiometer. Next, I soldered one PC speaker lead to the potentiometer input and the other wire to the pot ground and from the pot ground to the negative binding post terminal and the ground of the RCA jack. Then I soldered a wire from the middle potentiometer output terminal to the positive RCA jack and to the positive binding post (*Fig. 1*).

Now for the moment of truth. I connected a spare Focal 5″ driver to the binding posts, turned on the computer, and ran my C language function generator. First, I generated 440Hz and a smooth A4 resulted. Varying the potentiometer adjusted the output volume of the PC's internal amplifier. However, I needed an amplifier with greater output to measure speaker Q accurately. I connected an interconnect cable from the RCA jack to a 30W mono Pilotone tube amp, which I connected to the 15″ subwoofer I wished to measure. Setting the potentiometer for minimum volume, I turned on the amp.

Next I chose to generate a frequency of 25Hz. While the computer generated 25Hz, I slowly turned up the potentiometer's knob until the subwoofer started to rumble. I continued turning the knob and the output got louder until I judged the excursions were at their maximum. I turned it off, having decided it worked. Now I only need to buy a digital multimeter to determine any parameter.

Paul William
Los Altos, CA 94024

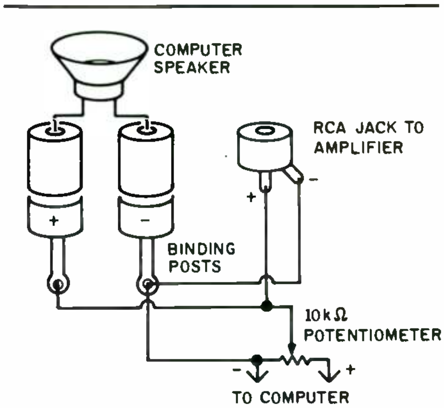


FIGURE 1: Paul William's modifications.

Paul William's frequency/warble generator is available through Old Colony Sound Lab (PO Box 243, Peterborough, NH 03458, (603) 924-6371, FAX (603) 924-9467). Without the modification: **SOF-FRQ1B5**, $9.95. With modification hardware: **SOF-FRQ1B5/H**, $19.95.

MINI TIPS

A good place to buy inexpensive tone generators and oscilloscopes is a store selling used music and PA equipment. Ottawa, a city of about 800,000, has two stores that deal in used equipment, including older, but usable scopes and generators in the $25–$150 range.

You can find lead for cabinet panel damping from demolition salvage stores. A store I frequent sells lead scraps and sheets (usually bought by stained-glass enthusiasts) for between 50¢ and $1 per pound, depending on condition and immediate supply.

Here's a lesson I learned the hard way. I got a little carried away installing multiple internal braces for my 4½ cubic foot bass cabinet, only to discover I couldn't insert one of my 10″ vents because it hit a rear brace.

Ian Dibbs
Ottawa, ON K1Z 8S4
Canada

LONG RIPS IN PARTICLE BOARD

It's been said 80% of the cuts in woodworking can be done on a table saw,[1] but most speaker builders aren't woodworkers and can't justify the purchase of such a saw. We suspect 80% of the cuts needed for speaker boxes can be done with a circular saw were it not such a problem to accurately rip down the full length of an 8′ board.

Many of us pay to have long rips done at the mill, leaving just the shorter crosscuts for home. Unfortunately, the costs for this service keep going up. One mill charges double per cut for a 5-by-8 sheet of ¾″ MDF because the operation requires two men. Another charges a minimum of $30 per half hour even though making the few cuts rarely takes more than five minutes.

After one such session we saved a piece of MDF scrap 8′ long and about 7″ wide. The width of the piece was important, for reasons we'll describe later. Since the edge of the MDF was now perfectly

Continued on page 60

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Old Colony will soon be offering a number of its kits in assembled form. If you have seen a good candidate in our ads or catalog, please be sure to let us know of your interest. Every vote counts!

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

At long last, a top-quality test mike at a great price! Mitey Mike is the answer to many an audiophile's loudspeaker testing dreams, providing flat free-field frequency response; high, undistorted SPL capability; excellent measurement repeatability; and guaranteed long-term stability. Plus, it's a snap to build! Typical specs—response (rel. 1kHz): ± 1dB, 20Hz–10kHz; ± 2dB, 10kHz–20kHz; – 3dB @ 3Hz and 25kHz; sensitivity: 39mV/Pa, ± 2dB; max. undistorted SPL: >120dBA; wideband noise level: <42dBA; power consumption: 5mW typ.; 7mW max.

Unassembled kit comes complete with PC board, mike cartridge, custom brass wand, all components, and (undrilled) blue case. For greater accuracy, a mike calibration service is available at a small additional charge (details come with kit), although most users do not find this step necessary. Tripod, 9V battery not included. From *SB* 6/90.

Other purchasing options available:

KD-2AM	Assembled Mitey Mike with calibrated cartridge	**$229**
KMW-1	Unassembled KD-2 (above) plus unassembled companion KK-3 Warbler Oscillator (case included; please see complete KK-3 specs elsewhere in this section), at a savings of $19!	**$229**
KMW-1AM	Assembled KD-2 with calibrated cartridge plus assembled KK-3, at a savings of $29!	**$349**
KMWP-1	Unassembled KD-2 (above) plus unassembled companion KK-3 Warbler Oscillator (case included) plus unassembled companion KSBK-E4 Super Switchable White/Pink Noise Generator (case included; please see complete KSBK-E4 specs elsewhere in this section), at a savings of $38!	**$289**
KMWP-1AM	Assembled KD-2 with calibrated cartridge plus assembled KK-3 plus assembled KSBK-E4, at a savings of $58!	**$449**

THE WARBLER OSCILLATOR — **KK-3**
Dick Crawford — **$99**

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%, and the output voltage is adjustable from 0 to 1V. When used with a microphone, the Warbler is more effective than 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 3 ¼″ × 3 ⅜″ PC board, transformer, all parts, and article reprint. Case included. This device is the most accurate tool available for determining a speaker system's actual performance in a room. From *TAA* 1/79.

Other purchasing options available:

KK-3A	Assembled Warbler	**$149**
KMW-1, KMWP-1	ALSO AVAILABLE IN COMBINATION WITH MITEY MIKE TEST MICROPHONE AND SUPER SWITCHABLE WHITE/PINK NOISE GENERATOR. PLEASE REFER TO KIT KD-2 ELSEWHERE IN THIS SECTION.	

SUPER SWITCHABLE WHITE/PINK NOISE GENERATOR — **KSBK-E4**
Bernhard Muller — **$79**

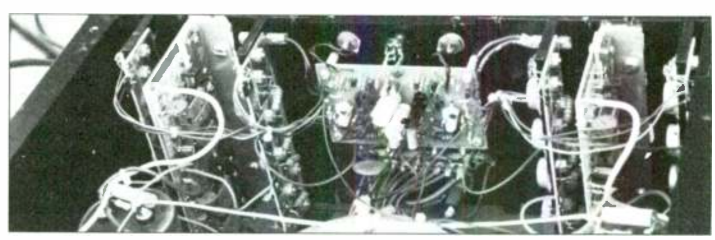
This unique kit features a stereo/mono/reverse-polarity switch that distinguishes it from other generators. CMOS digital circuits form a pseudo-random bit stream generator switchable between mono, stereo, and stereo reverse, and another switch selects pink or white noise output. Pink noise rolls off between 16Hz and 20kHz at 3dB/octave and at 6dB/octave above 20kHz, while white noise is constant through the 16Hz–20kHz range. The unit is powered by a 9V battery, not included. Included is an article reprint outlining the generator's use in audio system evaluation; the article is especially helpful with speaker evaluation methods and room placement problems. The kit comes complete with 4 ⅛″ × 2 3/16″ PC board, ICs, precision resistors and capacitors, and switches. Case included. From *SB* 4/84.

Other purchasing options available:

KSBK-E4A	Assembled White/Pink Noise Generator	**$129**
KMWP-1	ALSO AVAILABLE IN COMBINATION WITH MITEY MIKE TEST MICROPHONE AND WARBLER OSCILLATOR! PLEASE REFER TO KIT KD-2 ELSEWHERE IN THIS SECTION.	

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

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



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Continued from page 58

straight, we used it as a fence (guide) for making long rips with a 7½″ circular saw. The process wasn't difficult, and with care we could get excellent results.

Preparing the Saw

MDF is hard on steel blades. A long-lasting blade that makes smooth cuts is the 24-tooth Piranha carbide-tipped blade (about $13 at the time of this writing) from Black & Decker. The saw blade must be set at exactly 90°. It is difficult to determine this on the saw, but relatively easy (and much safer) in the work. Take a practice cut with a scrap piece, and then check the cut edge using a small square (*Photo 1*). Check at several points along the length of the cut and adjust the angle if needed.

Laying Out the Line

On the saw we use, the distance from the edge of the sole plate or shoe (the flat piece of metal on which the saw rides) to the edge of the kerf (the slot in the wood cut by the blade) is exactly 1½″ measured from the outside, and 5″ measured from the motor side. These dimensions may be different on your saw. The width of the saw teeth also affects this measurement, so measure to the side of the teeth instead of the blade body. The difference can be significant on carbide-tipped blades. Once the measurements are known, simply transfer the offset to the work piece, position the MDF guide, and secure it with a couple of clamps. A third clamp in the middle of the piece is helpful (*Fig. 1*).

Supporting the Work

A 5-by-8 sheet of ¾″ MDF is a struggle. You'll probably need a friend to help move it and a big, sturdy platform to support it while you cut. If you don't have a bench, try working on the ground. The earth is certainly big enough, and generally stable, too. First find a level spot. Space five pieces of 2-by-4s that are the same width as the MDF so the work is supported at both ends and the middle, and lay the MDF on top of them (*Fig. 1*).

Figure 1 shows the saw as commonly used by right-handed people. To rip the entire length of the board the operator may have to crawl on the panel. This isn't necessarily dangerous, just awkward. Note that a left side motor housing obstructs the operator's view of the cut and can hit the fence.

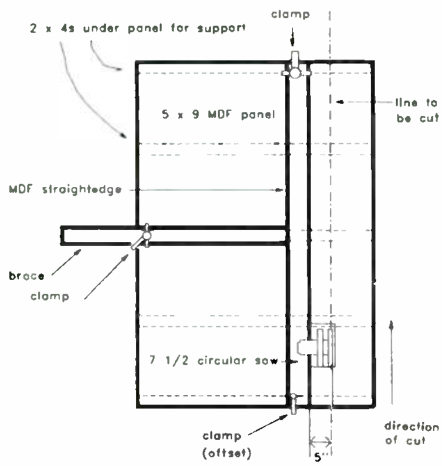

FIGURE 1: MDF panel supported by 2-by-4s allows long rips. Note width of fence is enough to offset clamps from saw motor housing.

Most of the problems can be dealt with as follows. We mentioned that the piece of scrap to be used as a fence had to be a certain width. This gives you room to offset the clamps and let the saw motor housing pass by. Set the saw's depth of cut just a bit more than the thickness of the piece. This will avoid any possibility of striking stones and ruining the blade, and will elevate the saw motor housing enough to clear the fence.

You may also reposition the guide, turn the saw around and cut from the opposite direction. On our saw this means moving the guide 3½″ closer to the line for the cut. This may be your only option if the piece to be cut is very thick and the motor housing no longer clears the fence. If the piece to be ripped is narrow and the saw base hangs into space, it will be difficult to keep the saw from tilting. Beside the possibility that the cut edge may not have a perfect 90° angle, an unstable saw is dangerous and this practice is not recommended.

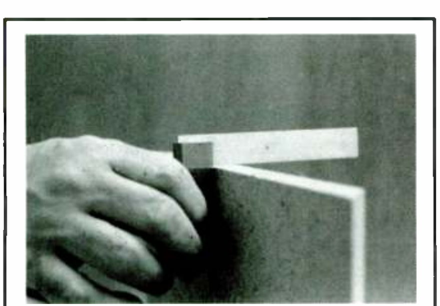

PHOTO 1: Method of holding the small square to check for a 90° angle on the edge of a board.

Cutting on a Bench

The advantage to cutting in either direction on the ground as shown in *Fig. 1* is that the cut-off piece is well-supported. If you are cutting on a bench and the MDF is substantially larger than the bench surface, you'll need to give a bit of thought to how the cut-off piece will fall. Let's imagine you need to cut a 14″ wide strip off the long side of a full sheet. In all likelihood, the bench will be narrow and a large portion of the work piece will be extending into space.

A dangerous problem is that the board may be balanced before the rip, but not afterward. If both pieces fall, nasty things could happen. Cut a small piece off a big one, not the other way around. Always plan your cutting so the bigger section remains stable. The idea is to prevent the saw blade from binding and the cut-apart pieces from falling when the saw exits the work. It's a good idea to have a helper stabilize the piece being cut off. The helper should not push or pull the piece, but simply stabilize it to float in place. Clamp the bigger piece to the bench for added security. Visualize the dimensions of the board after the cut to suggest which way things will go.

Caveat

Remember, improperly used power tools can be dangerous. Never make adjustments of any kind to the saw while it's still plugged in. Never try a long rip unless both the piece and the saw have adequate support. Both you and your helper should protect your eyes and ears! If you are not sure of your skills, get help from an experienced friend or have the cuts done at the yard.

We now have a number of scrap pieces in various lengths and thicknesses that we use for all cuts longer than a foot or two. Any object that is long enough and is known to be true can be used for a guide providing it's low enough not to obstruct the saw motor housing. In the worst case, commercial aluminum saw guides with integrated clamps are available for about $25–30 at hardware stores.

Robert J. Spear
Alex F. Thornhill
Accokeek, MD 20607

REFERENCE

1. Edgar, Bruce, "Table Saw Basics," *SB* 1/82, p. 25.

RIGID SPEAKER STAND

The ideal speaker stand should be rigid and stable, completely nonresonant, and perhaps devoid of mass, so speaker energy is not "lost" to the stand. The image I have is of a speaker suspended in the ideal position by an infinitely powerful "tractor beam." In the world of manufactured speaker stands, welded tubular metals, sometimes damped with sand or metal pellets, provide the rigidity. Steel spikes in the stand's feet, as well as the speakers' supporting platform, provide the stability. Most good stands are usually quite heavy, making it appear that rigidity is incompatible with low (or no) mass. Here is an idea, however, for inexpensive, rigid, and low mass stands, which I encountered in the integrated stand of the Nexus speakers by Linn (of Sondek fame).

Continued on page 62

OLD COLONY'S USER-FRIENDLY SOFTWARE

Speaker Designer™

Release 1.2 by Stuart E. Bonney

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

IBM 5¼" 360K DS/DD SPD-1B5

Loudspeaker Modeling Program

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

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

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Apple Macintosh 3½" SS/DD	**LMP-2M3G**
IBM PC/XT/AT 5¼" DS/DD	**LMP-2B5G**
Commodore 64 5¼" DS/DD	**LMP-1C5**

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

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Apple Macintosh 3½" SS/DD	**LMP-3M3G**

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

Driver Evaluation and Crossover Design

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

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

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

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

Active Filter Design

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

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

IBM 5¼" 360K DS/DD ACT-1B5

Two-Way Active Crossover Design

by Gary Galo (*SB* 5/88)

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

Specify:

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

Stepped Volume Controls

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

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

Apple Macintosh 3½" SS/DD SVC-1M3

BULLOCK & WHITE

BOXRESPONSE

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

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

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

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

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

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

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

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

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

Specify:

BOXRESPONSE

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Commodore 64 (Disk)	BOX-1C5	
Commodore 64 (Cassette)	BOX-3CC	
IBM	BOX-1B5	
IBM Plus Graphics		
BOXRESPONSE	BOX-2B5G	**$50**

PASSIVE CROSSOVER

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

Specify:

PASSIVE CROSSOVER CAD

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CALSOD

Computer-Aided Loudspeaker System Optimization and Design

by Witold Waldman

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

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

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

Specify:

IBM 5¼" DEMO	**CAL-2B6D**	**$ 5.00***
IBM 3½" DEMO	**CAL-2B4D**	**$ 6.00***
IBM 2 x 5¼" 360K DS/DD	**CAL-2B6**	**$65.00**
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Continued from page 60
The stand consists of three basic components: a hollow cylinder used as the stand "column," a base with spiked feet, and a pair of ¼″ threaded steel rods (available at most hardware stores) slightly longer than the hollow cylinder. The speaker, column, and base are clamped tightly with the threaded steel rods running through the hollow column (*Fig. 1*). The result is an extremely rigid stand, tightly coupled to the speaker box.

You can use any strong, rigid material for the cylinder: PVC pipe, Medite board panels, aluminum extrusion (rigid, but a bit expensive), and so on. Its cross-section can have any shape; unless you wish to fill it with some type of damping material, its circumference need not be closed—its cross-section can be in the form of a C or a U. You can also use any strong, rigid material for the base, although I have used only Medite board because it is the best inexpensive particle material to work with, especially for mounting spike feet.

You should fill the column before you assemble the stand. You may wish to experiment with more than the type of filling used, such as the quantity if it is a heavy substance. Very dense foam, metal pellets, and sand are well tried materials, though each sounds different. Try cat litter—it's inexpensive and works well.

These are not universal stands. You must tailor each one for speaker boxes prepared to accept them because you need something at the bottom of the speaker into which the rods will thread. T-nuts are best, installed from the inside of the bottom panel. They cannot be mounted on the outside because the pressure applied to attach the stand would pull them out. Some T-nuts have heavy threads on the outside (sometimes called nut-serts), designed for knock-down furniture, which you can flush-mount from the outside. Although these work, ordinary T-nuts mounted on the inside are the most secure.

Hilti steel spikes come in a variety of lengths, have a ¼″ thread on one end, and are inexpensive—perfect for speaker stand spikes. They are meant to be shot into concrete by a Hilti gun and used as attachment anchors.

With most speakers, it is best to avoid leaks, so you should flush-mount the T-nuts and seal the holes on the inside—another layer of panel board or even a couple of small pieces of ¼″ pressboard glued over the T-nuts. Or you can cut a ⅝″ or ¾″ Medite panel sized to match the bottom of your speaker, mount the T-nuts to this panel, and then screw or glue it to the bottom of your speaker. Mount the two T-nuts along the bottom panel's center line as far apart as the hollow column's depth allows; remember, the steel rods run within the column.

Aside from choosing a strong and rigid material, make the column ends as flat as possible to ensure good contact at the base/column and the speaker box/column interfaces. Your needs will determine its length, but it is difficult to make stands taller than 2′ stable.

I usually build the base out of two identical pieces of Medite board ½″ to ¾″ thick, and about 25% larger than the speaker bottom. You can cut it in any shape: rectangular, circular, triangular, and so on. After you have flush-mounted the T-nuts in the corners of one piece (for spikes), glue/screw the two pieces together to make a 1–1½″ base with the T-nuts embedded between them. It is easier to install the T-nuts from the base's bottom, but they may come loose as you tighten the nuts on the spikes. Nut-serts are more convenient and work fine. Drill two ⅜″ holes in the base to match the T-nut holes at the bottom of the speaker. It is a nice touch to rebate these holes into a ¾″ diameter, ⅜″ deep hole.

Cut the ¼″ threaded steel rods to the proper length so after you completely thread them into the T-nuts embedded in the speaker box's bottom panel, they will protrude approximately ¼″ beyond the base. (Be careful not to damage the thread during cutting.) [*Thread two hex nuts on the threaded rod above the cut and clamp them in a vise before cutting. Remove the nuts afterward to restore any distorted threads.—Ed.*]

Align the base and cylinder so they are centered as you mount a washer and nut on each steel rod to attach/assemble the stand. This procedure is best performed with the speaker upside down. Tighten the nuts as much as practically safe and possible. Finally, install the bottom spikes, position your speaker/stands.

Mike Chin
Vancouver, BC V5W 2V7
Canada

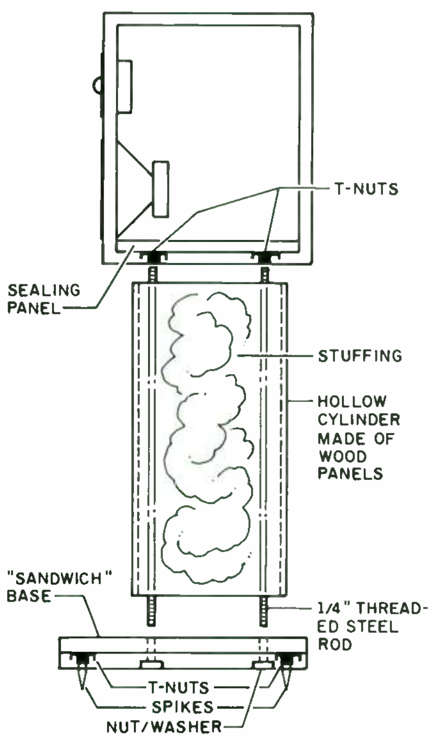


FIGURE 1: Speaker, column, and base clamped tightly with threaded steel rods running through the hollow column.

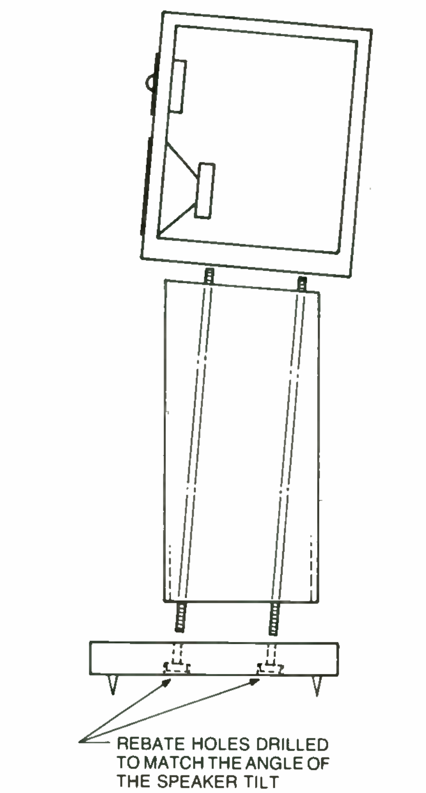
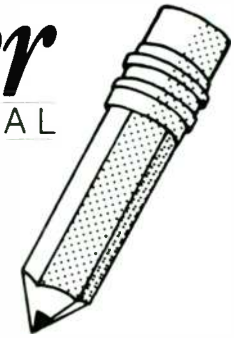


FIGURE 2: To tilt the speaker back, fabricate a column with the desired angle and ensure the correct placement of the T-nuts in the speaker box and holes in the base so the steel rods fit properly.

SpeakerBuilder
THE LOUDSPEAKER JOURNAL



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INCORRECT FORMULA

I have discovered an error in my article on modifying the Minimus 7 (*SB* 4/91, p. 38). The formula in the second column on page 39 should be C = 1 / (2πRf) rather than the published C = ½πRf.

James Lin
Galveston, TX 77551

WRONG WAVELENGTH

I believe G.L. Augspurger's informative article "New Guidelines for Vented-Box Construction" (*SB* 2/91, p. 12) contains a small error. On page 16, he states that the organ-pipe resonance of ducts used in vented speakers will occur at a quarter wavelength. That is, the wavelength at the resonant frequency will be four times the apparent length of the pipe. Actually, the resonance will occur at a half wavelength, since such a vent is open at both ends. A pipe with one open end and one closed one will resonate at a quarter wavelength as Mr. Augspurger suggests.

You can understand this point in a few intuitive ways. Consider a pipe with one closed end. At resonance, a node is at the closed end where pressure change is maximum and volume velocity change is minimum during cycles of the sound. Another node is at the open end where volume velocity change is maximum and pressure change is minimum. A pipe with two open ends can support only resonant modes that create symmetrical conditions at its ends, since the pipe itself is symmetrical. At resonance, a doubly open pipe will have a node at its center where pressure change is maximum and volume velocity change is minimum, and nodes at each end where volume velocity change is maximum and pressure change is minimum.

To try a simple but convincing experiment, take a short length of ordinary water pipe (say a foot of ½″ or ¾″ diameter copper tubing), hold one end closed with your palm, and blow across the open end. You will strongly excite its primary resonance, and if you do the calculation, you will find that the wavelength of the excited tone is four times the length of your pipe. Now with both ends uncovered, blow across one open end. This method will not strongly excite the new resonant frequency, but if you listen you will hear a soft tone one octave higher than that which you heard before.

Tom Sharpe
Bedford, MA 01730

G.L. Augspurger replies:

You are absolutely right and I was wrong. This is one of those inexplicable mental lapses that seem to become invisible the moment a first draft is typed. I missed it, the editor missed it, and even Don Keele missed it.

Allowing for end corrections, you would expect a 10″ open pipe to resonate around 520Hz, not 280Hz. My graph of duct response does show a peak near this frequency. However (and this was my point), it is no greater in magnitude than most of the other acoustic garbage coming from the vent.

STANDING WAVE EFFECTS

Regarding speaker cabinet standing waves, John Dugan (*SB* 3/91, p. 95) got part of the picture, but missed out on some important points, drawing a faulty conclusion that "the purported advantage of odd-shaped enclosures does not exist." I wish to correct some misconceptions.

It is true that:

1. All box shapes will produce standing waves.

2. Narrow fronts, offset tweeters, and rounded front corners tend to improve imaging.

3. Curved panels are stiffer than flat ones for the same thickness, and so typically provide "deader" enclosures.

4. Small enclosures have their lowest-order standing waves at higher frequencies where they and the sound degradation they create are less audible. This has to do with the reduced energy present at higher frequencies, the increased stiffness of small enclosure panels, the higher frequencies of resulting harmonics, and the ear's sensitivity to sound at different frequencies.

5. Slanted-front enclosures may be used to advantage to time-align™ drivers and angle the tweeter to increase the ratio of reverberant-to-direct sound.

The above points are well-known, so I will not attempt to prove them. However, the question is not of whether standing waves exist, but of what their effect is on audibility. Poor enclosure shapes do exist, just as do poor room shapes.

Standing waves within an enclosure produce regions of high and low pressure within the cabinet as Mr. Dugan explained very well. These become audible when they cause movement of the cabinet walls or cause the speaker cone to deviate in any way from following the movement of the applied electrical signal.

It follows that the greater the variation in pressure, the more such distortion takes place. The trick, then, is to minimize the lumping together of standing wave frequencies and the resulting increase in pressure variations by choosing box dimensions that spread them evenly and by choosing shapes that position the strongest nodes away from the speaker cones. In practice, this can be done by making rectangular boxes whose dimensions are not multiples of each other (the Golden Ratio, prime numbers, Avogadro's Series, and others are good examples), by using slanted sides to spread the resonances, by using egg-shaped or similar enclosures, by using semicircular enclosures, and so forth.

I agree that marketing departments exaggerate the purported benefits of angular enclosures to differentiate their speakers from a saturated market of well-meaning, but ill-informed customers. I have yet to build a speaker that did not sound better in a box with nonparallel sides, however, especially if the midrange was being carried by a thin, lightweight cone. Lumped resonances sound terrible in a listening room. The same is true of loudspeaker enclosures.

Finally, never forget the basic laws of physics. Enclosure design is an extremely complex field not only involving shape as discussed by Mr. Dugan and me, but also panel stiffness, wall damping, driver/vent location, construction methods, bracing, and other factors. Each design will necessarily be evolutionary, beginning with sound engineering practice and being refined through experimentation to the limit of time, money, and skill of the designer. Enclosures are the one area where the home speaker builder can easily outclass commercial designs, since production time is no object.

Marc Bacon
Ste. Julie, Quebec J3E 1H6

AMT KUDOS

In 1977, I bought a pair of ESS AMT-1A speakers. I renewed the woofer and drivers through ESS. Later I replaced the crossover with components from Zalytron. Now "the sound as clear as light" came back in depth, clarity, and dynamics.

Later, I bought a pair of ESS AMT-IIs and thought I'd get the same performance. The secondhand pair cost me $300. Bass was very good, but the massive Alnico magnet couldn't compete with some of the speakers I made with Philips tweeters. Was it the way they were made or was it burnt out components?

Both mid-tweeters in the AMT-1A and AMT II were the same. I quickly rewired the AMT II's mid-tweeter like the AMT-1A's, but left the woofer section alone. That crossover used a 12mH coil and a smaller 10″ woofer and driver. This made both systems sound excellent—great dynamics and brilliant highs, midrange, and bass. Actually, the AMT II's tweeter is smoother and doesn't go up quite as high, but it's just as dynamic.

Richard Renzella
Long Branch, NJ 07740

24W100 REPLY

This is a reply to Peter Joseph's letter in *SB* 3/91 about Dynaudio 24W100 woofer-midrange drivers. I've built several successful systems with this driver and may be able to offer helpful information.

1. The Dynaudio 24W100 *is* a back-vented speaker. The circular opening on the back of the magnet assembly filled with synthetic foam is the vent. The openings on the dust cap are used for voice coil alignment only and are not open to the voice coil chamber on the finished driver. Larry Hitch at Madisound verified this.

2. The 24W100 was not designed for sealed enclosures. Dynaudio suggested alignments list three vented alignments for this driver. I built a 50 liter vented system (with some modifications) that performs well. Introduction of a moderate amount of stuffing, away from the port, around the driver improved things in the midrange and let me tune the port a bit lower than the recommended 42Hz.

3. *Wheezing sound* from the speaker. The most recent shipments of 24W100s received included a rubber gasket that allows an airtight seal to be made between speaker flange and baffle. USE IT.

Make certain it has no leaks here. If it does, the unit will wheeze. Earlier shipments (two years ago) of the speakers did not include this gasket, so sealing the flange to the baffle without exceptional torque may have been a problem.

At low frequencies, below 80Hz, the *slightest* leak *anywhere* on the enclosure will result in wheezing as air pressures, even in vented boxes, build to high levels.

4. *Driver resonances*. All the 24W100s I've purchased had free-air resonances between 32 and 35Hz. After the speakers have been used for 20–30 hours (or broken in with moderate level low-frequency

signal for 1-2 days), this may be lowered by 2-3Hz, but I observed such a shift on only one of five drivers.

When a driver is mounted in an enclosure, it will become part of that enclosure and exhibit its acoustic idiosyncracies, including its resonant modes. This can be most elegantly demonstrated by moving a driver from a 50 liter enclosure to a TL and watching the midrange response smooth to a flat line. A closed enclosure will significantly raise the driver's resonance point.

5. The distortion these drivers exhibit will depend on the crossover and cabinet design. It is easy to overlook these interdependencies and blame the driver. If we deal with an amp of reasonable quality, it is hard to beat a bi-amplified design (no crossover at the speaker) for a clean sound, the cone following the signal with a high degree of compliance.

The improvement in sound is hard to believe. I obtained the best results with the 24W100 mounted in an 8′ transmission line, that is, incidentally, an *identical* design described by Roger Sanders (in his reply to Gary Beckstrom, also in the 3/91 issue of *SB*) except for the dimensions scaled to the 24W100 (40″ by 12.5″ by 17.25″). The crossover is at 2.5kHz to a D28AF.

6. *SPL*. A pair of 24W100s in a transmission line enclosure will fill a large listening room (28′ by 15′ by 9′) with performance hall sound levels. The maximum cone excursion is 26mm p-p. That prevents the speaker from bottoming out even by Billy's guns, in John Eargle's recording of *Billy the Kid* by Copeland (Delos 31042). (Note: *linear* maximum excursion is 8mm p-p.) 80W/channel will comfortably drive these speakers.

The Dynaudio 24W100 comes highly recommended by this speaker builder.

Andrew Tanos
Cleveland Hts., OH 44118

DOUBLY VENTED DESIGNS

After having read in *SB* 3/91 about the electrical filter equivalents for various box types, I became interested in the design of a doubly vented bandpass enclosure. Where do I need to look at the current to find the efficiency and the excursion limits for the system? Also, could you tell me how to convert those figures into real world equivalents like Eta (efficiency) and X_{MAX} (cone displacement)? Furthermore, could you tell me anything about the ratio of f_3 to total cabinet volume and how that ratio compares to other box designs?

Dave Degelau
Madison, WI 53703

Roy Mallory replies:

Thank you for your letter. I assume the doubly vented bandpass enclosure you talk about is the one I described in *Fig. 11* and *Table 10* (*SB* 3/91, p. 25) in my article.

Determining efficiency using PSpice is simple. Let PSpice plot the frequency response using the formula I give near the bottom of *Table 10*. Be sure to use the V_{AD} normalization for AC analysis given in *Table 1*. This normalization causes the frequency response of any direct-radiator loudspeaker enclosure to reach a value of 1 in the passband.

A bandpass enclosure's output in general won't be exactly 1 in its passband, but will probably range somewhere between 0.5 and 1.3 for most alignments. To calculate efficiency, scale the manufacturer's sensitivity rating by the normalized passband output. Because sensitivity is almost universally given in logarithmic form (typically dB at 1W/1m), the output given by PSpice must be converted to logarithmic form before the scaling is done. Use the following formula for the conversion:

$$\text{log output} = 20\log_{10}(\text{output})$$

Then add the result to the manufacturer's number.

Let's do an example. Suppose the manufacturer's sensitivity rating is 90dB and PSpice predicts an output of 1.2 in the passband. The log of 1.2 is about 0.08. Multiply this number by 20 to get 1.6, and add this to the speaker's sensitivity of 90dB to get 91.6dB 1W/1m.

PSpice can also provide a plot of absolute cone excursion versus frequency. Small[1] gives the following formula for peak cone displacement:

$$X_D = P_E^{0.5} \sigma_X k_X X(s)$$

Where x_D is the peak cone displacement in meters, P_E is the power input to the speaker, σ_X is the DC displacement sensitivity of the unloaded speaker, k_X is the system displacement constant, and X(s) is the normalized system displacement function.

Small shows that sigma can be calculated using the formula:

$$\sigma_X = \sqrt{V_{AS} / (2\pi \rho_o c^2 f_S Q_{ES} S_D^2)}$$

Where Q_{ES} is the total electrical Q of the speaker. All other parameters are defined in *Table 3* of my article.

The normalization for cone displacement in my article is effectively the product of k_X and X(s). Therefore, to plot the peak cone displacement versus frequency, normalized to 1W input power, simply multiply the cone normalization expression given in *Table 1* by σ_X. This product results in a new cone displacement normalization of:

$$V_{AD} = (1 / 2\pi S_D) \sqrt{1 / (2\pi f_S Q_{ES} C_{AS})}$$

Finally, you ask about cabinet volume. Since PSpice solves networks (circuits) numerically and not symbolically, it cannot directly provide you with an expression that relates box volume to f_3. Since the doubly vented bandpass enclosure type is patented by B__e Corp., the chance of finding any published alignment information on it are probably somewhere between zero and none. Therefore, I believe you have two options. The first is to solve the system equations for the enclosure. The second is to pick a woofer and iteratively determine V_B for different bandpass alignments and compare them to known direct-radiator enclosure types.

I hope I have satisfactorily addressed all your questions.

As a side note, a subtle typesetting error crept into my article. In all cases where I gave examples of PSpice circuit files, some lines were too long to fit into the allotted width and so were broken into two lines. Although humans can deal with broken lines, PSpice cannot. Any file entered in such a manner will produce an error message.

REFERENCE

1. Small, R., "Direct-Radiator Loudspeaker System Analysis," *JAES*, Vol. 20 #5, June 1972, p. 383.

PSpice VERIFICATIONS

When I saw the article "Calculating LF Response With PSpice" in the 3/91 issue of *SB* (p. 20), I read it with intense interest. I, too, have been working with PSpice to attempt to determine system responses. Let me start with a few questions concerning the articles so I can confirm the calculations presented and verify that I am following the steps and procedures correctly.

It was stated that V_{AB} was selected from a table, but I did not see a reference to its actual value. I suspect something close to 0.0256m³ (0.9 ft³). Is this close?

For port calculations what are the units of measure for the port length and radius? I suspect they are meters, correct? What length and radius were used in the examples? My guesses here are 0.004191 meters long by 0.00762 meters radius (1.65″ by 3″).

Formulas for Q_L, Q_A, and Q_P were given in *Table 1*. Values of R_{AL} were suggested to be based on a Q (Q_L?) of 7. Are there simple methods of coming close to approximate values of these resistors?

My PSpice approach, though unsuccessful as yet, has been to model the driver as shown in *Fig. 1*. The intent is to relate R1 to a function of voice coil resistance R_E. Coil L1 relates to voice coil inductance L_E. The tank circuit of C2, R2, and L2 would be related to the driver resonant frequency f_S and the driver Q, further modified by enclosure impact. The values of R3, L3, and C3 would be included relative to port parameters where applicable. I believe that most of the values can be manipulated from the various Qs of the driver, box, and port. Thus, it might be possible to reduce the mechanical type calculations and stay more in the electronic domain (which is the world of PSpice anyway).

What I had hoped to gain from this setup were input impedance curves and full-range output sound pressure levels. This would allow the simulation of a cabinet with multiple drivers and the associated crossover and pad circuitry. Has anyone given any consideration to this kind of an approach?

Stan Rohrer
Beavercreek, OH 45430

Roy Mallory replies:

You have asked several questions I will attempt to answer in order. First, you ask about a value of V_{AB}. I assume you are referring to the value selected for the acoustic suspension design. To calculate V_{AB}, take the value of C_{CAB} given in *Table 6* and multiply it by the density of air and the speed of sound squared as indicated in *Table 1*. The value is indeed 0.0256m³ as you state.

Second, you are correct: the units for port length and diameter are meters. In fact, as I mentioned in the article, all units are MKS (meters, kilograms, seconds). It is important to note that no explicit length or radius was selected for the port used in the bass-reflex example—only the equivalent inductance was used. Inspection of the formula given in *Table 1* that relates a port's equivalent inductance to its length and radius shows that there is not a unique mapping between inductance and port dimensions. You can always specify any radius, and calculate a length that makes the inductance value come out correct (or vice versa). Practical considerations, however, usually require port diameter to be between 2″ and 4″.

Next, you ask about resistor values to model cabinet Q. I used Small's[1] suggestion that cabinet losses in reflex enclosures can be modeled as leakage losses, and so calculated a value of R_{AL} based on the formula for Q_L given in *Table 1*.

Finally, there are, I think, several implicit questions in your discussion of an alternate model for a loudspeaker and enclosure. I'm afraid I must address them in a general way, as to discuss them rigorously would require a long diversion into dynamic analogies.

The circuit you show is topologically correct, although your description of what the various components represent is not correct in some cases. See Small[1] to get formulas for calculating the various component values. The topology you show is the electrical equivalent circuit. I used acoustical impedance analogous circuits in my article because I believed they were the simplest for the intended purpose. For enough insight into dynamic analogies to allow you to model various cabinet configurations with impunity, I suggest you consult Beranek,[2] the aforementioned Small article, and the appendix in an article by Leach[3] where he describes using a gyrator to model a loudspeaker. I should warn you that unless you have a math or engineering background, you might find this reading frustrating.

You certainly can use PSpice to model a whole speaker system including crossover/driver interactions. Although I have not read of anyone doing so, I can only imagine that any number of professional speaker designers have done so, given PSpice's ubiquity, power, and low cost. There are some caveats though.

First, voice coils act as if they had semi-inductance, not inductance (see Vanderkooy[4] and Koonce[5]). Second, phase and frequency response effects due to cone and suspension parameters must be modeled. In general, the behavior predicted by the simple types of models we have been discussing are only accurate within the piston range of the speaker: that is, within that range where the cone can be considered to act as a perfect piston on the air. So beware and good luck.

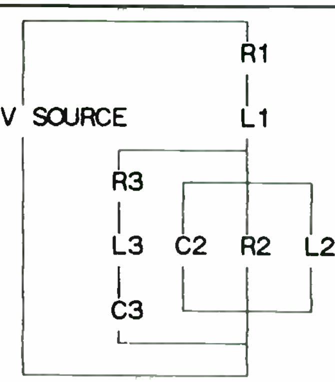

FIGURE 1: PSpice modeled driver.

REFERENCES

1. Small, R., "Vented-Box Loudspeaker Systems Part 1: Small-Signals Analysis," *JAES*, June 1973.
2. Beranek, L.L., *Acoustics*, Acoustical Society of America, 1986.
3. Leach, W.M., "On the Specification of Moving-Coil Drivers for Low-Frequency Horn-Loaded Loudspeakers," *JAES*, December 1979.
4. Vanderkooy, J., "A Model of Loudspeaker Driver Impedance Incorporating Eddy Currents in the Pole Structure," *JAES* March 1989.
5. Koonce, G.R., "Crossovers for the Novice," *SB* 5/90, p. 26.

SOLVENT SOURCE

In the letters section (*SB* 3/91, p. 93), Angel Rivera asks about a solvent for silicone rubber. V.M. & P. Naptha can be used to thin most silicones and is available in good hardware stores. To determine if it is compatible with your type of silicone, try a test patch to ensure it doesn't inhibit a full cure of the rubber.

Roy Mallory
Bedford, MA 01730

SILICONE THINNERS

Regarding Angel Rivera's letter, "Watered Down Silicone" (*SB* 3/91, p. 93), also see "The AR-1 Rejuvenated" (*SB* 2/82, p. 7). From my experience using acetic acid based silicone, I have used isopropyl alcohol to wet the stuff and allow it to become highly spreadable. Use your fingers or a brush. If too much sticking occurs, add more alcohol. I've used 98% or 90% isopropyl alcohol and even 70% rubbing alcohol successfully. Be careful around old and damaged foam. The alcohol can make it extremely soft until cured.

Oddly enough this doesn't work with alcohol-based silicone or similar "electronic grade," which I have recommended using around metal or electronic parts. That includes GE RTV-162 or Permatex Ultra Blue #81724 which may require naptha or xylene to thin.

I've stayed away from the "new" silicones, which are water cleanable. They don't seem to set as quickly. I don't think you should water down speaker cones.

For those interested in a two-part pourable silicon elastomer, try Dow Corning Sylgard #184. GE has a two-part mix (RTV-615) available from Newark Electronics. Dow Corning offers a pourable one-part white dispersion coating (#236) leaving a 4–5 mil. thick coating after room temperature curing.

Greg Szekeres
Pittsburg, PA 15236

WOOFER/MID PLACEMENT

I am writing to comment on Ralph Gonzalez's Delac S10 article (*SB* 3/91, p. 32). I have experimented with near-field constructive/destructive interference speaker alignments since 1986 and have found that forward/rearward placement of drivers gives a holographic and focused soundstage that he had mentioned. I'm building a pair of dipole speakers using eight SEAS 11FG-X drivers. My major concern is his placement of the two woofer/mids to reduce "reverberant response step."

Most single speakers including dipoles may be classified as single-slit inferometers. Sonic inferometry is easily demonstrable to anyone owning a pair of ordinary loudspeakers. Sitting in front of a speaker you can hear the high, midrange, and low frequencies equally well. As you move slowly to the side of the speaker cabinet, the high frequencies begin to diminish and then in some cases get louder and fade again. In a poorly designed multispeaker cabinet, midrange may also fade rapidly. The angle at which this fading begins is geometrically related to the wavelength and diameter of the speaker driver:

$$\sin \Theta = \lambda \ / \ \text{speaker diameter}$$

Where λ is the wavelength in millimeters and speaker diameter is the diameter of the individual speaker in millimeters; Θ is the angle from the driver axis at which destructive interference occurs and at which those frequencies will fade out.

Electrostatic speakers have incredible depth and a realistic image partially due to destructive interference. The sound emitted from the front of the diaphragm is 180° out of phase with the sound emitted from the rear. This causes almost total cancellation of all sound near 90° to the normal of the surfaces. Although this makes them inefficient, the cancellation prevents wall reflections, which smear and destroy spatial clues important in imaging.

Using two drivers for the same frequencies is classified as a double-slit inferometer. This changes the aperture of the soundfield by narrowing it. The destructive interference is now more dependent on the spacing between the two individual drivers.

$$\sin \Theta = \lambda \ / \ \text{distance between the speakers}$$

Now back to Mr. Gonzalez's Delac S10, the two woofers are rotated about a horizontal axis giving a sharp rolloff above the tweeter at frequencies above 2.1kHz, considering the distance between the virtual centers of the woofers is 15cm. This will also give an effect that the upper midrange music starts to "climb" the rear wall until the tweeter kicks in.

A suggested cure for this would be to rotate the woofers about a vertical axis to place the woofers at the same height, one firing forward and one firing rearward. This is the principle on which I have been working. An added bonus to

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this vertical axis of rotation is that destructive interference occurs mildly to the outside of the speakers cutting down early reflections from wall surfaces much like an electrostatic dipole. My speaker system has four woofers per module, the two forward firing located further from the listener than the rearward firing. The speakers are built as mirrored pairs as shown in *Fig. 1*.

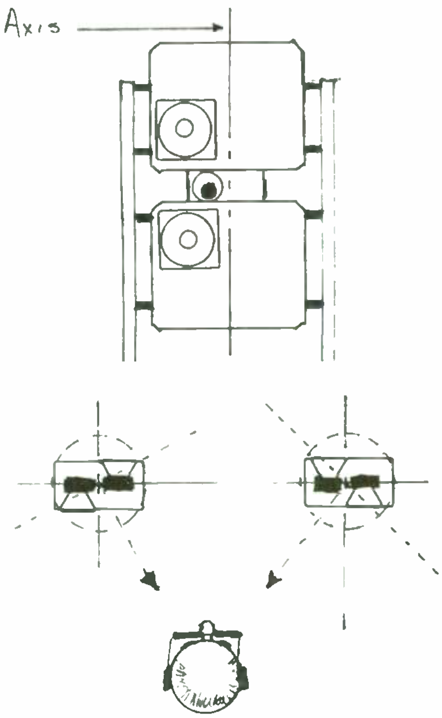


FIGURE 1: Speakers built as mirrored pairs. The left view of the speakers is shown on top, the top view is shown on the bottom.

In the center are two ¾" tweeters (also SEAS), rotated along the same axis. Be careful to position the tweeters as close as possible to prevent a "venetian blind" effect. I like to call this arrangement "The Ultimate Point Source" loudspeaker. It uses double-slit inferometry to produce a well-controlled, coherent dispersion pattern well away from the side walls, the ceiling, or the floor, and instead focuses front and rear. This eliminates at the same time the reverberant step as both speakers are forced to limit their energy to 90° about their normal. This coherency gives a high resolution to the final image formed by the left and right speakers. The double-slit phenomenon called stereo of the left and right speakers now has a highly coherent double-slit signal with which to start. This image produced is a true image not a B__e loudspeaker, or an artificial enhancement produced by a delay or inverting the opposite channel and passing it through a driver.

Finally, I thank Mr. Gonzalez for writing about his minimum-phase crossover. After reading it, I threw out the plans for my simple single-inductor, single-capacitor series crossover. I am still looking for a subwoofer for this system (crossover at 150Hz).

Michael T. Allen
Moreno Valley, CA 92388

Ralph Gonzalez replies:

My one criticism of the idea of modeling a dual-driver system as a dual-slit inferometer (other than the fact that I've never seen this term used in this context) is that this doesn't appear to take into account the filtering effect caused by enclosure diffraction. Rather than two identical point sources separated by a distance (as assumed in the dual-slit model), we have two sources with frequency-dependent directivity and different frequency responses. Namely, the rear driver's output on the listener axis begins falling above 500Hz, with an attendant phase shift in addition to that caused simply by the time delay.

For example, you needn't worry about a rear-mounted tweeter causing a "venetian blind" effect, since at high frequencies the forward contribution of the rear-mounted tweeter will be negligible.

For the above reasons, I didn't attempt to position the rear driver in a way that would "focus" the lobing pattern with the Delac S10. I was instead concerned with maintaining a narrow cabinet profile as described in the article.

Nonetheless, your design looks promising and ambitious, and I'm sure it works well.

IDEA DISCLAIMER

Scott Wolf's pipe and ribbon system (*SB* 3/91, p. 28) is quite interesting. I am a transmission line fan myself and can imagine his system's bass quality is great, not to speak of the ribbon's performance.

It is gratifying to know he is using the near-field listening arrangement. It is mesmerizing when used properly. I must disclaim its idea, since it has been in use by all those involved in close-range monitoring for years. Peter W. Mitchell wrote about it in a Boston newspaper and in a widely read audio magazine before I wrote about it here. It was Mr. Mitchell who presented a convincing case for this arrangement, as an exciting audiophile alternative.

Carlos Bauza
San Juan, Puerto Rico 00936-1220

HOW DO I...?

I have built the Swan IV speaker system using the Focal 5N412-0B5 and the D-28. In *SB* 1/91 (p. 91) in his answer to Mike Chin, Mr. D'Appolito states that the 29mm Scanspeak D2905 would be a good choice. I'm not an engineer, but I would like to know in reasonable layman's terms, diagrams, and so on, how to replace the D-28 with the D2905. Would this result in a more transparent sound? Are there any trade-offs?

David Smith
Bothell, WA 98041

Contributing Editor Joe D'Appolito replies:

I don't have a design worked out for the D2905 tweeter. I suggested its possible use, but it's up to the reader to carry out the design.

APPLYING DIFFRACTION LOSS

I have Ralph Gonzalez's LMP program and understand its operation. However, I am stymied as to how to apply diffraction loss as it pertains to my particular project. I have been working on a project similar to Peter Hillman's, although not nearly so large.

Mr. Gonzalez in his original three-part series in 1987 (*SB* 1/87, p. 18; 2/87, p. 42; 3/87, p. 38) spoke about limiting diffraction loss in the design of a speaker cabinet or taking it into account when designing the crossover. He mentioned that the effects of diffraction loss can be minimized by placing the drivers asymmetrically on the baffle surface so no driver sees an equal distance to the baffle edges.

The baffle of my cabinets is 21″ wide and 6′ high and no driver is equidistant from the baffle edge. The center lines of the drivers are 7″ from one edge and 14″ from the other. How do I figure diffraction loss into the mix of things in terms of the amount of loss and at what frequencies? Also, how do you arrive at these frequencies? I do not have the requisite equipment to start running frequency plots.

Herbert Meyers
Longmeadow, MA 01028

Ralph Gonzalez replies:

First, it is not true that asymmetrical driver placement will eliminate the usual 6dB "diffraction loss" that occurs at low frequencies where the wavelength is large compared to the enclosure dimensions. The best you can hope to accomplish by beveling the enclosure edges or mounting the drivers asymmetrically is to obtain a smooth 6dB loss at low frequencies.

LMP uses a simple model for diffraction loss, or what I've termed the "response step." The model is accurate for a spherical enclosure, but is still a reasonably good fit for a rectangular one, provided the drivers are mounted asymmetrically or the enclosure edges are beveled. It is permissible to position the drivers symmetrically so they are equidistant from the left and right enclosure edges, provided the listener is not positioned directly on-axis—the speaker should be toed in or out so it is not facing the listener directly (see "Diffraction—The True Story," Robert Kral, *SB* 1/80, p. 28).

If you do mount the drivers asymmetrically, it is best not to make the distance from one edge a multiple of the distance from the other edge as you have done. However, as long as you listen slightly off the main axis, you shouldn't have a problem.

As stated in my 1987 article, you generally can get a good LMP model of your driver by assuming a 6dB response step (be sure to enter a value of sensitivity that is 6dB *lower* than the specified value to account for this). The response step frequency you enter in LMP is that whose wavelength is equal to the enclosure width. For a 21″ wide baffle as in your case, calculate:

$$F = 13{,}500 / 21$$

Thus, you should enter 643Hz for the frequency of the response step in your LMP model.

As stated in my article, you may or may not see such a response step on the manufacturer's frequency response graph for your driver, depending on the measuring conditions. If the driver was measured in a large baffle (as is often the case for midranges and tweeters), the response step may take place at such a low frequency that the driver's response appears flat. However, when you use this driver in your own system with a smaller baffle, it will have a visible response step. On the other hand, many woofers are measured in small enclosures similar to those the speaker builder may use, and you will see a gentle 6dB rise in the manufacturer's curves.

Occasionally a driver's response will be tailored so the response step doesn't appear even though a small enclosure was used for the manufacturer's measurements. In such cases, you may try to make the LMP model match the manufacturer's measurements by reducing the height of the response step. This is why it is critical to know the measurement conditions. At least one company (SEAS) provides all the necessary information in their data sheets.

Once you have modeled your driver's response steps, you can attempt to eliminate the effect using the crossover. For example, place the woofer's low-pass filter a little lower than usual and pad the midrange or tweeters an extra 3-6dB. It's okay to leave about 3dB of the response step intact, in my opinion.

MYSTERY SYSTEM

I recently read a detective novel in which the victim is a cantankerous, elderly speaker designer who's invented a revolutionary new system and is killed in an aneochoic chamber. You find out whodunit in the end, but the exact nature of this superspeaker isn't revealed fully. The book is *The Dead Room* by Herbert Resnicow (Dodd, Mead & Co., New York, 1987).

Now, if this is a tip for recreational reading you didn't know about already, perhaps I can ask questions about a proposed system that holds some mysteries for me. I wish to build a set of satellites and subwoofers, and I have queries about both.

I am not an engineer, though I have built kits over the years: Dynakits and Heath, and to show my age, EICO and even an Allied Radio amp years ago. I have built speaker systems, the last being a three-way, with the crossover made by Madisound. I have built crossovers from schematics designed by others, but I do not have the knowledge to translate a more complex schematic into a nuts-and-bolts device, that is, into an electronic crossover.

I'd like to make the subs with two 10" VMPS woofers each, in a sixth-order box. In *The Loudspeaker Design Cookbook*, Vance Dickason mentions a *Speaker Builder* article by Robert Bullock in 1982 on the subject. He does not footnote the paper, however, so I cannot ask our library to get a copy for me. MIT doesn't get the magazine and the Boston College library doesn't have it back that far. Do you have a reprint? Or, if not, do you have the full reference?

Mr. Dicakson says an Old Colony filter KF-6 can be modified to make the things work. How explicit are the instructions in your article or those of Old Colony? I don't, without explicit, step-by-step guidance (or perhaps a stuffing guide, a photograph, or a circuit diagram) know how to connect a power supply to the device. If I use an Old Colony or Marchand crossover, can I use the same power supply for the crossover and the filter and put the whole mess in a single box?

The VMPS specs are f_S = 28Hz, V_{AS} = 3.2 ft³, and Q_{TS} = 0.31. I found formulas for a sixth-order box in R. Kaufman's *Enhanced Sound* (pp. 140–1): $V_B = 4.1V_{AS} \times Q_T^2$ and $f_B / f_3 = 0.3f_S / Q_T$. I get a V_B of 1.26 ft³ (doubled for two woofers) and a f_B = 26.5Hz. Does this agree with your design? How do I calculate the correct port diameter and length and tune the box? Am I correct in thinking a sixth-order involves deliberately misaligning (or mistuning) the system, which is then corrected by the filter?

Kaufman gives a design for a "bass enhancer" that can be modified for a sixth-order vented system. But, once again, he provides the schematic, and I, beginner as I am, need to know what to solder to what and how to lay out the wires and components.

Next, the satellites. A&S has a Polydax sale, and I'd like to use 6½" Bextrenes in a D'Appolito configuration, with the appropriate 18dB passive network. I wish to use Dynaudio D-28s as HFs. I have a choice: the MHD17B37T2C12 (f_S = 33, Q_T = 0.19, V_{AS} = 1.1, $48.50) or the MHD17B25R2C12 (f_S = 31, Q_T = 0.31, V_{AS} = 1.25, $32).

Both have cast frames and are 25% less the above prices. The B37 has a larger VC and 10W more power handling. The B25, however, has an SPL of 84 (versus 87). As the pair of speakers combined raise the SPL by 6dB, would the B25s be more appropriate?

Another consideration is that I'd like to start with the satellites and make the subs later, when I have the time/money. That means I'd like them to have the best low-end possible. With a QB_3 alignment and Mr. Dickason's tables, I get the following: B25R—V_B = 0.48 ft³ (times 2 for two drivers), f_3 = 46.7Hz. B37T—V_B = 0.125 ft³ (times 2), f_3 = 88.2Hz.

This would make the smaller VC B25R the preferable drivers *if* I have applied the *Cookbook*'s recommendations correctly. Is there, in fact, a way to get better low response from the larger VC speakers?

Another question applies to the whole business: subs, satellites, and filters. I don't have access to test equipment, or the training to use it. But, if I can get A&S to test the individual drivers, so that I have T/S precise figures for each, do I have a snowball's chance of getting the entire project to come out right? I can, if it would help, get a Radio Shack SPL meter and test CD to aid the tuning.

I have asked a lot of questions, which I hope give some idea of the stage of my amateur status. I hope I've asked the right ones and not omitted anything critical. Perhaps I could read other sources that would help me.

Niels Winther Braroe
Cambridge, MA 02139

Contributing Editor Robert M. Bullock replies:

The article you refer to is now part of a book called *Bullock on Boxes*. It is available from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458, (603) 924-6371, $10.95. The instructions for making the equalizer are straightforward and a stuffing guide is included. I have connected up to four boards to a power supply sold by Old Colony without any problems.

Your questions about calculating the correct port length and tuning the box are covered by other articles in the book. As far as the logic behind the sixth-order systems, it is not a matter of "mistuning" anything, but rather one of synthesizing a sixth-order instead of a fourth-order filter. The required design parameters are different.

Your questions about the particular designs you have in mind are best answered by using a program called BOXMODEL sold by Old Colony. You enter proposed parameters for your system and BOXMODEL produces the Thiele/Small response functions for the system. You can vary the system parameters in the program until you get the responses you like best. Then you can build the system using those parameter values. It probably won't be a tabulated alignment you wind up with, but they are not inherently better. It is simply that they are known to be well-behaved without having to examine their responses.

Once you get a system built, you usually have a long adjustment process if you wish to hit a tabulated alignment exactly. With BOXMODEL you can bypass this and check the performance of your system by putting its parameters into BOXMODEL. If you are satisfied with the responses, you are done. If not, you can vary parameters in BOXMODEL until you get something you like. Then go to the system and try to modify it to produce those parameters.

It is probably a good idea to have a copy of the book and the software, because the software assumes you are familiar with the jargon of Thiele/Small models.

ARIA QUESTIONS

I'm impressed with Joe D'Appolito's loudspeaker designs and I will be building an Aria 5 soon. I have a couple of questions regarding the Aria 5 and Aria 10 kits.

Would power handling at resonance or sound quality of the Aria 10 bandpass subwoofer be improved if the sealed cabinet section were aperiodically damped with a Dynaudio or Scanspeak variovent? Would this be a worthwhile mod to the Aria 10 subwoofer?

Is the Aria 5 max SPL limit due to the excursion limit of the dual midbass drivers or to the T90 tweeter? Could the max SPL be increased if crossed over to a sub at 150Hz?

Will Firstbrook
Delta, BC, Canada V4C 4J9

Contributing Editor Joe D'Appolito replies:

Aperiodic damping makes up for inadequate magnet strength to control the damping of closed-box systems around their system resonance. It also helps to control cone motion in these systems under large excursions. In the case of bandpass systems, like bass-reflex systems, cone damping is actually at a maximum around system resonance due to the optimized acoustic loading of the tuned front volume. In a properly designed bandpass system, aperiodic damping would be of little value and greatly decrease system efficiency.

BANDPASS EQUATIONS

Along the same lines as Abel Nemeth's letter in the 2/91 issue of *SB*, I decided to curve-fit the design data used for sealed/vented bandpass enclosures described in Jean Margerand's articles in *SB* 6/88 and

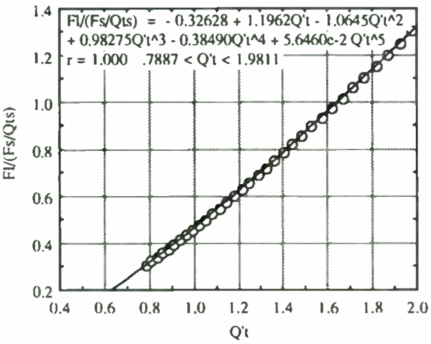


FIGURE 1: S = 0.4.

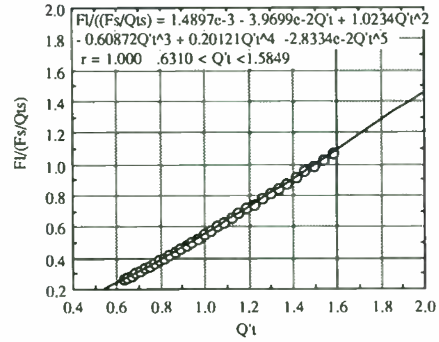


FIGURE 2: S = 0.5.

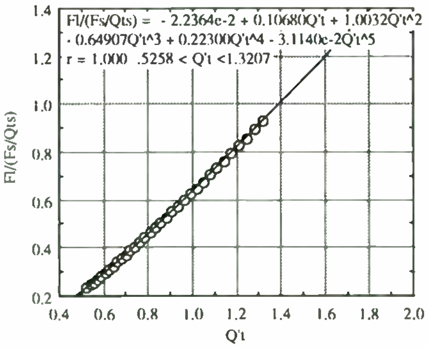


FIGURE 3: S = 0.6.

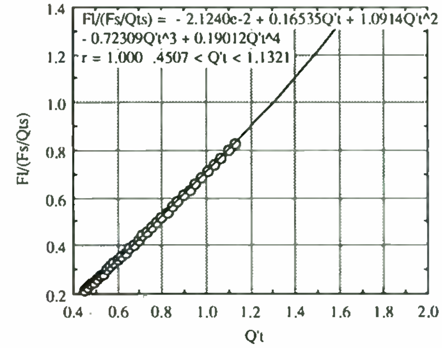


FIGURE 4: S = 0.7.

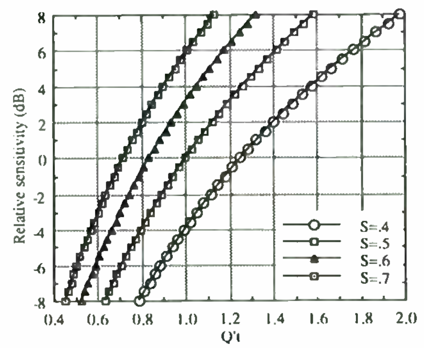


FIGURE 5: System efficiency relative versus Q'_T, S = 0.4 to S = 0.7.

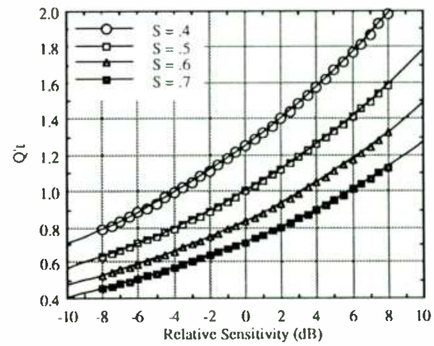


FIGURE 6: Q'_T versus relative system efficiency, S = 0.4 to S = 0.7.

1/89. They give most design equations needed, but only four design tables for the different S values (the damping factor of the system, $0 < S \leq 1$; design tables for S = 0.4, 0.5, 0.6, and 0.7). Different values of S will determine the amount of ripple in each design. In these design tables, Q'_T (the total suspension factor of the driver loaded by the back volume; *always* > Q_{TS}) is varied and f_L / (f_S / Q_{TS}), f_H / (f_S / Q_{TS}), and the relative system sensitivity are calculated.

The equations derived for calculating f_L (f_S / Q_{TS}) are fifth-order polynomials except for the S = 0.7 case, in which the fourth polynomial is used. Input is Q'_T, and the result is f_L / (f_S / Q_{TS}), from which f_H / (f_S / Q_{TS}) can be derived $(f_H$ / (f_S / Q_{TS}) minus f_L / (f_S / Q_{TS}) equals the values given in *Table 2*, p. 32). Multiplying these numbers by the ratio of f_S / Q_{TS} (determined by the driver used) yields the high and low −3dB points.

Figures 1-4 show the plots and equations derived. *Figures 5* and *6* show a family of curves from S = 0.4 to 0.7. *Figure 5* shows the relative sensitivity (dB)

S = 0.4, $0.7887 < Q'_T < 1.9811$
$dB = -3.8763 + 39.999 \log_{10}(Q'_T)$
$Q'_T = 1.25 \times 10^{(0.025dB)}$

S = 0.5, $0.6310 < Q'_T < 1.5849$
$dB = -4.9198e^{-5} + 40.001 \log_{10}(Q'_T)$
$Q'_T = 1.00 \times 10^{(0.025dB)}$

S = 0.6, $0.5258 < Q'_T < 1.3207$
$dB = 3.1673 + 40.000 \log_{10}(Q'_T)$
$Q'_T = 0.83334 \times 10^{(0.02499dB)}$

S = 0.7, $0.4507 < Q'_T < 1.1321$
$dB = 5.8454 + 40.000 \log_{10}(Q'_T)$
$Q'_T = 0.71427 \times 10^{(0.025dB)}$

FIGURE 7: Equations for relative sensitivity and Q'_T.

given Q'_T, while *Fig. 6* shows Q'_T given the relative sensitivity desired. The equations are shown in *Fig. 7*.

Obviously, the decibel equation and the Q'_T equation are the same, with different variables isolated. I gave both equations so you could find a desired Q'_T for a given decibel requirement, or vice versa.

The curve fits for all the equations conform well to the table data. I use the equations in a spreadsheet, and the equations for the first four figures show a maximum error of 0.003 in calculating f_L (f_S / Q_{TS}), at the extremes of the Q'_T range.

James Stratman
Carmel, IN 46032

LS DESIGN PROCESS

I'm intrigued by comments from Joe D'Appolito in *SB* 1/91 in response to a letter. He says the mathematics of designing asymmetric (double vented) bandpass subwoofers are well worked out, but complicated, and require computer iteration techniques. He also implies that computer design software is available to help in this process.

My perusal of *SB*; books by David Weems, Robert Bullock, and Vance Dickason; and software downloaded from the Madisound Bulletin Board provide no relevant mathematics or software. I have seen ads for commercial software (LEAP), which purport to be able to help with these types of designs. Unfortunately, LEAP and other commercial programs priced in the $200 range are beyond my budget.

Can Mr. D'Appolito provide me references to the mathematics required for such designs or to moderately priced software? If I can't find software within my budget, I'd like to try my hand at writing such software, or including it into the BASIC programs already written by Bullock, White, and others.

Roderick Prior
Farmington, ME 04938

Contributing Editor Joe D'Appolito replies:

I believe you have slightly misinterpreted my letter. Mathematical models for bandpass loudspeakers are available. If these equations are properly programmed, you can predict the response of a given driver in a given enclosure. This is called the analysis process. The design process is more complicated. You start with system specifications and try to find the driver and enclosure combination that meets those specs. Most of the software you mention can analyze a given bandpass configuration, but it cannot design to spec.

I am developing bandpass design software. With it, you will input the driver parameters you wish to use and specify upper and lower –3dB frequency points and the order of the bandpass design you wish to use. The program then generates an optimum bandpass design for that driver and the specified passband if such a design is possible. If not, the program indicates that the driver you have selected is not suitable for the design you requested.

Unlike many simplified Thiele/Small analysis packages, this program accurately models driver loading in the enclosure and enclosure losses. The program outputs a complete design, including front and rear volumes, tuning, port design, and the design of any auxiliary equalizer required. It also plots frequency response, max SPL, input power at max SPL, and input impedance. Multiple designs can be compared. Furthermore, this software has been integrated into a more general analysis and design package for conventional closed- and open-box systems with and without equalizers, so that all these designs may be made and compared.

I stress again this is a two-phase program. Phase one consists of automated and automatic design. The second phase is an analysis phase where you can modify the optimum design to accommodate other constraints. (I get to decide what the "optimum" design is.) For example, the optimum design may require an enclosure that is too large for your room. In phase two, you can examine the effect of reducing the enclosure volume.

The software is currently operational and under test. A few of my speaker builder friends are living with the software and noting deficiencies. When we are confident the program is correct and user friendly, it will be made available to *SB* readers at a reasonable price. The process of software validation is tedious and time consuming, but very necessary.

PORT PLACEMENT

I read with interest "The Diffusor Port" (*SB* 2/91, p. 45) as I have built the basic box in *Fig. 1* and cut out the front panel with the problem of port placement. I had considered a 4" I.D. port on the rear baffle until I read your article. With f_B 37-40Hz, will this present much "irregularity" in response? As a second option, I might squeeze two 3" I.D. ports on the front baffle either between the tweeter and mid or below the woofer. This places the ports close to the walls and woofer in the bottom position, and to the walls in the upper position.

With the dual 3" ports below the woofer on the front panel, the ducts will be about 1½" from the side and bottom and about 2½" from the woofer cone. Is that okay?

Finally, I could take ¾" by 3½" spruce and cut angles with a saber saw to make the ramps for the diffusor port, using 4" I.D. PCV schedule 40 pipe. If I do this and put a permanent bottom on the port, the cabinet height will be a little low. Perhaps I could use three spikes and adjust in the back one to give it some tilt.

If the rear port is okay for response, I assume I must move it up above my front/back brace. The 2-3" ports on front may give the most flexibility in placement if mutual coupling and proximity to enclosure walls aren't a problem. What is your opinion of the three port placement options (back, front, and diffusor) for my particular cabinet? It's holding me up in finishing the system.

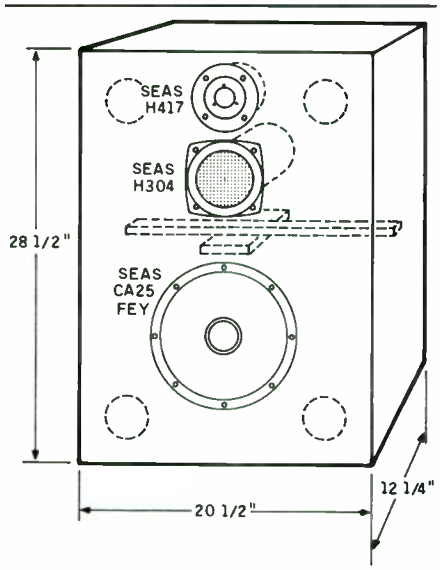


FIGURE 1: Mr. Ireson's port placements using a ¾" by 1½" side-to-side brace and a 1" by 4" front-to-rear brace. Approximately 3 ft.^{3}, f_B = 37–40Hz for SEAS CA25FEY woofer, Q_M = 1.9, Q_E = 0.35, V_{AS} = 135l, f_S = 33Hz, S_D = 350 cm.^{2}, X_{MAS} = 4mm.

Fred Ireson
Huntington, WV 25701

Contributing Editor G.R. Koonce replies:

Mr. Ireson has run into the problem all vented-box (or passive-radiator) builders face. Does the front panel have enough room for the port; if not, where does it go? The passive-radiator problem is more difficult than the vented-box port because the PR unit has a much larger diameter. Also, the problem can get acute for small vented enclosures where the port duct is sometimes longer than the enclosure is deep, unless you wish to reduce the duct diameter to that of a soda straw.

Mr. Ireson asks the following three questions:

1. How do you estimate any response anomaly effects of a rear-mounted port?

2. How close to the woofer can the port be located on the speaker board?

3. How close to the walls of the cabinet can the port duct be located?

I do not claim to be an expert on optimum port placement, but I will try to answer these questions from my experience and from what I have read.

Port not on the speaker panel.
The following discussion pertains to any time the port (or passive radiator) is located on any face of the cabinet other than the panel containing the woofer, or is on the same face but a large distance from the woofer. It thus pertains to rear ports, ports in the side of the enclosure, and the diffusor port in the cabinet bottom, which I like. In "The Diffusor Port," I showed that a rear-mounted port could affect the system response. I believe some

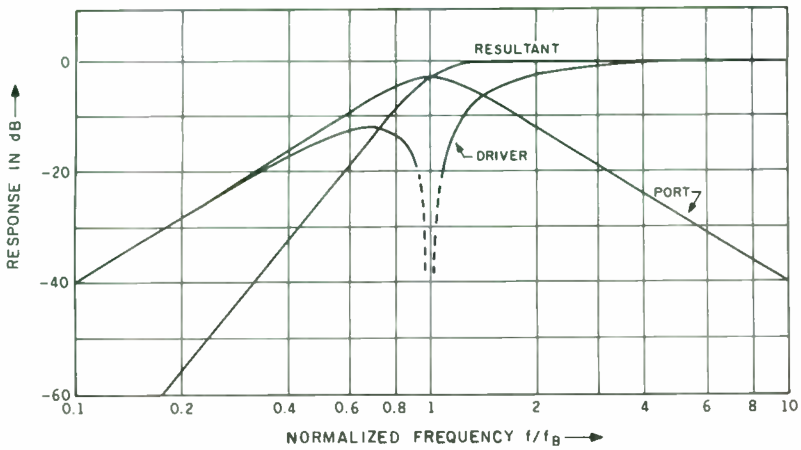


FIGURE 2: Driver, port, and resultant responses for lossless B4 alignment.

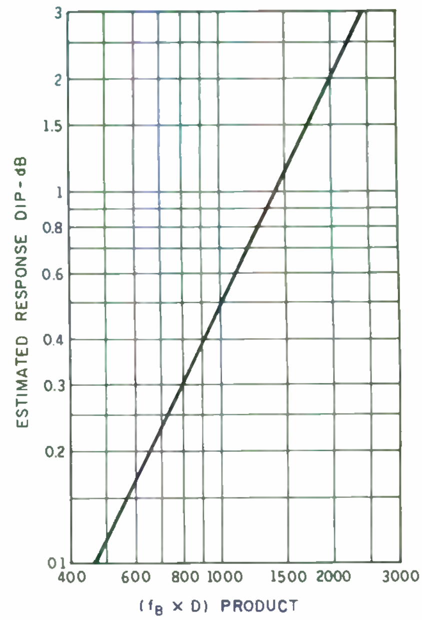


FIGURE 3: Estimated response dip versus (f_B × D) product.

readers became overcautious about the rear-mounted port based on this work. I will attempt to estimate the "irregularity" such a port can cause.

The port radiates major output over only a fairly narrow band about the box-tuned frequency, f_B. The frequency band to be considered is a function of the alignment used. *Figure 2*, from Benson,[1] shows the theoretical driver, port, and combined output for a lossless B4 vented-box alignment. Using this as a typical example, you see the port output dominates up to about 1.4 times f_B, at which point the contributions are equal and will be very close to "in phase" and thus adding directly.

If anything causes phase shift between the port and driver outputs, the combined response will dip somewhat. Such phase shift is introduced when the port output has a longer path to the listener than the driver output—when the port is not located fairly close to the speaker. You can estimate this phase shift from the increased path length and the speed of sound in air. Once you know the phase shift, you can estimate how much reduction will occur in the summation of the equal vector port and driver outputs and thus how much dip in response will occur.

I have done this for the conditions assumed above; *Fig. 3* shows the results. Establish the increased distance from port to listener versus driver to listener (D in inches), then multiply it by box-tuned frequency (f_B in hertz). You will find this product on the X axis and the curve will give an approximation of the response dip at 1.4 f_B. You can make many assumptions with this curve, so use it only as a guide.

In Mr. Ireson's case, a center-mounted rear port could force D up to about 22.5″. With f_B = 40Hz, this gives a product of 900, which *Fig. 3* indicates will give an approximate 0.4dB dip. This variation is insignificant compared to what room effects will do to the response. Remember, *Fig. 2* is an example based on a single ideal alignment, so don't try to use it to find a port location producing 0.1dB dip rather than 0.2dB.

This thus reinforces the comment in my article that the rear port exhibits a response problem only due to phase shift if f_B is rather high or the increased path distance, D, is rather large. Even so, I have never used a rear port. I do not keep the speakers I am building and worry about what acoustic environment the rear of the enclosure will see in the user's application. Will it be backed right up against the wall or even worse be built into the wall? It is this consideration rather than the time delay that forces me away from the rear port. When building for yourself, you can control this environment and the rear port becomes a viable option. Certainly many successful vented-box and passive-radiator systems have been built with rear-mounted vents or a PR unit.

Port located close to the woofer.
If the port and woofer are close together, you will get a mutual interaction. This effect was investigated by Small[2] with the following results:

1. A practical limit on woofer center-to-port center distance is about 1.5 times the woofer cone radius.
2. This placement can produce a mutual interaction of 2-8%.
3. Unless the woofer cone mass is very low, this amount of interaction did not have a significant effect.

4. With light coned woofers, you might see a slight alignment shift and a slight reduction in f_3.

Let's examine the case of a system using a 10″ diameter woofer and a 4″ diameter vent duct constructed of 0.25″ material (port O.D. is 4.5″). Such a woofer might have a piston diameter of about 8.2″ or a radius of 4.1″. If the driver and port are installed so they barely touch on the front panel, the center-to-center spacing is 7.25″ or about 1.77 times the piston diameter. With this combination, you cannot even get to the worst case examined by Small. In terms of mutual interaction, it appears that getting the port too close to the woofer is not a major problem.

I do see a practical problem with close woofer-to-port placement. I believe it makes the front panel too weak. To prevent the front panel from "talking" with vibrations from the woofer frame, I insist on the panel being stiffened on all sides of the woofer. You can do this by using a narrow cabinet structure. I usually tie the front to back of the cabinet just above the woofer and put a stiffener under the woofer. You shouldn't have this stiffener right against the woofer, but you usually can't get the port any closer than about 2-2.5″. I have never had any problem with this spacing. If you do use a port duct against a front-panel stiffener, be sure the duct is long enough to stick into the cabinet well past the stiffener height.

Port duct close to cabinet walls.
When standing waves are set up in your enclosure, the points of high pressure are the enclosure walls. If the port duct is located close to the enclosure walls, it may transport these undesired resonances out of the enclosure. The effect of port placement in this regard was indicated by doing scaling with The Listening Room program to show the situation inside the enclosure (*SB* 6/90, p. 63).

My own building experience has shown you shouldn't have the inside end of the port close to any wall of the enclosure. Along with the problem discussed above, the port's ability to handle the required air flow is sacrificed if it gets too close to the far wall (generally a small enclosure problem) and the port will always appear to be longer than it is and make tuning difficult if it runs close to the side wall or, even worse, close to a corner with the side wall and the top/bottom.

My own rule, which I am sometimes forced to violate, is that the inboard end of the port always be one port diameter from all the walls. So when using the diffusor port technique, you should make the actual duct length at least equal to the duct diameter. You must also be careful not to run into the driver magnet structure when the port duct comes in a face other than the speaker mounting board. If you plan to have an internal passive crossover, keep its placement in mind when you select the port location.

I have discussed the problem of finding room to locate the port duct for a vented box on the speaker board—a problem that gets worse with small enclosures or with passive-radiator systems. When the port is located on a face of the enclosure other than the speaker board, you must consider the effects of increased path length. This is only a problem with high f_B systems or large enclosure dimensions; a method to estimate the effect was introduced. You should consider the acoustic environment of the face holding the port in its placement. Speaker panel ports close to the driver are not a major problem in terms of interaction; the real limitation may be in maintaining a sufficiently stiff speaker board. Letting the inside end of the port duct get close to any cabinet wall can cause tuning and possibly performance problems and so you should avoid it.

REFERENCES

1. Benson, J.E., "Theory and Design of Loudspeaker Enclosures, Part 1—Electro-Acoustical Relations and Generalized Analysis," *A.W.A Technical Review*, Vol. 14, No. 1, p. 47 (1968).
2. Small, R.H., "Vented-Box Loudspeaker Systems, Part II: Large-Signal Analysis," *JAES*, Vol. 21, No. 6, p. 443 (July/August 1973).

IMPROVING A MULTIPLE-7 SYSTEM

Many readers have commented on the good sound of the Radio Shack Minimus 7 speakers, using the original 1022A woofer/mid driver, for the size and price. Other contributors have offered crossover alterations and "MTM" style configurations in an attempt to better its sound even more. For listeners having four of the speakers, I believe I have come across an elegantly simple scheme that will heighten overall enjoyment of their sound.

I think the 7s have a good sound when listened to within the nearer stereo sound field, with the left and right speakers rather closer together (perhaps 3′ apart) and the listening position on the central axis perhaps 3′ to front-center. When the units are used in normal listening positions, driven by substantial hi-fi systems, the sound falls apart and they almost always seem to be a small, rather thin-sounding system with a very forward character. This is typical of most small, limited range "mini-monitor" types.

That sound is probably due to many factors including the rising on-axis response step as well as the limited bass range, if not due to driver peaks and crossover problems. I have tried all schemes including placing the 7s against a wall, which worsens the hi-fi sound balance. I have also tried all manner of crossover alterations but with the drivers supplied, any really good components are not economically justified. This change costs nothing but a little time.

Let's consider a system using multiple bass drivers but only one tweeter (*Fig. 1*). For each stereo channel, take one 7 unit and remove the tweeter mounting screws, unplug and tape one of the tweeter wires, and reseal the box. What I did is simply to place the 7 with the disabled tweeter upside down so the mid/driver is on top. Place the other 7 on top of this, also upside down. Wire the units in parallel and fire them up. Of course, I was more elegant in my design and built a surrounding

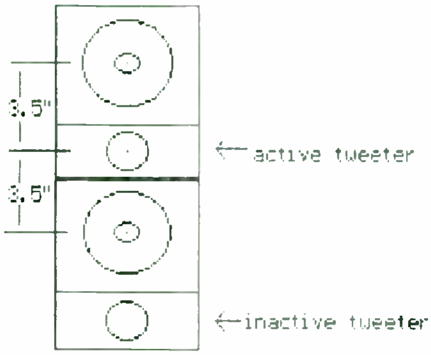


FIGURE 1: MTM array using two Minimus 7s.

box frame of stained pine to finish and hold the units. A natural progression would also be to remove the unused tweeter for other projects and fabricate a new cover plate of a hard material such as tempered masonite to seal the hole.

You will notice that the units are now configured into perfect MTM arrays with the two 4″ drivers close mounted to the one operational tweeter. The woofer/mid cones are about 7″ center-to-center and the interdriver spacing to the tweeter center is about 3.5″. Using the ⅔ wavelength rule for MTM arrays, this works out to a projected crossover frequency of near 3kHz, which is close to the 7's design, as close as you can get without box alterations.

To me, the system seems extremely well-balanced. Gone is much of the forward, thin sound of small systems and gained is a more robust, natural sound. Stereo spread and stage position accuracy are extremely good. It has no inter-tweeter comb filter effect due to the use of two active tweeters per channel. It seems that much of the thin, strident sound attributed to inter-driver fighting due to a poor crossover is eliminated.

Running flat, the system is respectable and with a bit of properly applied equalization, they begin to sing like $800 a pair systems (mine cost $120, on sale). I have a heavily equalized pair of large MTM arrays, 43″ high with 2.7 ft³ volume, two 8″ woofer/mids, and a D28AF tweeter. When the same equalized input of my larger system is wired into the 7 MTMs, they make your hair stand on end for their size in relation to the unmodified units. The systems are nearly identical in sound with "only the last low octave of bass missing." Oh, what a difference that last low octave can make on the impact of realism, however.

Should we scorn a little 4″ engine that is really trying? I think not. This idea might take only ten minutes to try and no changes are irreversible so give it a shot. I believe you will have new respect for your mighty 7s.

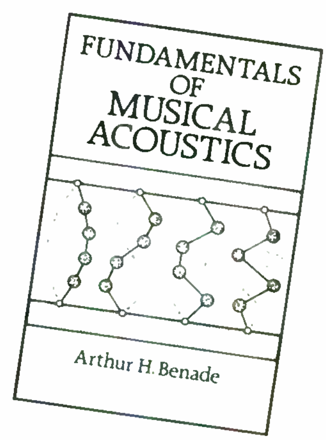
Frank B. Horner
Spring Lake Heights, NJ 07762

APERIODIC DAMPING

A short time ago, I finished building my first satellite and subwoofer loudspeaker system. As a first-time builder, I'm quite pleased with the results, but not being close to any hi-end audio shops for comparison, it's difficult to tabulate its character flaws if any.

I'm using Dynaudio drivers in a two-way sealed system using a D28AF tweeter and a 24W75 driver with a Zennon 2 crossover, which I believe is a first-order crossing over at 2.5kHz recommended by Dynaudio and Madisound where I bought the raw materials. Dynaudio recommends a volume of between 20–28 liters for vented, sealed, and aperiodic damped cabinets. The loudspeaker cabinets were built with ¾″ HDF and are 1 ft³ or 28 liters inside. Extensive bracing and lining the entire cabinet with oil-base plasticlay, however, reduced the internal volume to about 20 liters.

For sound absorption, I used cutting



wedge foam blocks, which I cut and installed in such a manner to create a matrix-type effect. The satellites are inert and the sound is flat, neutral, responsive, and transparent, and it images well. Highs are what I had hoped for—silky smooth. Both drivers seem to integrate seamlessly, but at times sound a bit thin overall. It's difficult to place my finger on why. They have an impressive, tight low end, but vocals don't sound as warm and full-bodied and yet they sound accurate.

Would using the Dynaudio variovent help in this area without compromising any of its other attributes? Dynaudio says a variovent will extend the low end, but I'm not so sure that's what I'm looking for since I'm using a subwoofer. Will a variovent add a little warmer character in the lower mid area?

Rocky D. Craig
Fort Dodge, IA 50501

Contributing Editor Joe D'Appolito replies:

I have no experience with the Dynaudio 24W75s or their aperiodic damping technique. My impression (perhaps unjust) has always been that aperiodic damping made up for inadequate magnet strength. Lack of warmth in vocals is usually associated with recessed response in the midbass region, say 200-500Hz. Aperiodic damping is most effective in the frequency range around system resonance, which is probably well below 100Hz for your system. Will aperiodic damping help your situation? Frankly, I'm not sure, but I can construct a scenario under which it could help.

You said the low end is impressive. It may be underdamped and actually too impressive. As exaggerated bass response could make the midbass seem recessed by comparison. If this is the case, aperiodic damping will reduce bass emphasis and thereby raise the apparent level of the midbass, producing the warmth on vocals you now lack. Give it a try. You can always plug the vent hole if it doesn't work.

HIGH-END AUDIO

For several years, I envied others for their manual skills. What I yearned for was the ability to build my own loudspeaker cabinets. No one manufactured what I wanted (or could afford). And yet, a virtual laughing stock, I realized I could not be trusted with sharp tools.

An avid reader of *SB*, I longed for the means to execute the ideas and incorporate the principles illustrated. A breakthrough came when Joe D'Appolito and Focal published plans for the Aria series. I had come to respect Joe's science and to admire his designs. This, then, was the one—the Aria 5.

But how could I cut and join the pieces? Worse, how could I rout and fit the face plate for the drivers? For the hundredth time, I read the article in *SB* 5/88 (p. 34) on the plans and directions for the Swan IV satellites, which seem closely related to the Arias—both of D'Appolito parentage. But, heck, I can't saw a straight line, don't have a table saw, and lack the hand/eye coordination to operate a cookie cutter. It seemed I was forever destined to dream and buy rather than plan and build. And no one offered the cabinets I so fervently wanted.

My paradigm was an affordable system similar to Dave Wilson's Watt design, smallish, asymmetrical, rock-solid—absolutely nonresonant—and with the best available drivers and crossovers. In short, an idealized Aria 5. Oh, I knew that Whale Cove, Watters, Madisound, Audio Concepts, and others offer Aria 5 kits, but somehow these missed the mark for me. And then in *SB* 1/91 (p. 10) I read that Elliot Zalayet of Zalytron Industries was providing boxes and components for high school students. Shortly after, I noted in advertising copy, the availability of proven boxes, including "optimum" models double thick in sandwiched materials having complimentary resonant indexes, fully radiused and covered in a choice of laminate. This foreclosed uncertainty. I was off and running. If those kids could assemble speaker systems with help from their instructor, so could I, with help from Elliot.

About this time, a friend in Ontario, Canada, sought advice on a stereo system. He is a master craftsman, machinist, boatbuilder, cabinetmaker, and home contractor, who can build anything. I had no hesitation in recommending the Aria 5 design. His was a resounding success. He

Continued on page 86

PHOTO 1: Tom Oldfield's Aria 5 speakers.

OLD COLONY SOUND LABORATORY

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

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

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

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

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

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

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

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

MICROPHONES: TECHNOLOGY AND TECHNIQUE — **BKB3**
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Beginning with a brief history of the relevant technology, this book then goes on to explain the basic theory of acoustics, electricity, and magnetism. The working principles and design of all types of microphones are explained in considerable detail, with examples of popular current models and descriptions of microphone accessories. The second half of the book provides guidelines on the creative balance techniques to be used for musical instruments, voices, and ensembles of all kinds, in both classical and pop music. Production methods are outlined both for studios and for on location, with notes on public address operations for live shows. Borwick is considered THE authority. United Kingdom, 1990, 241pp., 7 ½ × 9 ½, softbound.

THE ART OF ELECTRONICS — **BKC1**
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Completely up-to-date with the latest technology and standards, this second edition features completely rewritten chapters on microcomputers and microprocessors, digital electronics, and low-power and micro-power design (both analog and digital). Many new tables have been added, including ones for A/D and D/A converters, digital logic components, and low-power devices. The quintessential electronics text and reference. 1989, 1100pp., 7 ½ × 10 1¼, hardbound.

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

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

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

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ACOUSTICS — **BKAC5**
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This volume is an indispensable source of practical acoustical concept and theory for acoustical and electrical engineers, scientists, and consultants, with new information on microphones, loudspeakers and speaker enclosures, room acoustics, and acoustical applications of electromechanical circuit theory. 1954, 1986, 491pp., 6 × 9, softbound.

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

HEARING: ITS PSYCHOLOGY AND PHYSIOLOGY — **BKAC6**
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This multidisciplinary book leads readers from the fundamentals of the psychophysiology of hearing to a complete understanding of the anatomy and physiology of the ear, including the relationship between stimulus and sensation. 1938, 1983, 512pp., 5 ½ × 8 ½, softbound.

ELECTROACOUSTICS: THE ANALYSIS OF TRANSDUCTION AND ITS HISTORICAL BACKGROUND — **BKAC7**
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This volume provides a comprehensive analysis of the conceptual development of electroacoustics, including the origins of echo ranging, the crystal oscillator, the evolution of the dynamic loudspeaker, and electromechanical coupling. 1954, 1982, 260pp., 5 ½ × 8 ½, softbound.

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

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

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

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

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

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

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

Continued from page 83

opted for the new and improved Focal T90K tweeter with the most recent iteration of recommended crossover with shaping and compensation circuits optimized. A marvel.

Our project was intended to build on his success. We included in the contemplated system every logical idea ever presented, and some perhaps unnecessary ones, including four coats of Acoustical Magic borosilicate on the cabinet interior and on the driver baskets, a generous fillet of West 105/205 system filled epoxy resin in all the seams, and a division and separation of the high- and low-pass crossover complexes.

After consulting Rick Schultz of Electronic Visionary Systems, we elected to bi-wire, but not to feed audio through binding posts, instead to join conductors at the posts and route them through separate holes in the cabinets. Quantum I cables were covered with Monster boots, which sealed the holes and gave a finished appearance.

We used Audio Concepts' Acoustic Foam to line the Aria 5s—although both Joe D'Appolito and Jim Bock have suggested that convoluted mattress pads are effective at lesser cost.

We were careful to assure that the drivers fit flush with the cabinet face and had no gaps between the driver frames and the cabinet face—this stressed by Joe D'Appolito. This was no problem, as profiling had been done to perfection via a template in Zalytron's shop.

You can ask about drivers and crossovers. At Elliot's suggestion, we opted for the Accuton C11 tweeters and Focal 5K 013-L mid-woofers. Zalytron supplied the crossovers, separate high- and low-pass which, for the Accutons, were simple, no shaping or compensating networks being required. The crossover boards were mounted inside the cabinets at right angles to each other and as far apart as possible (*SB* 5/90, p. 26).

The results were magisterial. The excellence of timbre, range, dynamics, and dimensions of the soundstage are noteworthy and outstanding. Treble sounds soar with effortless ease and a total absence of stridency is apparent. I've often noted that most tweeters listened to in solo and up-close have a rather unpleasant sound. No more! The Accuton tweeter gives a sweet, as well as extended and accurate, range, even far off-axis.

Many speaker types emphasize soundstaging at the cost of "lobing," broadcasting sound straight ahead—a narrow window. This gives spectacular separation, a hyper-realistic presence quite unlike the original source, an effect analogous to earphones.

In literature, D'Appolito says, "No subwoofer is needed—you will be stunned!" I was. In truth, no supplemental bass *is* needed. All the sounds are there.

We had, however, a set of woofer enclosures for a Swan IV system, and somewhat reluctantly added these to the system using Joe Curcio's Pedal Coupler II active crossover. This combination gives music a bass foundation, a visceral sensation that can be felt as well as heard. You must experience this in Dorian's "Pictures at an Exhibition," a test to measure any speaker system.

The system gives an airiness to soprano voices (*Les Arts Florissant*— HM1901316), wherein individual soprano voices are clearly discernible from massed chorus. Also, violin overtones, barely audible, are now distinct in Vivaldi's *The Four Seasons* (EMI5X9557, the first few seconds of band 9). Once the domain of midrange sounds, the previously hidden poetry in the very high treble has opened doors to the high-end experience. With the Aria 5/Optimum double-thick cabinet/Accuton tweeters, music no longer sounds good—it sounds real.

Given the right music, performance, and engineering at the recording, reproduction excites my "goosepimple index" to a degree not previously experienced.

Highly recommended. We're satisfied. Focal, Accuton, D'Appolito, and Zalytron have a winner that anyone can assemble.

Tom Oldfield
Niagara Falls, NY 14304

Party Speaker

continued from page 41

This change will remove some of the lens which directs the output to the center of the room.

Whatever system I finally develop, the output will be efficiently spread throughout all three rooms.

REFERENCES

1. Robert Wiley was granted the patent I applied for one month after my patent searching. Although Mr. Wiley's lens is technically different from mine, the differences are too slight or subtle for a financially expedient appeal.
2. I exchanged this midrange driver for a MTX 236MTM44 4″, 4Ω midrange. The lens was originally designed around the MTX driver, but was not used in the speaker because I couldn't locate a MTX supplier. I finally purchased it through Crutchfield. The efficiency of this particular driver allowed me to remove all attenuation that was necessary to balance the speaker using the Polydax midrange.
3. Dickason, Vance, *The Loudspeaker Design Cookbook*, Old Colony Sound Lab.
4. Halliday, David and Robert Resnick, *Fundamentals of Physics*, John Wiley & Sons, Inc., New York.

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MIT AUDIO CABLES, Van den Hul, Cardas, MAS Masterlink, OCOS, Magnan, including Teflon insulated hookup wire for rewiring components; MIT capacitors; Solen polypropylenes to 200μF!, Wonder-Caps-solder-wire; HOLCO, Vishay & Resista resistors; Music posts & all types of audio connectors; RAM tubes; Tone Cones; NAVCOM products; Shallcross attenuators, many parts & accessories. $1 catalog ($3 overseas). ***MICHAEL PERCY***, Box 526, Inverness, CA 94937, (415) 669-7181. T6/91

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One Altec 409T 8" coaxial, mint, $25; Lafayette SK180 trihelix 10" three-way with brilliance control, no cabinets, mint, $60 pair; one 5" Philips 5061 coaxial, new, $10; Realistic APM-100 VU meters, mint, $20; two 17 ½ x 11 ½ x 25, professionally built oiled walnut cabinets, undrilled baffles, mint, $60 pair. (703) 578-4929.

IMF Mark IV professional monitors, four-way with transmission line, upgraded with stands, mint, $795. Bill (215) 481-6181 9 a.m.–4 p.m. EST.

dbx 118 stereo compressor/expander, mint; $50; two Audax TX 1125 RSN, modified with Tandem magnets (*Speaker Builder* tips 6/90) new, $45 pair. Matthew, (708) 260-1628.

Trade/Sell duplicates; tube equipment collection of amps through tuners and test equipment, over 275 pieces of equipment. Dennis Boyle, (214) 428-3901.

Two Gauss 4883B 18" extended bass woofers, never used, perfect $200 each. One new Dynaudio 30W54 in sealed box, $90. Andy, (516) 269-9796.

JBL 075 tweeter, 8;Q with N2400 crossover, mint condition. C.A. Howlett, 3 Hickory Nut Lane, Lake Wylie, SC 29710, (803) 831-2733.

AKG 240 headphones, $50; Azden DSR-12 headphone, $50; dbx 3bx, $100; dbx 120x, $100; Hafler preamp, $100; NEC AV-20 surround, $100; KEF vintage speakers, $200; Realistic STA 2200 receiver, $100; Pioneer CD-700 disc, $75; Infinity RS-8 speakers, $100; Marantz 4100, $100; B & O turntable, $50. Ed (305) 891-2267.

Target T-20, 9" speaker stands with upper and lower spikes, like new, $75 pair (original $175); Infinity RSA-250 car amp, new in box, $225; assorted woofers and tweeters. Write for list; Marrs modules, $12 each. Barry Kohan, 5024 Beckley Ave., Woodland Hills, CA 91364, (805) 375-2629.

One pair Borbely 388 linestages (revised per *The Audio Amateur* 3/90) built on 2.2 DA Teflon PCBs, POOGE'd with WBT solder, Resista & Holco Rs, $75 each; one pair Jung-Markel voltage regs per "Ask TAA 4/90", Rubycon & Panasonic HF caps, FR-4 PCBs, $50 each. Paul Kelly, Ft. Wayne, IN, (219) 485-4641.

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AudioQuest 30" Lapis/Cardas interconnects, $150 pair; Paradise VGA card, $80; Western Digital 20M hard card, $25; 287-6 co-processor, $50; Admiral (used) dual channel pulse generator, $25; all other items new or best offer. John, (203) 872-9210.

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Jordan 50mm modules. Tom Dausner, (516) 825-2038 after 7 p.m., or (516) 568-9356 anytime.

If you deal in Pro Sound products, please send me info on what you have. Mark A. Fiore, 1110 Wallum Lake Rd., Pascoag, RI 02859.

Can pay cash or trade tube amps. Mike, (708) 679-0900 ext. 295 days, or (312) 868-1696, evenings.

Information on Dynaudio Twynn or Focal Aria 7 for first-time builder. Happy with your system? Have recommendations? With your help I will do my first project right. Greg White, 10201 Viceroy Ct., "C," Cupertino, CA 95014, (408) 235-4753 days, (408) 257-9153 evenings.

Empire 598 turntables and parts, (703) 464-1100. Noon-6 p.m. EST.

Old Klipsch catalogs, JBL literature; *Electronics* magazine, September 1948; books; old books showing point contact transistors; old Hewlett-Packard and Tektronix catalogs; Throne of Merlin, (R.C. Schaller). D.R. Schaller, 6704 Schroeder Rd., Suite 6, Madison, WI 53711.

Technics SB 7070 speaker systems. Cabinet must be in good condition. Margherita, (904) 392-0622.

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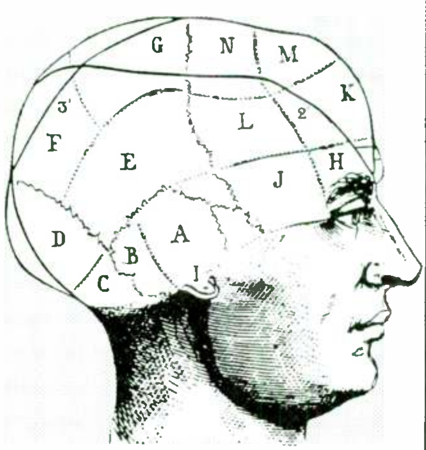
AUDIOPHILES IN THE DAYTON/SPRINGFIELD, OHIO AREA: We are forming an audio club. Please contact me if you're interested in construction, modifications, testing, recording or just plain listening to music. Ken Beers, 1756 Hilt Rd., Yellow Springs, OH 45387, (513) 767-1457.

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



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Advertising Index

MORAN *in the* MARKET

OH INCONSTANCY!

By David R. Moran

Everybody talks about constant directivity—well, okay, a lot of loudspeaker designers do—but does anyone really do anything about it?

I went on a roll last issue about what some goals might be for a loudspeaker's horizontal directional response (its radiation pattern) which would result in truly accurate imaging, much better than what we get today. By such imaging I mean the genuinely precise recovery and "laying out" in the space of your listening room what the mikes picked up during the recording session: placement, size, and internal relationships of the sources.

Horizontal directional response that would do this must almost surely be uniform with angle, that is, having the same shape and smoothness in all horizontal directions above, say, 200Hz, but varying in *level*. Listened to in mono over such a speaker, a piano or an announcer would not be wider in size and space in the lower midrange than in the treble, as is the case currently; it would be the original source width within all ranges.

This kind of uniformity is so tough to achieve most loudspeaker designers don't even try. First of all, the crossover seams must be unprecedentedly smooth, and worse, the range under 1kHz somehow must be made directional. Let me repeat a figure from last issue that depicts such goals for a horizontal radiation pattern (*Fig. 1*).

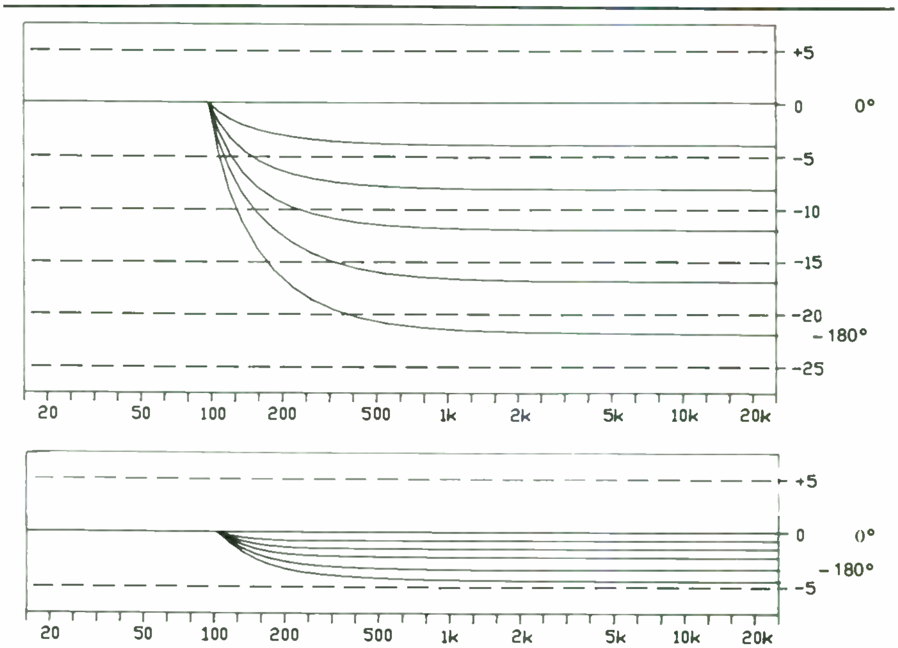

FIGURE 1: At the top, the spread curves, is the idealized horizontal directional response to achieve very tight uniform imaging, showing output from 0° (axis), then −30°, −60°, −90°, −135°, to −180° (output directly behind speaker). At the bottom, the bunched curves, is the same but for very spacious, more "equi-omni-like" imaging.

Now, it's easier to aim for the uniformity of *Fig. 2*, which is perhaps an acceptable interim goal. Many people call this constant directivity, and in fact, most ambitious modern designers say they do aim toward this—even claiming it's the ideal. Well, good thing, since virtually all drivers and loading designs get close to such behavior on their own. This is called turning reality into a feature. With most drivers, the coverage gets beamier as they produce higher and higher sound. As systems over the last decade have gone toward smaller woofers (6″ ones are now the norm worldwide, it seems, except in rack systems), however, natural directivity shows up in the treble alone; the midrange and below remain pretty equi-omni. These smaller woofers have predominated largely because of the widespread bogus notions about "fast" transient response; so much for understanding audio physics.

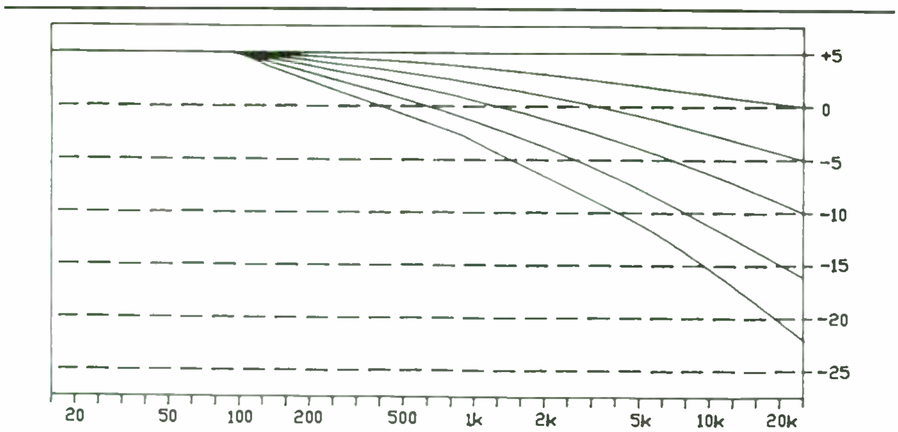

FIGURE 2: This is the time-honored goal of "increasing treble rolloff with increasing off-axis angle" horizontal directional response, aka constant directivity.

Lumps

As hard as anything else is smoothly joining the upper-range driver(s) with the lower one(s); for the most part it's lumpily done. See *Fig. 3*, the normalized (normalized meaning the axis response has been made ruler-flat) horizontal radiation, measured anechoically but with a floor, of the Cambridge Soundworks Ensemble. This system is well-regarded by many, with fair sound and tight imaging in some ranges due to unusual beaminess from the wide-

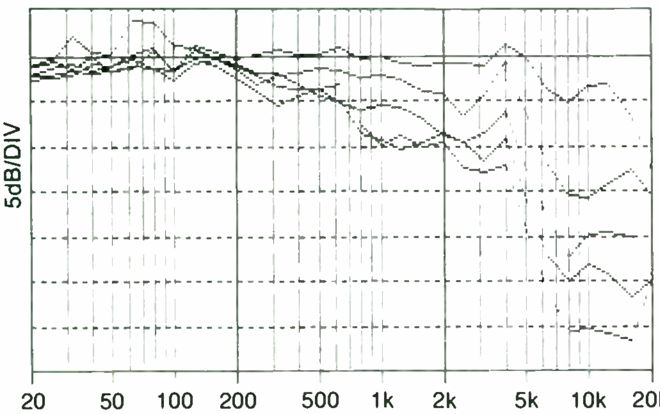

FIGURE 3: The Cambridge SoundWorks Ensemble horizontal direction response, normalized (meaning the axis response is equalized to be ruler-flat). Note the equi-omni behavior of the 4″ midrange at −30°, and above it the system remains as loud to 5kHz as on-axis. More important, note the tweeter flare-in beginning at 3kHz (its 4kHz output at −60° is as loud as its axis response). Finally, above 4kHz observe the pronounced rolloff due to tweeter diameter width.

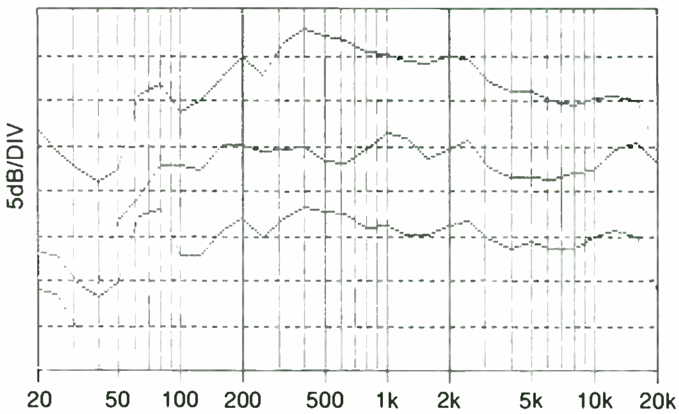

FIGURE 4: KEF 102/2 room responses. At top is the curve for the speakers on stands well out in the room; in the middle they are on the floor; and at bottom they are back on stands but with severe tone-control correction (bass and treble turned up). Response below 40Hz is to be ignored; see text.

diameter tweeter. But note the large tweeter flare-in starting around 3kHz, even though the lower driver is not 6″. (Like many two-ways, the Ensemble satellite also has a good-sized lobe vertically, centered on 2.5kHz, which means you might get better imaging if you turn it sideways, tweeters inside.)

To make any crossover less lumpy, you must curtail this flare-in by the upper driver, at the point where it begins its output as you blend it in. To get everything smooth below flare-in, you must do what only a few companies have even made the effort to do, to achieve directivity where it's hardest: in the lower midrange, the two octaves below 1kHz down to, say, 250Hz (refer again to *Fig. 1*).

Listen Anywhere—or Not

Most efforts so far to achieve midrange-and-below directivity have been part of a Stereo Everywhere/Listen Anywhere, wide-angle-imaging effort, although uniform, broadband directivity from 200Hz or so on up needn't be aimed only at that lofty, hard-to-hit target. dbx succeeded for the most part in the original, phased-array Soundfield designs, as did dbx sub-brand ADC in an aborted duplication attempt after the destruction of the dbx brands by takeover yuppies. Allison Acoustics currently has authentic accomplishments in this area in the top models of their magnificent IC (Image Control) line. Altec Lansing pro, I believe it was, made its stab in a vast horn system released only in Japan. AR's Magic MGC speakers combined such attempted midrange directivity with delayed sound, to other purposes.

Numerous shaky claims in this regard too turn out merely to entail crossfiring the stereo tweeters in front of the sweet spot, with the midrange left to spray freely in its usual equi-omni fashion—Ohm, NHT, several B__e and Infinity models, and others. You can turn any pair of speakers slightly inward, an old idea worthy for other reasons, and with most you get better sound and imaging, again for other reasons, especially if you turn up the treble a little.

Some promising current rumors in the true midrange/lower-midrange-directivity area, unconfirmed by me thus far, concern work by B__e in a still-under-wraps video project; by NEAR in a still-under-wraps "cardiodlike" execution of radiation pattern involving two drivers; and, *most* intriguing, by Canon in its recent announcement of a reflecting/concave loading technology. Finally, THX licensees like Snell stress, in their surround speakers, cancellation to the side by the pairing of drivers back to back, as the THX spec appears to call for. There may be surprises when these systems' horizontal directional response is actually measured. You *can* get such sideways "nulls" down quite low from a ribbon on a big baffle, it seems—stay tuned next issue—but in the ear-sensitive treble and upper treble such designs I have measured turn out to be quite wide-dispersion, not beamy, and not uniformly directive.

Even Rolloff and the KEF Entry

Okay, let's forget for the moment about making the midrange coverage as directional as the tweeter coverage. As I've said, most designers make a virtue of necessity when they talk about horizontal radiation pattern: you want the frequency response to roll off with off-axis angle, they opine, and thank goodness it does so already. Of course even then many don't get it right, paying too little attention to the off-axis responses of each driver in the lower part of its range. These are the comparatively louder responses that create an extra-broad image when bounced off the walls of a room. The designer may have wrongheadedly aimed either for axis smoothness or for total power smoothness, missing the importance of the details of directivity in between those two outdated goals.

Enter KEF, quietly: KEF the accomplished, maybe a little vain, successful loudspeaker company from the UK, being bought by Polk even as I write. KEF is the company that, apparently through misguided notions about bass "phase shift," supplies its top-end, super-expensive consumer speaker systems with insane equalizers that, if I read the reviews right, give more than 20dB of boost just above dc! KEF is also the one that introduced a standard-setting "kangaroo" design in the early 1980s which became the reference loudspeaker for many reviewers—but couldn't be driven to healthy levels.

More to the point, KEF introduced some three years ago a speaker system comprising a conventional cone woofer and a conventional 1″ dome tweeter—but with the tweeter voice coil housed in the same chassis as the woofer's. Much is made of this Uni-Q design for its coincident driver placement, same source for all sound, and consequent proper "time-domain" behavior (actually, it's not notably better than conventional designs, oddly, according to reviews thus far). But fie to all those claims. To me, what is really cool about Uni-Q is its constant directivity!

Coaxial speakers have been with us for a long time and are a great idea on paper. Yet they've never caught on in consumer audio. With Uni-Q, KEF has rediscovered something important, and has executed it in superior fashion: very smooth stitching and blending at the woofer/tweeter crossover, with the latter "horn-loaded" by the former.

I got a chance to audition and measure the new KEF 102/2 at some length recently. It's a small bookshelf-sized speaker with one Uni-Q driver per hefty cabinet.

Continued on page 94

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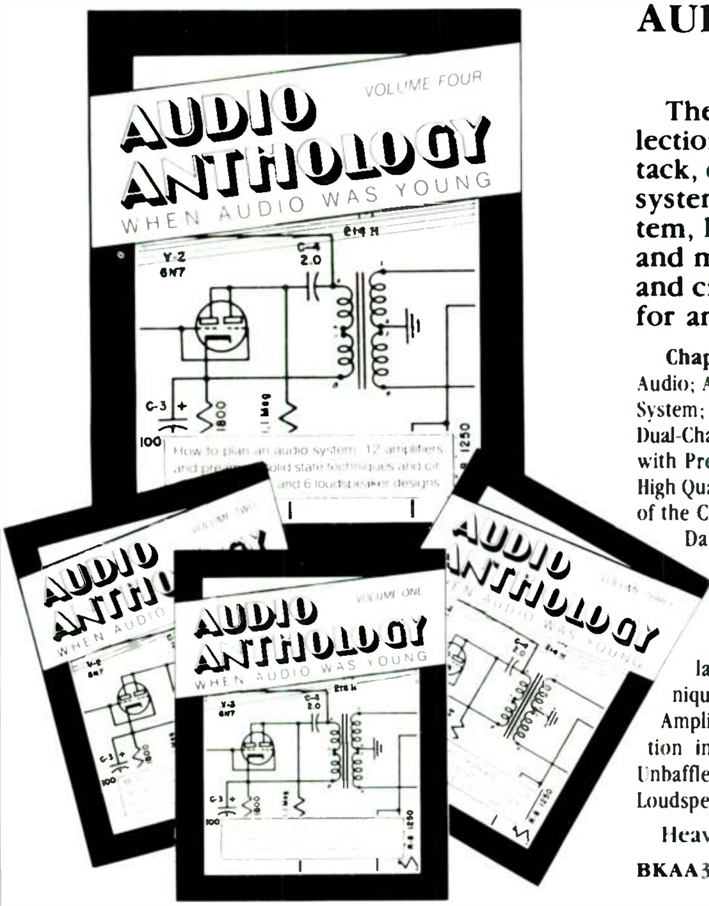

AUDIO ANTHOLOGY, Volume Four

Edited by C.G. McProud

The 34 articles from 1955–1957 in this fourth collection from *Audio Engineering* magazine take a new tack, emphasizing an overall view of the home sound system and offering guidance on how to plan a system, how to keep it simple, and how to understand and maintain the equipment. Solid state techniques and circuits are introduced here, along with designs for amplifiers and preamps and 6 loudspeakers.

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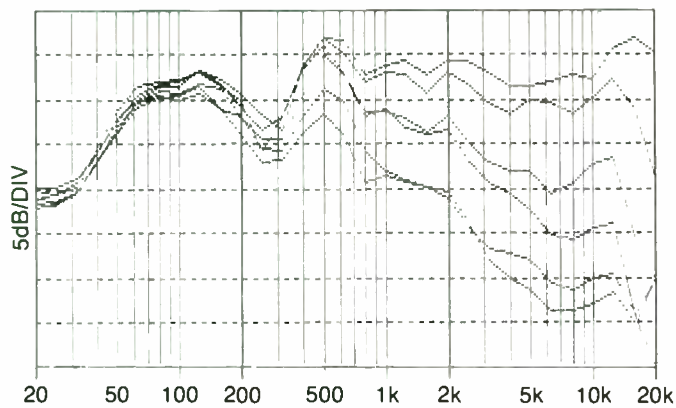

FIGURE 5a: KEF 102/2 horizontal directional response measured outdoors on a stand, from on-axis to directly behind.

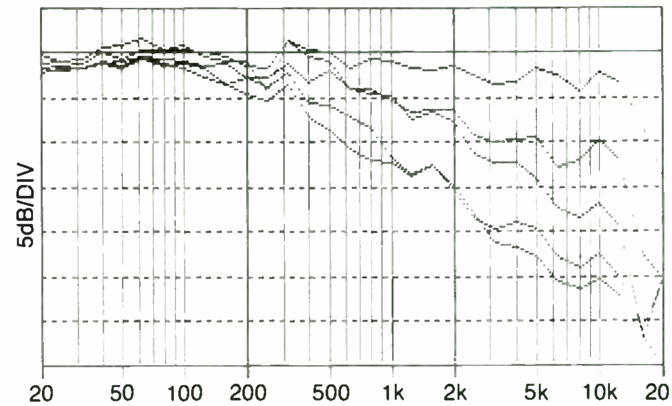

FIGURE 5b: The 102/2 curve set again has its axis response, at top of the group, normalized to perfect flatness.

Continued from page 92

US msrp is a non-inexpensive $1,200 for the pair. On stands, likely the company's preferred placement, the 102/2s did not sound great—thin in some locations, more full but still honky in others (nasal, chesty, cupped, hooded; choose your adjective), with airless and constricted highs, and never smoothly voiced overall. The manual is typically vague as to specifics of boundary loading.

An audiophile colleague where I work took them home and noted frequency response was "Rough, boxy, peaky, tedious to listen to." The top curve of *Fig. 4* shows reasons for our reaction, being a curve of stereo room response to pink noise (continuously averaged in time and space), with a broad hump centered on 400Hz, a most displeasing place to have such excess. In the middle of *Fig. 4* is the 102/2 on the floor but still out in the room, the lower range now augmented, the cabinet angled backward a little, much more listenable despite that sway-back above 2.5kHz. I suggest that wall placement will achieve something similar in overall tonal balance. At the bottom of *Fig. 4* the 102/2s are back on stands, now with strong tone-control adjustment: respectable, more like good-quality modern high-fidelity loudspeaker sound. (Ignore data below 40Hz; that's my furnace.)

However: in rock-steadiness, focus, stability, clarity of instrumental positions, these speakers imaged on the centerline like no stereo speak pair I have ever heard. In this regard the 102/2 performs as claimed, to my mind. And I don't believe its imaging has anything to do with "point source" talk or claims about time-of-arrival coherency. No, I think it's the Uni-Q design's unprecedentedly "constant" directivity.

Check It Out

Check out *Figs. 5* and *6*. *Figure 5a* is the 102/2 outside on a stand, with no front or side wall reflections to affect the floor-reflection notch at 300Hz. *Figure 5b* shows the top set normalized, so the below-axis curves here show the *difference* in output with angle. *Figure 6a* is the same as *Fig. 5a* except the 102/2 is up against a perfectly reflective front wall, which increases and alters its lower midrange.

So look: compare *Fig. 5b* and *Fig. 6b* with *Fig. 2*. Stick these KEF babies against or, preferably, in the wall; EQ liberally; get a Velodyne or equivalently superior subwoofer; and you've got something.

I see in these pages that SEAS now is marketing drivers with a similar design. (Uni-Q is probably unpatentable because there's so much "prior art" for coaxial.) Likewise Tannoy, whose products have for years been well-thought-of in the pro world, has an entire new consumer line of similar execution. I would conclude then that KEF has started something in the area I consider supremely important to good sound at home: smooth control of directivity. Oh, I say, old chaps, jolly good show!

REFERENCE

1. To review my measurement protocol: the compensated mike's output was continuously averaged for half a minute or more, 7-10′ from the speaker, 32-40″ from the floor (seated ear height), and at 0°, −30°, −60°, −90°, −135°, and −180° over ±15°. The speaker was fed precision pink noise from a flat-response low-output-impedance amplifier. The curves plotted here are ⅓-octave-averaged via an instrumentation-grade real-time analyzer.

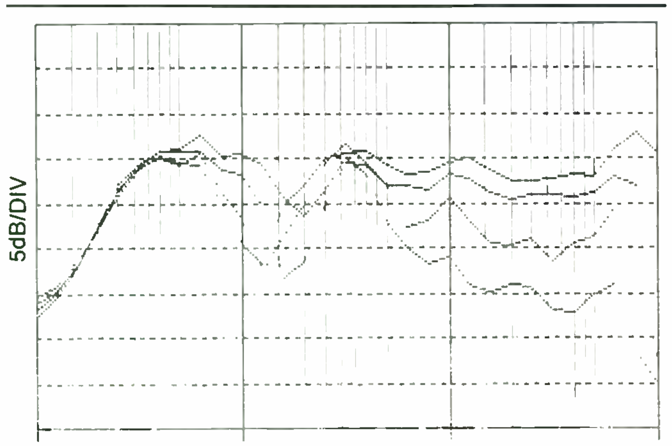

FIGURE 6a: Same as *Fig. 5a* but with the KEF 102/2 up against a simulated perfect wall. This placement helps fill in the floor-bounce notch somewhat.

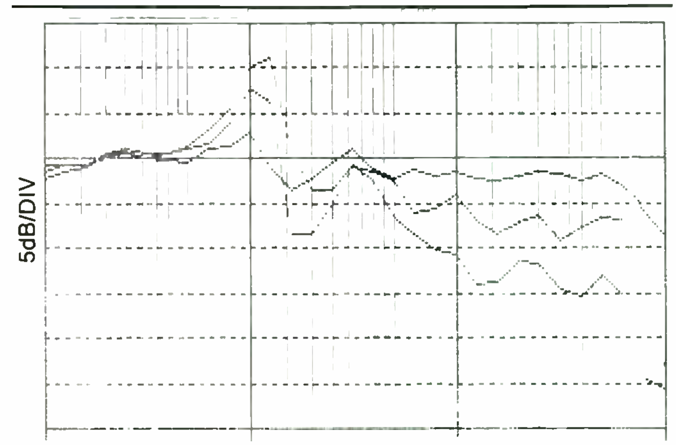

FIGURE 6b: The curve set normalized to perfect flatness. Note the resultant loud sideways lobes and image-spreading at 200–250Hz, which can be a common occurrence with wall siting.

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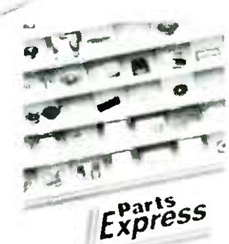

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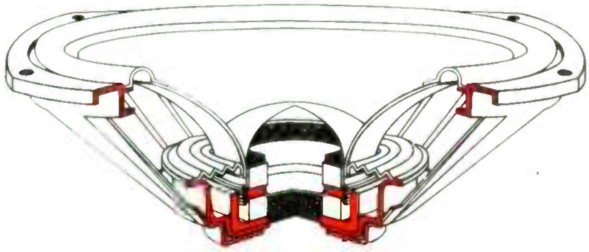

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Frequency Response	20-12000 H
FS — Resonant Frequency	22 H
Sensitivity 1W/1M	90 d
Z — Nominal Impedance	8 ohm
RE — DC Resistance	6.4 ohm
LBM — Voice Coil Inductance @ 1 Khz	0.62 m
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HE — Magnetic Gap Height	6mm(0.25
Voice Coil Height	14.5mm(0.57
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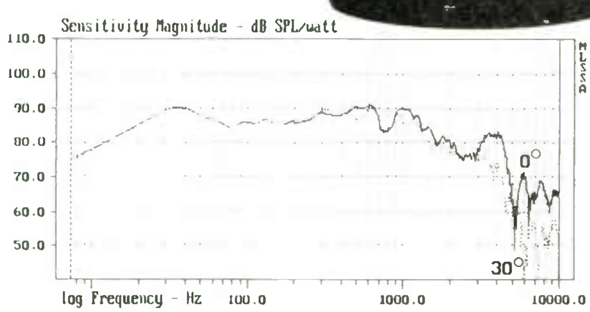

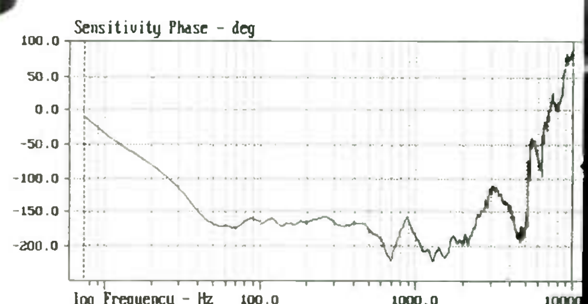

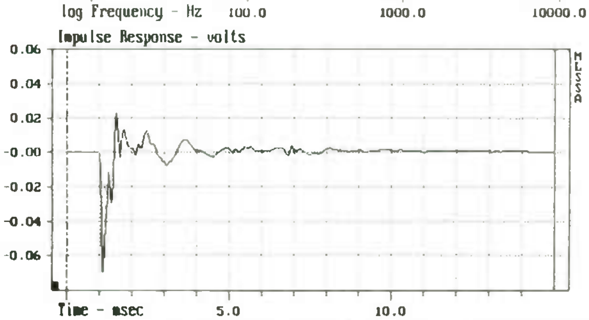

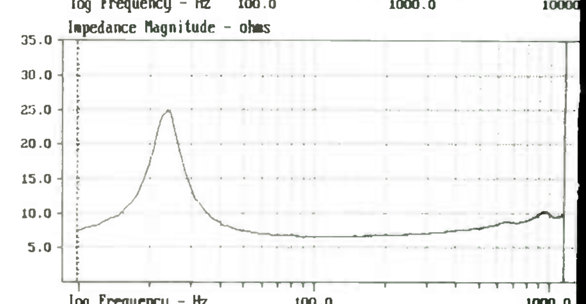

Morel operate a policy of continuous product design improvement, consequently specifications are subject to alteration without prior notice.

Fast Reply #IF1