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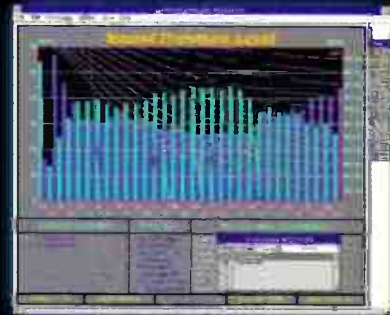

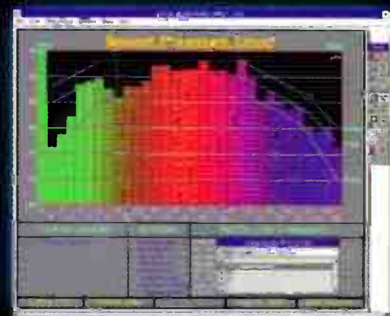

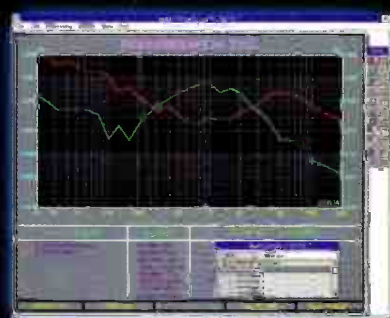

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Good News

■ HALL EXPANSION

Adam Hall USA announces the purchase of Fane Acoustics Ltd., a manufacturer and supplier of high-quality loudspeakers. Clive West, formerly of Precision Components and Turbosound, will focus on new product development and streamlining the company's existing product line. Jay Peabody, previously the Image Communications' General Manager, will expand its US operations. Adam Hall has facilities in England, Germany, and the US. Adam Hall USA, (847) 854-2492, FAX (847) 854-2507, AHALLUSA@aol.com.

■ HOW TO SCORE

AudioControl Industrial's HAL-30 built-in scoring computer enhances the SA-3050A real-time spectrum analyzer by measuring frequency response and SPL, with both USAC (United States Autosound Challenge) and IASCA scoring. Although the computer is factory installed in new analyzers, upgrades to include the HAL-30 in existing SA-3050As and free calibration are available direct from the factory. A Sierra White Special Edition SA-3050A, incorporating the HAL-100 scoring computer and a white front, is also currently available. AudioControl, 22410 70th Ave. W, Mountlake Terrace, WA 98043, (206) 775-8461, FAX (206) 778-3166.

Reader Service #108

➲ THIEL'S COHERENT SOURCE

Thiel Audio's latest Coherent Source® loudspeaker, the CS6, is a three-way system for home-music and video-sound use. In addition to a 12″ passive radiator, the CS6 features three drivers with metal diaphragms: a 1″ coaxially mounted dome tweeter, 4″ midrange driver, and 10″ woofer. Its cabinet includes a concrete baffle, internal bracing, cast magnesium driver chassis, and a real-wood finish. Using a first-order crossover, the CS6 achieves 4Ω nominal impedance and a ±1.5dB frequency response at 28Hz–18kHz. Thiel Audio, 1026 Nandino Blvd., Lexington, KY 40511, (606) 254-9427, FAX (606) 254-0075.

Reader Service #106

■ KNOCK ON WOOD

Cubicon Corporation, manufacturer of paper tube products, now offers veneered bases in small quantities directly to consumers. Cubicon's bases are made of a 3/8″-thick, nine-ply material, constructed of 100% recycled paper, and are available in more than 75 different geometric forms. The units range from 6″–32″ in diameter and 29½″–10′ in length to meet consumers' needs. Veneers offered include red oak, cherry, maple, walnut, mahogany, ash, white oak, unfinished wood, and metal or plastic laminate. Cubicon Corp., PO Box 28745, St. Louis, MO 63146-1245, (800) 635-3060, FAX (314) 567-0046.

Reader Service #102

■ A SUB WITH THE WORKS

Meniscus Audio Group presents the PSA-200 powered sub amplifier module. Designed for use with powered subwoofers and studio monitor designs, the PSA-200 incorporates line-level inputs and bi-polar outputs. This fully discrete module, capable of supplying 200W RMS into a 4Ω load, offers controls to adjust gain, phase, and continuous frequency. Meniscus Audio Group, 2575 28th St. SW, Wyoming, MI 49509, (616) 534-9121, FAX (616) 534-7676, meniscus@iserv.net.

Reader Service #101

➲ ROCK SOLID'S SOLUTIONS

A division of B&W Loudspeakers, Rock Solid Sounds introduces the Solid Solutions series. The S100 two-way mini-monitor includes a wide-range 4″ mid/bass unit, 1″ dome tweeter, and 100W peak power handling in a fourth-order vented-box enclosure. The C100 center channel combines a pair of 4″ mid/woofers and 1″ dome tweeter in a magnetically shielded horizontal cabinet with a diffraction-free front baffle. The PB100 active subwoofer features a single 10″ driver, on-board 70W amplifier utilizing MOSFET outputs, low- and high-level inputs, and built-in electronic crossover. Rock Solid Sounds, 54 Concord St., North Reading, MA 01864, (800) 370-3740, FAX (508) 664-4109.

Reader Service #104

■ HUH?

Hearing Education and Awareness for Rockers (H.E.A.R.) is a non-profit health organization devoted to raising awareness about hearing loss due to exposure to loud music. To raise funds and educate the public to the dangers of excessively loud sound levels, H.E.A.R. launched an independent record label. H.E.A.R.'s albums feature songs and public service announcements by a variety of artists. Aided by individual donations and corporate affiliate sponsorships, H.E.A.R. operates seven non-profit hearing centers and offers a Hearnet website, interactive access to health professionals, and information on tinnitus, hearing protection, and assistive listening devices (ALDs). Hearing Education and Awareness for Rockers, PO Box 460847, San Francisco, CA 94146, 24-hour hotline (415) 773-9590, FAX (415) 476-7113, hear@hearnet. com, http://www. hearnet.com.

← *Reader Service #36*

Good News

LEFT, RIGHT, AND CENTER

The S-75 series, new from Miller & Kreisel Sound, consists of bookshelf, center, and LCR (left/center/right) speakers. A multi-channel digital monitor, the Bookshelf-75 can be converted from a sealed box to a vented box with the removal of a foam plug from its front baffle. The Center-75 center-channel speaker incorporates M&K's phase-focused crossover and ferromagnetic shielding system. The LCR-75 monitor, although based on the Center-75, adds a satellite tweeter to the list of features. Miller & Kreisel Sound Corp., 10391 Jefferson Blvd., Culver City, CA 90232, (310) 204-2854, FAX (310) 202-8782.

Reader Service #103

RIDE A NEW WAVE

Now Hear This, celebrating its ten-year history, introduces its NewWave expandable subwoofer/satellite system. Available in three separate packages, NewWave offers an 8″ long-throw subwoofer in an 11″ square cabinet, a high-pass filter, and built-in 50W amp. Each of the three satellite speakers is housed in an ABS resin cabinet and includes a 3½″ midrange, ¾″ soft-dome tweeter, and magnetic shielding. Now Hear This, 535 Getty Ct., Benicia, CA 94510, (800) NHT-9993, FAX (707) 747-1252, website http://www.nhthifi.com.

Reader Service #107

SONIC ABSORPTION

Tekna Sonic's in-wall Vibration Absorbers, available in TF-10 and TF-5 versions, are designed to reduce box coloration in speakers. Intended as OEM versions for loudspeaker manufacturers and custom home installers, these vibration-control systems are utilized by NEAR's Model 50Me and Model 15M speakers. These devices consist of elasticized polymer plates, responding to frequencies of 40Hz–2kHz, which convert enclosure energy to heat to reduce sound distortion from panel vibrations. Tekna Sonic, Inc., 442 Houser St., Ste. E, Cotati, CA 94931, (707) 794-1512, FAX (707) 794-1513.

Reader Service #105

WIRELESS SOUND

Sennheiser released the RS 8 and RS 6 wireless stereo headphone systems, operating in the 900MHz bandwidth and three switchable frequencies. Powered by a BA 150 NiCad rechargeable battery, the RS 8 includes an HDR 8 stereo headphone with radio-frequency receiver, a T 8 transmitter with two battery recharging ports, and an adjustable headband and volume control. Also powered by the BA 150, the RS 6 offers a similar HDR 6 stereo headphone and T 6 transmitter, a fixed antenna, and automatic on/off switch. Sennheiser, 6 Vista Dr., PO Box 987, Old Lyme, CT 06371, (860) 434-9190, FAX (860) 434-1759, website http://www.sennheiserusa.com.

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Reader Service #35 →


D6G
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Resonance Frequency(Fs):47Hz
Sensitivity(E):85dB
Voice Coil Diameter:76 mm
DC Resistance(Re):4.5Ω
Voice coil Length:14.5mm
Force Factor(BL):6.5N/A
Linear Excursion(Xmax):5.5mm
SUSPENSIONCOMPLIANCE:(Cms):690uM/N
Mechanical Q Factor(Qms):5.3
Electrical Q Factor(Qes):0.54
Total Q Factor(Qts):0.49
Moving Mass(Mms):16.3g
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JOHN STUART MILL

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About This Issue

Many audiophiles are familiar with the Mitey Mike loudspeaker tester. This top-quality mike just became more versatile, thanks to **Publio Morera**'s simple wand modification ("Mitey Mike Goes Mobile," p. 8). You'll achieve portability and protection for your precious cargo to ensure accurate measurements.

Don Jenkins continues to share the results of his driver-performance tests ("Dynamic Characteristics of Driver Impedance," p. 10). His study determines that there's more to impedance than a simple, direct measurement and calculation. For a meaningful value, you must consider other circuit characteristics as well.

Just because you may have exceeded your credit limit at your local hardware and lumber stores doesn't mean you need to produce inferior-looking speakers. **Ken Bird** offers a practical construction application using mostly precut lumber and off-the-shelf components, and without the benefit of a workshop full of power tools ("The Pre-Cut Speaker," p. 16).

At certain frequencies, wave distortion can be heard in conventional loudspeakers, which can wreak havoc with your favorite music performance. To determine how much of this distortion is audible, **Dennis Colin** presents his study beginning on p. 18 ("Waveform Phase Distortion").

Acoustic designer **Philip Newell** is much in demand these days building control rooms/studios across Europe (including Portugal's first Dolby Digital Surround film post-production studio). In "Monitors for Nonenvironment Rooms" (p. 22), he explains the special demands confronting today's designers and describes the axisymmetric horn and monitor system used in rooms built with the nonenvironment principle.

Otto Doering's restoration project of the classic Altec drivers is an inspiration. With a little modern knowledge and experience, careful work, and adherence to classic design, he demonstrates how an old-time wonder from 30 years ago rivals the best we have to offer today ("The Classic Altec 604 Reborn," p. 32).

Also in this issue, you'll discover a useful software program to help you design speaker crossover networks ("Software Review—NETCALC," p. 40).

Speaker Builder (US USSN 0199-7920) is published every six weeks (eight times a year), at $32 per year, $58 for two years; Canada add $8 per year; overseas rates $50 one year, $90 two years; by Audio Amateur Corporation, Edward T. Dell, Jr., President, at 305 Union Street, PO Box 494, Peterborough, NH 03458-0494. Periodicals postage paid at Peterborough, NH and an additional mailing office.

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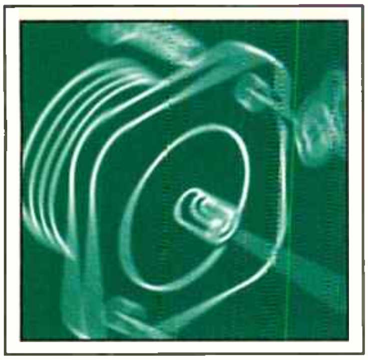

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MITEY MIKE GOES MOBILE

By Publio Morera

When I first began using Mitey Mike, my plan was only to measure the response of all my loudspeakers. These included a fairly large number of mildly successful, but also some very disastrous, modifications, creations, and repairs, as well as some hidden speakers that had escaped my wife's spring cleaning. It wasn't long before the sound of sine waves and warbles turned all three of our once affectionate cats into biting, scratching, and totally demonic beasts.

When my wife threatened to move out, I reluctantly agreed to temporarily "can" Mitey Mike. I was not having much success anyway at keeping the neighbors, their pets, cars, kids, and low flying planes from disturbing my outdoor measurements. Then I realized that most of my nonaudiophile friends and acquaintances also had loudspeakers. By presenting my interest as a noble attempt to assess the performance of their systems, I was assured an ample supply of speakers to measure, as well as an occasional free lunch.

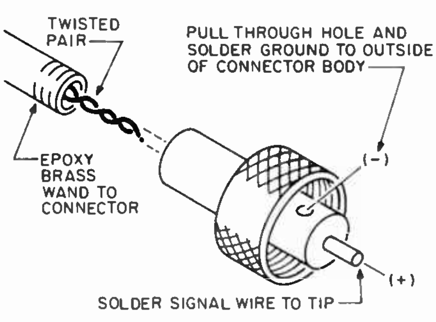

FIGURE 1: Wand modification.

HANDLING WITH CARE

The front of my delicate measurement apparatus has two ends, the sensitive capsule and the plastic enclosure housing the electronics, neither of which would react favorably to the rigors of traveling to friends' homes. To transport Mitey Mike, I needed a suitable and affordable carrying case. Unfortunately, most commercial instrument cases were priced much higher than Mitey Mike itself.

Measuring approximately 23″, the microphone was too long for the molded-plastic tool boxes readily available in hardware stores. I decided that the best way to reduce the effective length of the instrument was to make the wand assembly removable by mounting corresponding mating PL-type connectors on the case and the end of the wand.

The first step was to open up the plastic case housing the electronics, note the polarity of the wires and then unsolder them from the pc board. With the wires disconnected, I had to remove the brass nut on the end of the wand inside the case, along with the washers. I had intended to attach a PL-259 (Radio Shack part #278-205) by threading it onto the end of the brass rod, but the thread type and the diameter of the hole on the back of the connector did not permit this.

ADHESIVE SOLUTION

My alternative in order to use the PL-259 connector was simply to glue it in place using five-minute epoxy (*Fig. 1*). Before mixing the adhesive, I had to cut, strip, and tin the wires so I could solder them to the connector.

The microphone was constructed using the blue wire (supplied) as the ground. By sliding back the threaded connector collar, I was able to fish the blue wire through one of the holes on the connector and solder it to the connector's body. I then brought the yellow positive wire through the 1/8″-diameter hollow-center contact and soldered it to the tip. Once the electrical connections were made, I carefully pushed the excess wire lengths into the wand as I brought the PL-259 to rest against the one remaining brass nut.

I held the wand gently in a vise while applying the epoxy, and I protected the connector body from epoxy residue with a single turn of masking tape. With the capsule end facing upward, I applied sufficient epoxy to fill the small gap between the outer surface of the brass wand and the inner threads on the connector.

In order to ensure that the connector and wand were concentric, I had to hold the connector firmly against the remaining brass nut. After the glue set, I removed the tape and the slight residue left on the connector body. Any epoxy remaining on the connector would make it difficult to turn the threaded collar.

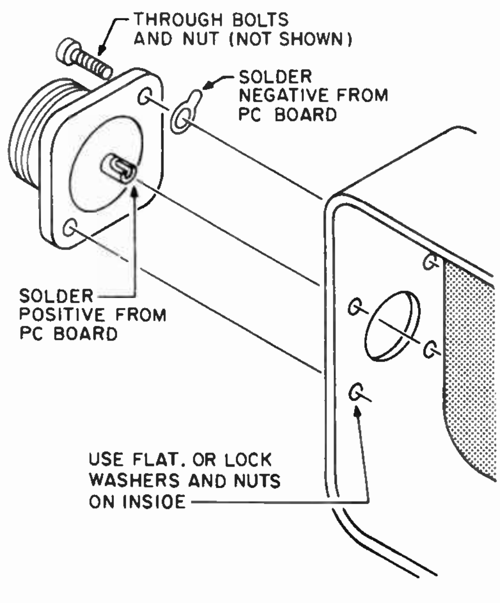

FIGURE 2: Connecting the wand to the case.

The complete wand modification took me only about 20 minutes. Because of the close fit of the wand into the connector, as well as the roughness of the threaded surface, the adhesion is quite strong.

Since the wand was intentionally grounded inside the case with the original mounting arrangement, I decided to make a small solder connection between the brass nut and the connector body using a 30W soldering iron. This allows the wand to remain grounded once it is mounted.

JOINING THE CASE AND WAND

The case requires an SO-239 (Radio Shack #278-201) chassis-mount connector in

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order to accept the modified wand and its connector. I had to enlarge the original mounting hole on the plastic case and drill four 1/8″ holes in order to mount the SO-239 (*Fig. 2*). The plastic case is quite soft, enabling me to make the screw and mounting holes with a reamer. I decided to mount the connector on the outside of the case so as to cover the rough edges of the enlarged hole.

Inside the case, the same polarity as the wand-mounted connector is observed. I soldered the yellow signal wire to the center terminal and the blue wire to a clean point of the connector body. I separated the blue ground wire from the large grounding lug that had previously been secured with the brass nut from inside the case, and substituted a small crimp-on lug. Once installed on one of the SO-239 mounting screws, this lug maintains the electrical continuity and so properly grounds the wand.

With these modifications, you'll achieve an ease of deployment and protection for your mike equipment—not to mention new-found freedom to roam and protection against angry wives.

DYNAMIC CHARACTERISTICS OF DRIVER IMPEDANCE

By Don Jenkins

My dictionary defines impedance as "the resistance met by an alternating electric current on passing through a conductor; the ratio of the effective electromotive force to the effective current of an alternating current circuit." This seems simple enough: divide the effective electromotive force by the effective current (which must also be alternating) and the result will be the impedance.

In *Electromagnetics*, Krause states that "the magnitude or modulus of the intrinsic impedance is equal to the ratio of the electric to the magnetic field, and the phase angle of the impedance is equal to θ." The function θ is described in a footnote: "The dot on the Z indicates explicitly that the impedance is also a complex quantity. However, it is a complex function only of the phase angle θ."

This seems somewhat more complicated than the dictionary version. Maybe you cannot simply divide the voltage by the current. It appears you must also consider the phase angle.

In calculating the impedance of a circuit, several considerations are necessary besides simply measuring the voltage and current and finding the quotient. You must also determine, at a minimum, the *effective* values of each, numbers that may not appear on your voltmeter or ammeter. Also, the current must be alternating.

By definition, the effective value is related to the power dissipated in the load. Since only resistance can dissipate power, the effective voltage and current values for AC circuits are relative to the power that would be dissipated by DC of the same magnitude. In other words, if an AC circuit is delivering 100W to a load, the product of the effective values of the current and voltage is, by definition, 100.

CALCULATING EFFECTIVE VALUES

The effective value of each is a time average of the instantaneous product of each value. This is the RMS value, which for constant results must have a duration of at least one complete power cycle. In the real world, the frequencies are always high enough so that many cycles are "read" by the instrument and RMS averaged. Can you then measure the voltage across a load and measure the current (using meters that claim to measure "true RMS values"), take their quotient, and declare this value to be the impedance? Unfortunately, the answer is both yes and no.

You can determine the impedance magnitude from these values, but to be complete, as required by the definition above, you also need the phase relationship, because the effective values are those that cause power to be dissipated in the resistive load. Using measured values, even from "true RMS" instruments, simply won't do the job. Without either knowing the phase relationship of the variables, or having some direct measurement of the power produced (heat, mechanical motion, etc.), you cannot determine the impedance of the circuit. It's as simple as that.

Usually, the direct measurement of power is difficult. Fortunately, you can use theoretical circuit analysis to good advantage. Unfortunately, like the definition of impedance itself, there is first-order analysis, and there is detailed analysis. The basic difference is in the assumption of pure sinusoidal currents in the circuit elements.

For the most part, all analysis treatments assume a pure sinusoidal current, but not

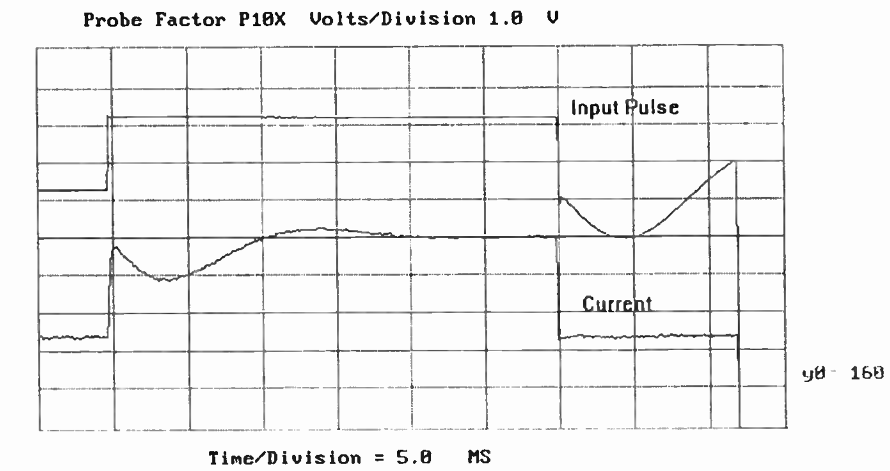

FIGURE 1a: 30ms 2V DC pulse, time record of 50ms.

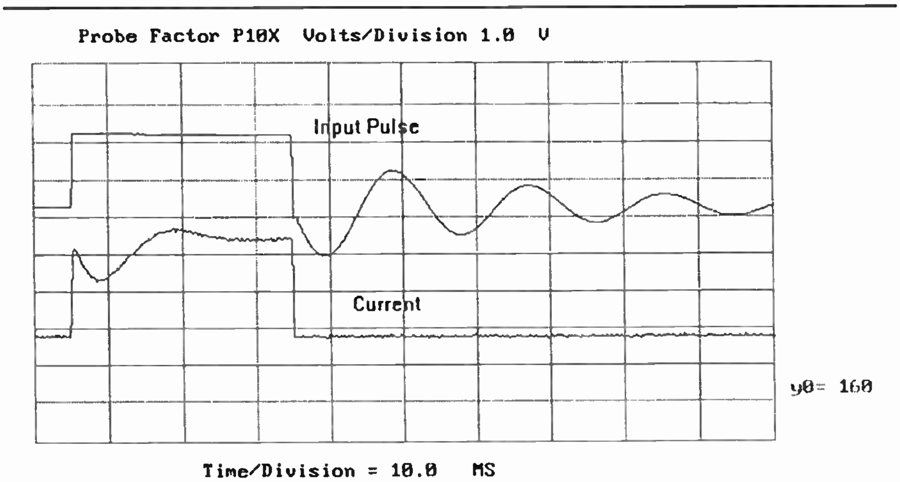

FIGURE 1b: 30ms 2V DC pulse, time record of 100ms.

necessarily a single-frequency one. Advanced analysis techniques can handle sinusoidal harmonics with phase shifts. but such analysis is complex and not for the average, or even advanced, experimenter. When transients are in the real circuit, this analysis is highly complex.

INPUT CHARACTERISTICS

It is clear that you must determine the input characteristics (phase and waveform) if the impedance value is to be meaningful. When the phase is input, the impedance becomes a complex variable, so that you cannot use impedance values as real numbers in calculations (real in the mathematical sense). That is, you cannot add, subtract, multiply, or divide impedance magnitudes without using complex-value methods.

You cannot put an impedance value of 10Ω in series with a 10Ω resistor and assume it will produce either a 20Ω resistance or a 20Ω impedance. Without additional data on the impedance and other circuit characteristics, the resultant effect on the circuit is simply indeterminate.

Impedance is also a point function. That is, the impedance of a circuit may vary with the input. (This is always true if the application requires mechanical motion.) Even with steady sinusoidal input, you can define the magnitude of the impedance as a set of instantaneous values. The value usually quoted for circuit analysis is a time-averaged value, based on the definition of impedance relative to power.

The question now is, what relationship does all this have to speaker-driver use and applications relative to power matching and filter design in the driver-to-amplifier circuit?

DRIVER INSTALLATION

First, consider the driver installation. Driver impedance curves are for the most part produced by a slow sine sweep over the frequency range of interest. You can then calculate the impedance as magnitude versus frequency, and—using the known voice-coil resistance—express it as a complex variable determined by the solution of the impedance triangle. Through this procedure, however, you have really determined only the steady-state characteristics using a single sinusoidal frequency input.

You have learned almost nothing about the transient impedance characteristics of the driver, and only slightly more about its impedance when subjected to a variable-frequency input. Practically all matching filters in crossover networks are based on this first-order, single-frequency impedance characteristic.

An understanding of the impedance characteristics of a driver can begin with a description of its various circuit and mechanical responses when pulsed with a constant direct current. *Figures 1a* and *1b* are plots of the voltage and current records of a 30ms, 2V DC pulse. *Figure 1a* is a time record of 50ms, while *1b* is the same record with a 100ms time record, which gives a "better view" of the current transient.

As you can see, while the voltage is rock solid at 2V input, the current is not constant. If the voice-coil resistance is constant over this time increment (30ms), why does the current vary? The answer is, of course, that the impedance—the resistance to current flow—is not constant. It is, in fact, changing with time as the driver mechanical components move.

Figure 2 is a computer simulation of the same driver. From these simulation outputs,

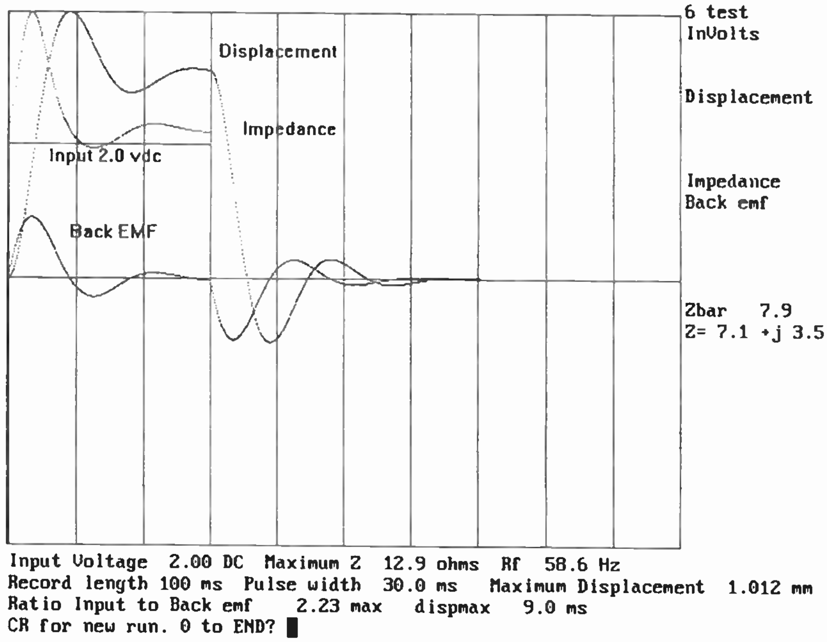

FIGURE 2: Computer-simulation outputs.

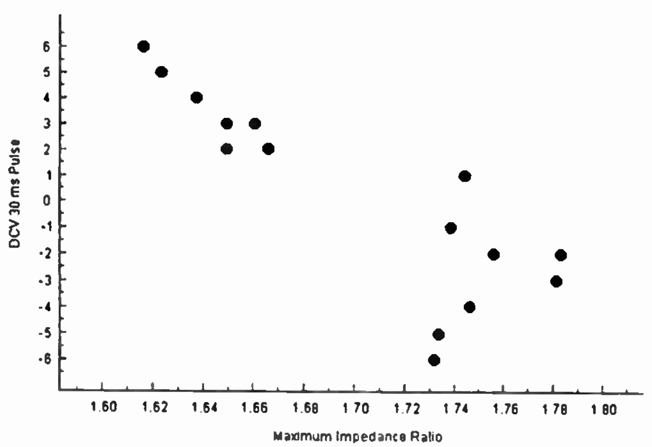

FIGURE 3a: 40ms 3V DC pulse.

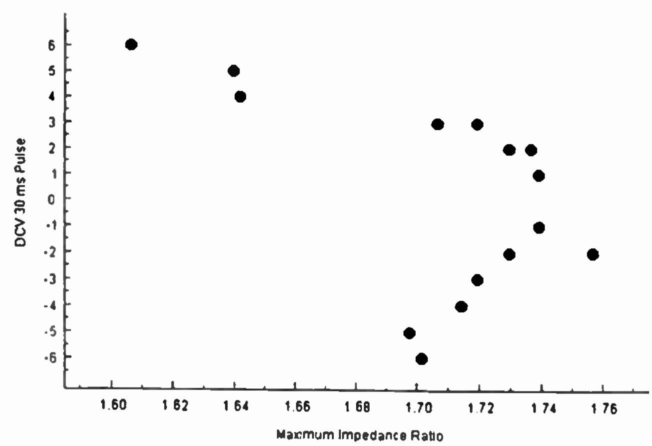

FIGURE 3b: 40ms 6V DC pulse.

you can see how the driver cone moves and the impedance changes with time. Also, a plot of the back EMF produced by the voice-coil motion through the driver magnetic field shows *why* the impedance changes.

BACK-EMF EFFECTS

As the input current accelerates the driver cone, a resistance to this acceleration is produced by the mechanical resistance of the cone (compliance), the weight of the accelerated components (mass), and the back voltage generated by the velocity of the voice coil through the driver magnetic field. As the velocity increases, so does the back EMF, resulting in an apparent increase in voice-coil ohmic resistance as the back EMF reduces the input current.

(The back EMF is a voltage of opposite polarity in this example. However, as the driver motion and input voltage change with time, it is possible for these two components to add together and provide "negative impedance" over some parts of the cycle.)

In this example, the voice-coil resistance is 7.1Ω. The maximum impedance seen during this transient is 12.9Ω. The "integrated" impedance magnitude for the complete 30ms pulse is 7.9Ω, and the complex "integrated" impedance for this pulse is 7.1 + j 3.5Ω. Having measured and calculated these three different values, you can now examine what each one means, and bring to light the interpretation of impedance measurements and quoted values.

Voice-coil DC resistance is the actual value of the voice-coil winding's resistance. But this value is not the impedance of a driver; it is only the voice-coil resistance.

Impedance magnitudes given as so many ohms must be referenced to some frequency. This magnitude will always be higher than the voice-coil DC resistance—significantly so near the driver resonant frequency. This value should be quoted as a complex variable, just as the 7.1 + j 3.5 number above. This value indicates that at the stated frequency, the current is not in phase with the applied voltage. This value is classically given for a driver and input system in equilibrium, and must be so for correct interpretation. This means that the driver mechanical motion and the input voltage are not changing, relative to each other, over the measurement time.

MEASURING OVER POWER CYCLES

Since RMS impedance values are defined as power equivalents, the equilibrium measurement must be made over at least one power cycle. For most real-world cases, as noted above, this is always the case. Impedance measured in this manner, i.e., over a number of complete power cycles, is the integrated average of each point impedance value over the measurement time.

For RMS voltage magnitudes, a value of 100V RMS is, for a sine wave, a peak-to-peak voltage value of 289.92V. A 100Ω impedance magnitude, for a variable signal input, will have peak values always greater than the 100Ω RMS magnitude. You cannot determine power in the circuit from the quotient of the squared RMS voltage and the impedance magnitude.

For the nonequilibrium transient condition as shown by *Figs. 1* and *2*, the impedance is a time-variable value, and can be completely described only by the time record. While the integrated complex impedance value may have some utility for circuit analysis, the complete time history is the only true measurement of the impedance.

In summary, when the impedance of a driver is quoted, you must note additional measurement conditions for the impedance value to be meaningful. The complex

DC PULSE TEST

The DC pulse test, using the impedance transient, is useful as a driver-evaluation tool both for self-damping characteristics and for "balanced" assembly and design. A perfect driver would have a linear compliance over the maximum input power range, along with a perfectly linear magnetic field within the maximum power travel of the voice coil. The voice-coil suspension would also be exactly centered in the field, so that the acceleration force on the cone is at all times linear with the voice-coil current.

By using the DC pulse-impedance response of a specific driver, you can evaluate the sum of these design parameters. *Figures 4a* and *4b* are plots of pulse-impedance response for both positive and negative pulses. The first is for the driver "right side up," and the other for the driver "upside down." The axis of cone motion is always vertical in these tests.

The reason for the two cases is that for a driver having a large accelerated mass, gravity will have some effect on the results. By running the test both ways, this effect, if any, is eliminated.

Using this test with the motion axis horizontal, which is the way most drivers are installed, puts an imbalance on the cone suspension that test orientation cannot eliminate. For most drivers, any position will yield the same results. The two plots shown here are in fact statistically the same. Only when the suspended mass is large will there be much measurable difference.

Looking at *Fig. 4a*, the impedance ratio (IR) decreases with greater positive-amplitude pulses. A decrease in IR indicates a lower maximum cone velocity, i.e., the maximum velocity does not increase linearly with current. This can be caused by increasing suspension compliance with displacement, the usual case, or by a reduction of the magnetic-field strength with displacement.

From this test alone, you cannot separate the cause. Looking at the negative pulse series, the IR values are much more vertical. For an ideal driver, all points would be vertical, indicating linear compliance and linear field strength

For this driver, the assembly performance is much better when the cone moves in the negative direction (actually the plus and minus directions are purely arbitrary). You can see the effect of gravity through understanding that the right-side-up positive motion is the same direction as the upside-down motion. Both of these curves are statistically the same. For this driver, gravity has no effect on the range of these measurements.

PERFECT PERFORMANCE

You can judge the data repeatability from the ±1V pulses. The lines connecting these points on both plots are nearly vertical, indicating that over this small power range the driver is performing perfectly. This connecting line also crosses the 0V axis at an IR of about 1.74 for both sets of data, indicating that the measurements have only a small experimental error.

The judgement about the driver performance from these tests is that the suspension and coil position is not well placed. A power transient producing a 6V peak will see a maximum impedance difference of about 6% from the high peak to the low. Can you hear this difference? Who knows?

Fortunately, driver cones have the same "forgiving" characteristics as unbalanced push-pull transformer circuits, where the power null moves to the magnetic null—unlike a hard-linked mechanical suspension that remains nonlinear where the null is fixed. A suspension imbalance or misalignment is obviously unacceptable if the system "bottoms out," that is, if the IR flattens out with increasing displacement.

This may indicate that the voice coil has reached the limit of the magnetic field where additional current cannot provide additional force on the suspension. It may also indicate that the cone suspension has reached a mechanical limit, either as a "hard" stop, or because the compliance has overcome the force product of the voice-coil current and the magnetic field. — **DJ**

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Reader Service #62

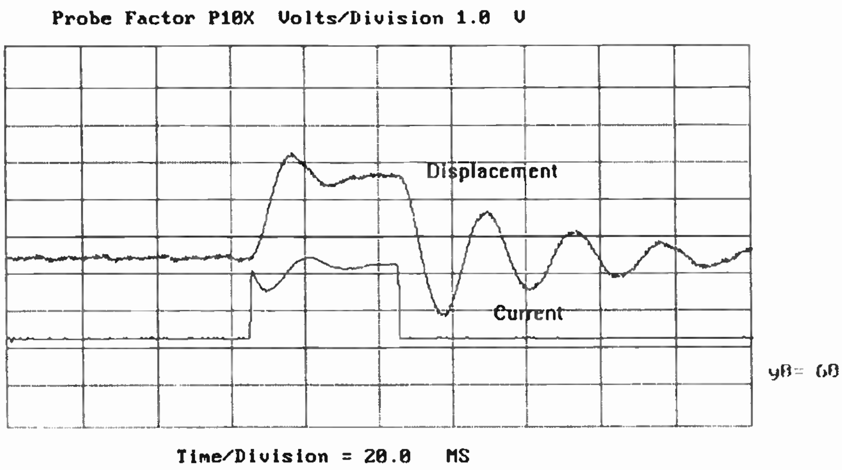

FIGURE 4a: 8″ test driver motion axis vertical, right side up.

expression should also be given, rather than the magnitude, and you should note the application of these quoted values. Filter-element design using a quoted impedance value should use a value representing the conditions of application. If the complex value is given, you should at least make an extrapolation to the frequencies of interest.

TRANSIENT INPUT EFFECTS

For power transients, which are the real application problems for audio systems, the impedance can change significantly over certain transient inputs. Balance and notch-filter design may need to be different for various types of program material. Filter-attenuation characteristics, when used as circuit modifiers with a specific driver and amplifier combination, will have different response characteristics for smooth violin passages than for Sousa marches. Slow-sweep sine inputs to a filter, driver, and amplifier combination will show different impedance characteristics than a program source with rapid and high-amplitude transients.

A detailed analysis of *Fig. 2* will illustrate this point. When the cone accelerates forward as a result of the input 2V DC pulse, the back EMF generated by the cone's increasing velocity also increases. Resistance to this acceleration is provided by the compliance of the cone, the inertia of the accelerated mass, and the effective reduction of the input current by this back EMF.

The cone finally stops its forward acceleration when the compliance force equals the effective driving current. This point is at the first maximum value of the back EMF, at which point the cone velocity starts to decrease and equals zero at the first displacement maximum (3.5ms). During this acceleration part of the cycle, the impedance will have reached its maximum when the cone velocity is also a maximum (maximum back EMF).

For the case shown, the impedance seen by the driving circuit at this point is 12.9Ω. Since the voice-coil resistance is 7.1Ω, if you were to use this impedance magnitude to calculate the complex impedance value, the result would be 7.1 + j 10.8. In electrical terms, this would indicate an inductance value of 10.8Ω.

CLASSIC VIEW OF INDUCTANCE

Classically, inductance is regarded as energy stored in a magnetic field. In the case of drivers, this energy is stored in the compliance of the cone and in the momentum of the cone mass. Since both of these parameters are constantly changing during driver excitation, it follows that the effective impedance of a driver is also changing. When these values have come to equilibrium, that is, when the input signal and the mechanical response show no change relative to each other over continuing power cycles, then you can calculate the magnitude of the impedance by using the RMS values of measured input voltage and current.

The resultant impedance value will be unique for this condition. In effect, the calculated impedance is an RMS value. With the conditions quoted in *Fig. 2*, this value is calculated as 7.1 + j 3.5. This value has meaning only for a 30ms pulse at 2V DC. Any other pulse duration or voltage would yield a different value.

This statement is true for all drivers. However, most driver specifications and design objectives assume that the compliance of the cone structure will be linear over the design range. This results in the impedance integral being unchanged by input amplitude. For small displacements, this is surely true.

However, for large transients, this will not be the case. In the computer simulation used for *Fig. 2*, the compliance is input as a constant. When actual, nonlinear measured compliance data is used in the simulation, the impedance integral will in fact change with input-signal amplitude.

Impedance always changes with transients. For the case of *Fig. 2*, as the DC pulse width increases and the initial transient becomes a smaller and smaller fraction of the total pulse width, the impedance approaches the voice-coil DC resistance.

Figures 3a and *3b* show actual measurements for an 8″ driver. Obviously, the model outputs of *Fig. 2* show very good simulation. By using a model, any of a set of selected parameters can be presented on the same plot. Unfortunately, that is not possible with a two-channnel scope.

However it is clear that the model simulation is correct. In these tests, the voltage-input pulse is held rock solid with a 90,000μF capacitor across the DC bus. The current record is a mirror image of the impedance record. Compare the actual current record with the model impedance curve (*Fig. 2*).

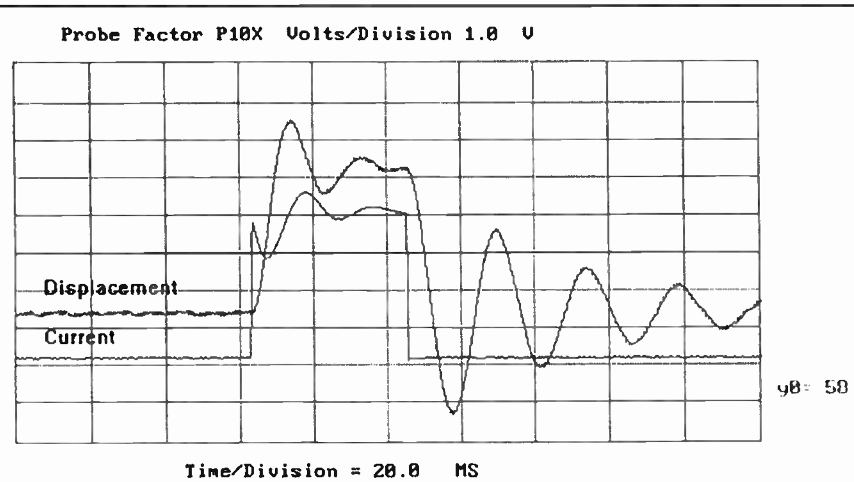

FIGURE 4b: 8″ test driver motion axis vertical, upside down.

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Reader Service #91

THE PRE-CUT SPEAKER

By Kenneth L. Bird

Having only simple woodworking tools, I always try to design my loudspeaker systems around the dimensions of lumber available from my local Builders Square or other large lumber-supply house. This is a challenging part of the speaker-building game, and has produced some easy-to-build, fine-sounding systems.

The model shown in *Photo 1* is a 7ft^3 sealed-box type designed with the simple BASIC program recommended by David Weems in his book, *Great Sound Stereo Speaker Manual*. These are large speakers, yet their footprint is only 13.5″ × 24″, and they will fit into most rooms without dominating the area. They work best spaced 10–15′ apart, and a corner location will give you all the bass you'll ever need.

PHOTO 1: The finished speaker.

CABINET CONSTRUCTION

Figure 1 shows the dimensions of the box parts. The front and rear are composed of 24″ × 48″ pieces of ¾″ pre-cut plywood panels finished on one side. The top and bottom pieces are also ¾″ plywood, 13.5″ × 24″. I cut these pieces out myself, but your local lumber supplier will do it for you.

The sides are 12″ × 48″ ¾″ particleboard shelving, another ready-to-go product at most lumber-supply stores. I reinforced the panel joints with 2″ × 2″ framing lumber, screwed and securely glued with Elmer's™ Glue. I caulked all the joints and placed a 2″ × 2″ brace between the front and rear panels above the uppermost woofer.

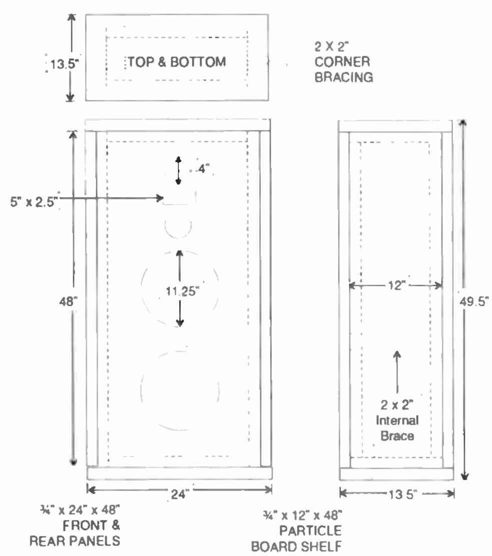

FIGURE 1: Box design and dimensions.

DRIVERS AND CROSSOVER

The drivers are by Madisound and are among the highest quality I've experienced for the money. The woofers are dual-voice-coil Model 1252DVCs. The dual voice coils are wired to achieve an 8Ω impedance, making crossover design more traditional.

The midranges are 4.5″ Madisound Model 5102Rs—4Ω versions wired in series to produce 8Ω. Each midrange is in its own enclosure made from round plastic food-storage bowls. I cut driver-clearance holes in the lids large enough to allow the speakers to fit, and then attached them behind the front-panel holes with Liquid Nails™. Once they were set, I snapped the bowls onto the mounted lids and then stuffed them with polyester fill.

For the tweeter, I used the Motorola Twin Bullet piezo (Radio Shack #40-1392), mounted between the midranges in MTM fashion. The treble is smoothed out a bit by the addition of an 8Ω resistor across the tweeter, and the MTM arrangement also provides better response, with no serious peaks in the upper range.

The crossover (*Fig. 2*) is of a conventional design of 6dB per octave, crossing at 700 and 5kHz, respectively. I used commercial capacitors and wound my own coils, using a B+K Model 878 component tester to check the values. The 878 is a valuable tool, as it accurately measures capacitance, inductance, and resistance. For those who want a ready-to-go crossover, I did test the Radio Shack #40-1299 three-way unit, and it worked quite well in the system.

I brought all the driver wire leads out through a single ¼″ hole in the rear panel and terminated them with the crossover in a plastic Radio Shack project box (#270-224). I mounted five-way binding posts on the box lid and soldered them to the crossover input, an arrangement that allows an easy change of crossover components or future biwiring.

FINAL DETAILS

The cabinet is lined with R7 household fiberglass insulation, stapled to the rear and side walls. I mounted all the drivers with screws and caulking to ensure a good seal, and all connections were soldered. A set of casters on the cabinet bottom makes moving the unit much easier.

ABOUT THE AUTHOR

Kenneth Bird has been building speakers since 1954, and was installing 12″ speakers in the rear decks of 1958 Chevys and Cadillacs long before the word "autosound" was coined. Armed with a BSEE, and much later an MBA, he has spent much of his career in audio-related pursuits with such firms as GTE, Rockwell, and Shure Brothers. He is currently a product manager with MAXTEC International, makers of the B&K line of test equipment.

TABLE 1

PARTS LIST

QTY.	DESCRIPTION
2 each	24″ × 48″ × ¾″ plywood handy panels (front & rear)
2 each	12″ × 48″ × ¾″ particleboard shelves (sides)
2 each	13.5″ × 24″ × ¾″ plywood (top & bottom)
8 each	2″ × 2″ × 8′ framing lumber (for interior & corner braces)
48	1½″ #8 wood screws
2 tubes	silicone caulking
1 roll	6″ fiberglass insulation
1 quart	Elmer's wood glue
2 each	Rubbermaid 1.47-quart food containers
1 quart	flat black latex paint
OPTIONAL:	
2 pairs	1″ rubber casters for bottom mounting (one pair fixed, one swiveled)

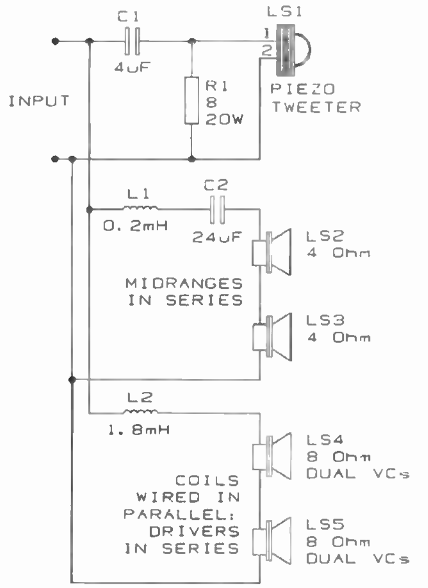

FIGURE 2: The speaker crossover.

I have yet to install grilles, although I did build them out of screen-door trim reinforced with 1/8″ plywood gussets. I like looking at the drivers, so it may be a while before I attach the grilles.

The cabinet finish is up to you. I elected just to paint it flat black for use in my den. For locations that need a more decorative approach, you could use a veneer treatment, or perhaps attach speaker carpet to the sides and top.

SYSTEM OPERATION

The measured f_3 of the system is 28Hz, and a continuous 100W sine wave did it no harm. These speakers love pipe organs, large orchestral works, opera stages, and massed choirs, while the mid- and upper-range response is great for the individual instruments of jazz combos and big bands. Vocalists are right where they should be—center stage.

These speakers would work well for a portable DJ or medium-sized auditorium system. They are examples of what you can achieve with minimal woodworking tools, dimensional lumber, and well-engineered, yet low-cost, drivers.

WAVEFORM PHASE DISTORTION

By Dennis P. Colin

Did you ever wonder why that speaker with a precisely aligned fourth-order crossover sounds smooth, but isn't stunning in its reproduction of drums, bass, guitar, piano, and bells? It could be phase distortion. I am not speaking of spatial phase distortion, such as the image smearing caused by excessive relative driver phase difference or large spacing/wavelength ratio in a multiway speaker system.

Waveform phase distortion is delay dispersion: various frequencies being delayed differently, distorting the shape of any except a pure sine waveform. It is also known as group-delay distortion. Group delay—defined as the negative of the phase versus frequency slope on linear scales—is a fair representation of the actual time delay encountered by transient events such as pulses and signal amplitude changes.

How much phase distortion is audible? As you will see, it's a matter of degree (pun intended)!

RESEARCH

In 1971, I designed and published (*JAES* 4/71) a version of the op-amp state-variable filter, originally for the analog ARP music synthesizer. By suitably combining the filter's outputs, you can realize an all-pass response, which has a flat amplitude-versus-frequency response, but the phase of which changes by 180° from low to high frequencies. (This is a first-order all-pass response, which an ideal second-order speaker crossover produces.)

Listening to electronic pulses or close-miked percussion sounds, I could plainly hear the effect of two or more all-pass stages. The maximum effect occurred when the phase curve was centered around 200Hz. The sharp transient "click" was smeared into a quick but audible downward-sweeping "teeooup" sound! (All-pass responses delay low frequencies more than high ones.)

ABOUT THE AUTHOR

Dennis P. Colin has a BSEE from the University of Lowell (MA) and is currently an analog-circuit design consultant for microwave radios. Previously a band keyboardist and recording engineer, he has been published in the *Journal* of the Audio Engineering Society.

In a demonstration at the Boston Audio Society, with orchestral recordings played in a large room, it required five to ten all-pass stages to hear an effect. But when I recently repeated the experiments with headphones and various speakers in a small, acoustically dry room, I found that tapping a pencil on the plastic lid of a towel-stuffed coffee can, close-miked, produced a very sharp, well-damped, repeatable acoustic transient.

With this sound, I could barely hear the smearing effect of only one all-pass stage. With two stages (similar phase shift of a third-order crossover), the blurring and loss of impact were quite noticeable. With three or more stages, the smearing was dramatic! Even steady-sounding tones rich in coherent harmonics—such as bowed string bass and low horn tones—lost much of the "pulsiness" that you can physically feel.

With the phase shift centered at frequencies higher than 200Hz, it required progressively more phase shift to hear. But since the delay dispersion of an all-pass response is inversely proportional to its center frequency (90° point for a first-order all-pass), the audibility threshold was a roughly constant value of group-delay smearing—about 1.6 ms (first-order all-pass centered on 200Hz) for sharp, dry transients. With the phase-shift center frequency moved much below 200Hz, the audibility decreased, due to the ear's reduced sensitivity to timing details at very low frequencies.

ANALYSIS

Note that the first-order all-pass phase curve in *Fig. 1* is symmetrical about the center frequency on a logarithmic frequency scale. The circuits shown have 0° phase at DC and approach −180° at the highest frequencies. If you interchange R and C or invert the absolute polarity, the phase is +180° at DC and approaches 0° at high frequencies.

In either case, the phase slopes downward as frequency increases, representing a positive value of group delay. (Negative group delay is possible, but uncommon, over a restricted frequency range—that's why I previously described group delay as a "fair representation" of actual time delay. It usually corresponds to what is heard and seen on an oscilloscope.)

PHASE DISTORTION IN LOUDSPEAKERS

Most good drivers have a phase shift reasonably consistent with the amplitude-response rolloff. Known as "minimum-phase response," anything that alters amplitude versus frequency response has a necessary phase shift, but with an ideal first-order crossover, the combined response has a reasonably linear phase curve (constant group delay). However, all-

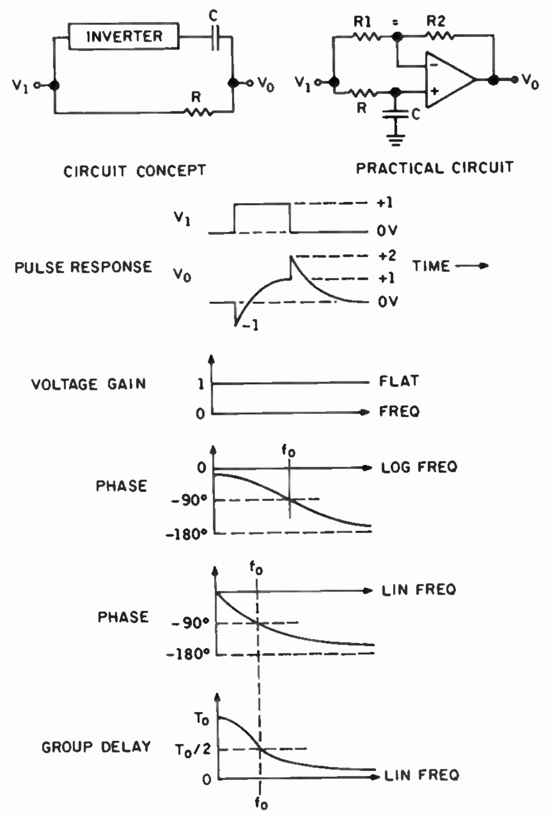

FIGURE 1: First-order all-pass.


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pass responses are nonminimum-phase, and cannot be compensated for by any real-world circuit, since a positive exponential (endlessly growing) response would be needed in the compensator.

Crossovers higher than first order all have phase or group-delay distortion (*Fig. 2*). For example, a fourth-order crossover at 200Hz has a delay variation of 4.8ms, plainly audible with sharp transients. This audibility is demonstrable in scientific tests, unlike, for example, the "sound" of a resistor, capacitor, and most cables.

From *Fig. 2*, it's apparent that higher-order crossover transient responses approach a "chirp" effect (used in radar), where the highest frequencies arrive immediately, but the lower ones are progressively delayed.

Phase distortion has another audible effect besides time-smearing. Loud sounds cause the ear itself to distort (about 1% THD and IM at 100dB SPL). Ask any musician who has heard two nearby flutes playing different notes; she'll tell you she hears low-frequency intermodulation tones, known as "wolf notes."

Now, notice in both *Figs. 1* and *2* that peak square-wave amplitudes can double, and then decay to the opposite polarity. Conversely, short but rounded pulses (prevalent in music waveforms) can have their peak amplitudes greatly reduced as different frequency components arrive out of phase relative to the all-in-phase spectrum of a single quick pulse. Therefore, the ear's natural distortion tones will be affected by the altered peak levels at high SPL, resulting in a changed sense of transient impact.

The combination of this effect and the time-smearing can result in an unmistakable loss of realism and detail in loudspeakers with high-order crossovers—unless you listen only to pink noise or sine waves!

SPATIAL DRIVER OFFSET

The distance between the two drivers crossed over increases the audibility of phase distortion: in a second- or higher-order crossover, you hear the highs first, and then the lows, but the delayed lows originate from a different location than the highs.

Consider a second-order crossover between midrange and tweeter at 3kHz, where the delay dispersion is only 0.11ms—well below the audibility threshold of about 1.6ms with no spatial offset. I did a test on two speakers, an A/D/S 300 C (5″ woofer, 1″ tweeter, and second-order crossover around 3kHz), and a Synchron SYN-519A coincidental unit (5″ woofer with a 3/4″ tweeter mounted inside the woofer voice coil). With both, I switched between a second-order and first-order crossover.

Listening to electronic pulses (1Hz

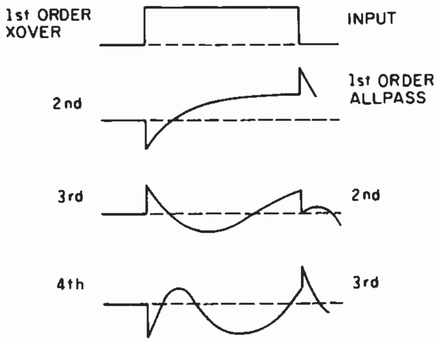

FIGURE 2: Higher-order all-pass step responses.

square waves), I could hear a difference between crossovers with the spatially separated A/D/S drivers, but not with the Synchron. Yet the microphone scope waveforms showed approximately the same crossover difference on both speakers. The waveforms were similar to those in *Fig. 2* (second-order vs. first-order crossover), except for the exponential decay due to low-frequency rolloff of the speakers. Also, I could hear the effect, though not dramatically, with drum sounds.

Thus, the phase distortion or delay dispersion of even a second-order midtweeter crossover can be heard in conventional multidriver loudspeakers, at least up to 3kHz. Of course, the ear is most sensitive at this frequency. I could not hear the delay dispersion of a second-order crossover at 5kHz or above.

NOW TRY THIS

Try this test: disconnect your speaker leads from your amplifier and touch them across a flashlight battery. The sharp click you hear when making the connection (as opposed to breaking it) is the speaker damped by the battery's low impedance, similar to that of an amp. Listen carefully to this sound. You may notice a sharp high-frequency "click," followed very quickly by a low-frequency "thunk," or the "teeoooup" effect mentioned earlier.

You should perform this test in a dry-sounding room, listening up close to the speaker. With no phase-delay distortion, and with reasonably flat frequency response, the battery pulse should sound like a sharp crack with no tonality. Of course, it will also dramatically reveal poor bass damping and significant resonances.

Fortunately, most of us don't often listen to flashlight batteries. But if you try this test with a full-range electrostatic, planar magnetic, or well-aligned first-order system, the battery connection will sound like the cabinet splitting in half.

FIRST-ORDER CROSSOVERS

Many so-called first-order systems "cheat" by inverting the polarity of one or more drivers. With drivers flush-mounted on the same plane, this usually is necessary to compensate for the higher phase lag (delay) of the lower-frequency driver, due both to its more recessed acoustic center and its larger mass. This causes a 180° difference between the drivers at a frequency that's often conveniently near the desired crossover point. So, inverting one driver avoids the deep response notch that would otherwise occur. But this polarity inversion results in a first-order all-pass phase shift with its delay distortion.

To achieve true first-order crossover response (no added phase distortion), there's no substitute for time alignment. Also, the drivers must have a smooth response to at least an octave on either side of the crossover frequency, because of the shallow 6dB/octave slopes of a first-order crossover.

The best way to achieve time alignment is first to design the crossover, taking driver impedances into account; then adjust the drivers' front-back separation to get the smoothest frequency response. (For a first approximation, have the voice coils in the same plane.)

THE OMNI-FOCUS

I designed and built a pair of speakers around the Synchron SYN-519A and the Focal 10″ woofer, which has superb bass fidelity. The 2ft^3 sealed cabinet top is tapered for time alignment and has one Synchron in front, directly above the woofer, plus a second Synchron in back for bipolar radiation. The crossovers are at 300Hz and 3kHz.

I spent two months designing and tweaking the crossover, finally achieving roughly ±4dB flatness from 30Hz–20kHz. Phase variation is only about ±20° from 100Hz–20kHz, and it reproduces a reasonably good square wave.

How does it sound? Well, the Synchron unit is not quite as flat as the best separate midranges and tweeters, but it is still quite good. However, the overall sound is excellent. Percussion instruments have extremely good impact, and the bass quality from the Focals with the linear-phase crossover is probably the best I've heard. With the phase-coherent image the Synchrons maintain at all angles, plus the bipolar ambience from the front and rear units, the realism and soundstage quality with orchestral music are excellent.

Regarding the bass accuracy of those Focals, I observed a clean, undistorted sinewave output from a microphone with a 6Hz speaker input at 20W—low power, but

to page 59

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250 pieces **Kef B200 8" woofer**, 4 ohm, part number SP1238, polypropylene cone, rubber surround, 4mm x-max, 32mm voice coil, 90dB, Fs 29 Hz, Vas 87 ltrs, Qms 1.47, Qes .35, Qts .283, 50 watts nominal, F3 of 45Hz in a .9 cubic foot enclosure with a 2" diameter vent x 4.2" long. A very special price of **$22.00 each.**

108 pieces **Vifa K10MD-19-08 4.5" Midrange**, 8 ohm, coated paper cone, rigid plastic frame, 111mm diameter frame with a 81mm diameter cut out, depth of 37mm, 89dB, 60 watts IEC, Fs 180 Hz, suitable for use in a small sealed chamber of about 1/2 liter for frequencies from 500Hz to 9KHz. You can buy like an OEM for **$8.00 each!**

51 pieces **Vifa P13MG-00-08 5" Midrange**, mineral filled poly cone, vented pole piece, rubber surround, cast frame, smooth response to 8KHz, 88dB, Fs 65, Vas 9.6 liters, Qms 1.36, Qes .67, Qts .45, usable down to 150Hz in a small sealed enclosure, 140mm diameter frame with a 115mm diameter cutout, **$12.00 each.**

50 pieces **Peerless S95095 12" Woofer**, part number 315 SWR 39 115 PPX 4L AL, 8 ohm, similar to 831857, polypropylene cone with rubber surround, 88dB, Fs 30.8 Hz, Vas 129 liters, Qms 4.34, Qes .85, Qts .71, 39mm diameter voice coil, X-max 9mm, 30 oz. magnet, F3's in the mid 30's for sealed boxes of 3 to 5 cubic feet. **$45.00 each**

200 pieces **Stanford Acoustics MSF-100B20U, 4" woofer**, Shielded magnet, paper cone, foam surround, 4 ohm, 89dB, Fs 68.5 Hz, Vas 4 liters, Qms 1.86, Qes .43, Qts .35, smooth response to 3KHz, F3 of 140Hz in a 1.3 liter sealed enclosure or an F3 of 85Hz in a 2 liter enclosure with a 1" diameter vent x 4" long, only **$7.50 each.**

This offer expires April 30th 1997

67 pieces **Vifa P13MG-10-04 5" Midrange**, 4 ohm, optimized for autosound use, mineral filled poly cone, cast frame, very flat response to 9KHz, 90dB, Fs 80 Hz, Vas 5.7 liters, Qms 2.7, Qes .67, Qts .54, recommended for frequencies above 200Hz, 140mm diameter frame with 115mm cutout, very good off axis response to 3Khz, **$12.00 each**

300 pieces of **Hi-Tech HT10-08 10" Woofer**, Polypropylene cone with foam surround, 16 oz magnet, vented pole piece, 1.5" diameter voice coil, 4mm X-max, 8 ohm, 89dB, Fs 23 Hz, Vas 163 liters, Qms 3.32, Qes .37, Qts .33, F3 of 49Hz in 1.7 cubic feet sealed or F3 of 32Hz in 2.7 cubic foot vented box, 3" diameter vent x 7" long, a steal at only **$26.00 each!**

12 pieces **Davis 20SCA8 8" Carbon fiber cone Woofer**, rubber surround, 8 ohm, 93dB, Fs 41.4Hz, Vas 65 liters, Qms 2.11, Qes .35, Qts .30, 39mm diameter voice coil, 100 watt RMS power handling, F3 of 95Hz in .5 cubic foot sealed or F3 of 60Hz in 22 liters vented 3" diameter vent x 5.75" long, cast frame, smooth response to 3KHz. Close out at 1/2 our cost, **$30.00 each.**

8 pieces **Davis 20KLV8A 8" Kevlar cone Woofer**, rubber surround, 8 ohm, 94dB, Fs 38.5Hz, Vas 81.7 liters, Qms 2.94, Qes .44, Qts .38, 39mm diameter voice coil, 100 watt RMS power handling, F3 of 70Hz in 1 cubic foot sealed or F3 of 40Hz in 2 cubic feet vented 3" diameter vent x 3.7" long, cast frame, very smooth to 5KHz, Close out at 1/2 our cost, **$30.00 each.**

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MONITORS FOR NONENVIRONMENT ROOMS

By Philip Newell

PHOTO 1: Recording studio constructed according to "nonenvironment room" principles.

With the advent of the "nonenvironment room" concept in the mid-1980s, a new problem arose for the designers of suitable monitor loudspeakers. With this type of room, the entire space is made to be as absorbent as possible, except for the front wall and the floor. By mounting the monitor loudspeakers flush in the front wall, you are effectively terminating them into an anechoic room.

From the point of view of the recording engineer, the majority of the floor reflection will rise to ear level only behind the listening position, and will then be absorbed into the rest of the structure. What the design concept seeks to do is to remove the variability of the room from the monitoring by making it acoustically dead from the loudspeaker position. You make the front wall and the floor hard in order to add life to the speech and general activity in the room and to create a pleasant environment in which to work, but without adding any environmental artifacts to the monitoring (*Photo 1*).

The concept is discussed in detail in "Non-Environment Control Rooms," *Studio Sound*, Nov. 1991, and more recently in the paper "Control Room Reverberation is Unwanted Noise" presented to the "Reproduced Sound 10" Conference of the Institute of Acoustics in the UK in 1994.

ENERGY REFLECTION

When rooms are to any significant degree reflective or reverberant, they reflect back into the listening area energy that is spectrally characteristic both of the surfaces from which they reflect, and of the on- and off-axis responses of the loudspeakers that excite the different reverberant and reflective fields. The aim in good control-room design is to achieve a uniform, smooth, and spatially distributed total-power response, total power being the direct energy from the loudspeaker plus the reflected energy from the room. No easy electronic rebalancing of this response is possible if the two are not linear in themselves.

Given the sensitivity of human aural perception, and as there is a considerable non-minimum phase content in much of this variability, it is hardly surprising that so many room designs sound different, both from one another and within their own groupings.

Indeed, in all except the "nonenvironment" approach, there is the further problem that any reverberant or reflective energy tends to mask much of the low-level detail in the direct signal. The masking often has a greater effect than the overall level of the reverberant energy would suggest, since this energy, unlike a noise-type masking, is concentrated in bunches around the frequencies in the direct signal; obviously, it is the direct signal that drives it.

So in the nonenvironment situation, you have few reflective problems to consider. The obvious first choice for a monitor system would appear to be a type with a wide and uniform pressure-amplitude (frequency) response. Such a system in this kind of room would probably give a good account of itself, but in a large room, it is difficult if not impossible to construct systems that can deliver enough high-frequency energy before running out of steam.

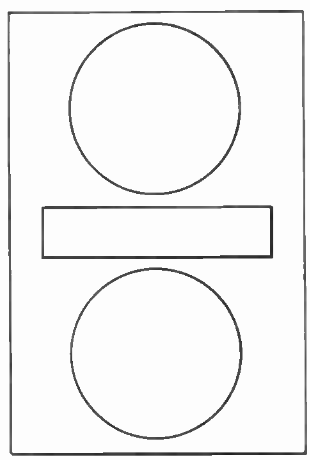

FIGURE 1: Kinoshita layout—16″ drivers above and below the mid/high horn.

ABOUT THE AUTHOR

Philip Newell has spent much of the last half of 1996 designing and building control studios in St. Petersburg (Russia), the Ukraine, Lisbon, and Barcelona.

PHOTO 2: Building a nonenvironment room. Note accommodations for the monitor loudspeakers mounted flush in the front wall.

NO HELP

As the room gives no help to the speakers in terms of added volume, the loudness in each part of the room must come from the direct drive of the loudspeakers. Consequently, if you drive an entire large room, you're wasting 12 or 15dB of monitor power (up to 90%) through driving entirely unused portions of the space. In effect, you must designate the working area of such a control room, and then cover that area with as smooth a response as possible. If achieving this objective creates a directivity anomaly at, say, 50° off-axis, it is usually of no great significance, as nobody is likely to be doing any serious listening from that position. The room will absorb the unwanted sound; it will not be reflected back into the listening area to color the response. The important thing is that when you *do* venture towards the extremes of the usable area, the response should change only gradually as you move progressively further off-axis.

In a highly absorbent nonenvironment room, any erratic response changes would be unacceptable, as every different listening position in the room would be subject to a different frequency balance, leading to a great deal of confusion. Only in the "hot seat" would the spectral response be as desired, and though many hi-fi enthusiasts accept this in their homes, it simply would not do in the working environment of a control room, where personnel must inevitably move around while discussing the same musical balance. Such a room offers no smoothing of responses from any uniform reverberation. It is once past the ear, and once only.

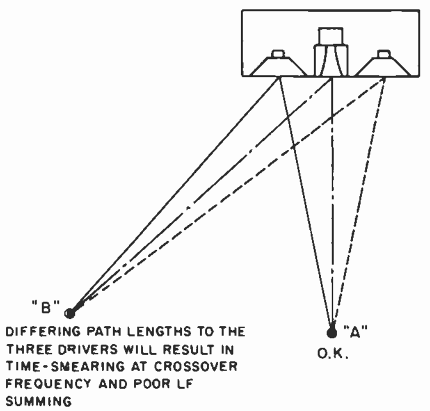

FIGURE 2: Horizontal mounting of monitor system shown in *Fig. 1*. At position A, all is well, with symmetrical positioning allowing equal arrival times from the bass drivers to the ear. For position B, however, the path length and hence the arrival time of the signal from the left-hand bass driver is shorter than that from the horn, whereas that from the RH driver is longer.

PHASE RESPONSE

What does become very noticeable in rooms free of reflection is that the differences in transient and phase responses of loudspeakers become much more obvious. An accurate phase response is necessary because of its effect both on the attack of notes, which carries much of the character of the timbre, and on low-level signal detail, which carries much of the spatial information.

I have referred in detail elsewhere to Manfred Schroeder's "phase organ" experiment, in which he could make his 31-frequency pulse train with zero-phase relationship produce "notes" above the buzz of the train by moving any one of the frequencies in relative phase. But you could hear this only under anechoic conditions. Any reflections in the auditioning room would make his "organ" unable to play "tunes," by rendering the notes completely inaudible.

The most nonminimum phase aspects of so many loudspeaker designs are the crossover regions, because of electrical summing problems and the noncolocated drivers. This problem often increases in proportion to the number of crossover points. The directivity matching of multidriver systems is also not a trivial problem, so in the highly phase-sensitive rooms I'm referring to here, the minimum number of drivers and crossover points would seem to be an important prerequisite.

PRACTICAL LAYOUTS

Certainly in the larger versions of these rooms, the higher sound-pressure levels required of the monitors—typically two to four times those of conventional rooms—have tended to dictate horn-loaded drivers for the midrange and high-frequency sections of the built-in monitors. Such a design philosophy does, however, allow for the minimalist approach of constructing systems capable of high output levels and low distortion, yet utilizing only two-way designs—the single crossover point creating fewer phase anomalies.

So far, the tendency in these rooms has been to place drivers in a single vertical line, either with one bass driver mounted below a horn, or, in larger rooms, with two bass drivers mounted with the horn in between them. *Figure 1* depicts the latter

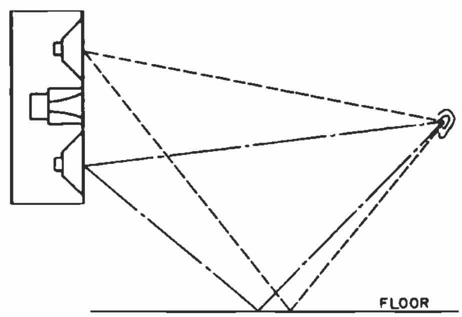

FIGURE 3: In vertical format, direct paths to the ear are similar, but the vertical reflection distances are considerably different. If these reflections were in the horizontal plane, then image smearing would almost certainly occur. Fortunately for most, reflections in the vertical plane are relatively innocuous in terms of image stability and definition.

ZALYTRON

Focus on Focal!

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FOCAL is pretty secretive about whom they supply with their famous drive units (and Klmon Bellas the FOCAL importer sounds like a locked bank vault when you ask him about it), but if you visit any high-end show, you see an awful lot of FOCAL drive units by many very respected speaker manufacturers. Well these high end OEMs, like any manufacturer on earth need to change models and change drivers. Sometimes only the paint of the frame changes, sometimes a little stiffer spider ... That is how you create very attractive obsolete product. When the French woke up form their traditional August siesta, we made an offer to FOCAL: ZALYTRON would take all these brand new OEM drives (with the usual warranty) they had on their shelves if they would sell it to us ... at an appropriate price. We would be hard pressed to print here that we made a killing, just see for yourself in the ZALYTRON tradition, we pass the benefit of our good deals directly to our customers. If this reads like the D.I.Y. dream list it simply is. If only these OEMs could keep changing models...

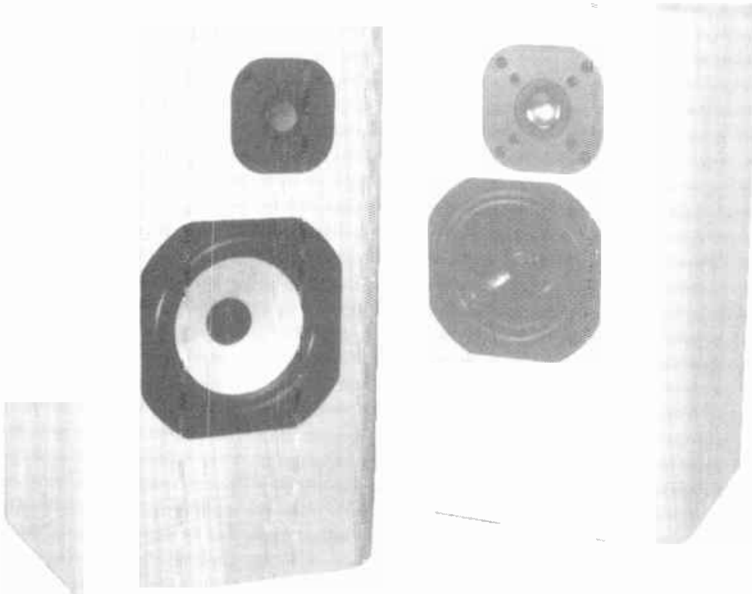

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PHOTO 3: The finished room shown in *Photo 2*.

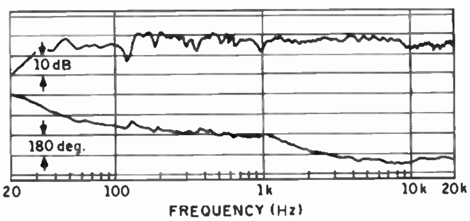

FIGURE 4: Two-way monitor measured on-axis at 2m in situ (24dB/octave Linkwitz-Riley electronic crossover).

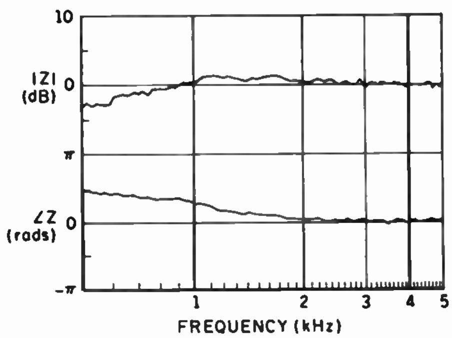

FIGURE 5: Throat impedance plot of AX2 axisymmetric horn.

arrangement, which was the system layout published by Shozo Kinoshita in the early 1980s. This layout ensures that as you walk around the room, the relative distance from each of the three drivers remains the same.

On the other hand, *Fig. 2* shows how the path lengths to each drive unit vary (as would their relative phase responses and, hence, coloration of the signal) if you turn the cabinet onto its side. In the horizontal layout, moving about the room produces constantly changing coloration of the sound, plus imprecise and ambiguous stereo imaging caused by the multiple path lengths and varying arrival times from the different drivers. This would be incorrect for all positions except those on a perpendicular to the center of the cabinet (e.g., position "A" in *Fig. 2*).

You would usually mount the cabinets flush in the front wall, with the center of the horn approximately 4′10″ above the floor (*Photo 2*). This height is a good mean between the location of the ear of a short person sitting down and that of a tall person standing up.

PSYCHOACOUSTIC OPTIONS

Although when standing up and sitting down, movement relative to the three drivers (as in *Fig. 2*) does take place, it does so at 90° to the effect shown therein. Movement in the vertical plane, however, has little or no effect on stereo imaging; moreover, the majority of people are relatively insensitive to coloration caused by vertical anomalies. *Figure 3* shows the typical reflections expected from a hard floor. When I first read Kinoshita's paper on this layout, I had many reservations. Indeed, when I now discuss the ideas with other designers, many of them immediately point to the problem of floor-reflection irregularities in the response at the listening position; but in practice, the effects are hard to detect by listening.

In some instances, you can use floor absorbers in the area where the direct wave would reflect to the ear from the floor. At worst, the effect produces a sharp notch from each bass driver, usually in the 100–200Hz range, but this has proved to be of little audible consequence. In the midrange unit, the vertical directivity causes the first direct wave to strike the floor at a point where it bounces up below the ear and is absorbed by the trapping systems.

If you were to install a loudspeaker and an omnidirectional microphone in a large anechoic chamber and place a large reflective surface some feet below, you could measure the effect of the reflective surface on the overall response. If you were then to rotate the whole anechoic chamber through 90° to bring the reflective surface into a vertical position (in effect, tipping the whole chamber on its side), nothing in the measurements would change.

Despite the fact that you could measure no response change, if you were to stand at the microphone position and remain vertical, your perceived sensations would be radically different, depending on whether the reflective surface was below or to one side. When compared alone to the direct sound, the reflector to your side would be far more disturbing than when the reflective surface was below you.

VERTICAL VS. LATERAL COLORATION

There are two mechanisms at work here that do not always stimulate equal reactions from different listeners. In the first instance, with the reflective surface *below* the loudspeaker, the reflection arrives at the listener from the same horizontal direction as the direct signal. It is then not surprising that the horizontal stereo imaging of the sound will be little disturbed. A reflection from the

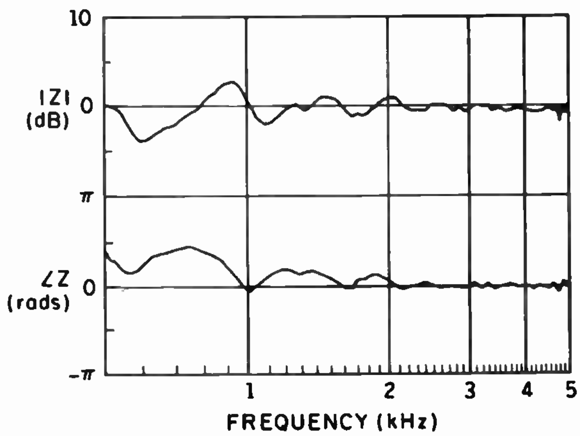

FIGURE 6: Throat impedance plot of a typical, commonly used rectangular horn.

RAVEN

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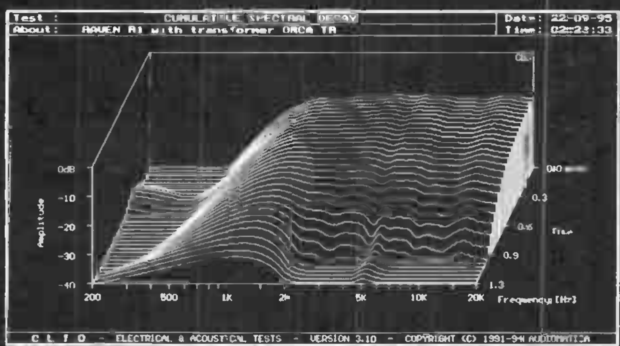
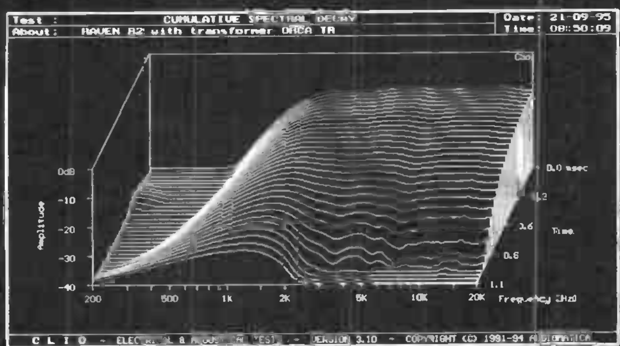
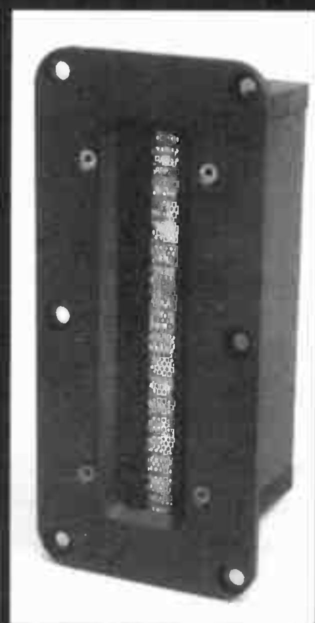
ORCA is sincerely proud of introducing these exceptional high frequency transducers from France. The RAVEN tweeter is a true ribbon tweeter, possibly the purest transducer available today. In a dome tweeter the signal is carried through the voice coil wire, and the sound is radiated by the dome attached to the voice coil. Here, the carrier of the electrical signal and the radiating diaphragm are one and the same part: the ribbon itself. Furthermore, the RAVEN ribbon is 100% pure conductive material, no metalized film. To have an idea of the high frequency performance of the RAVENs, imagine that the moving mass here is about 30 times less than a high quality dome tweeter. The music comes through effortless, almost immaterial. The special and massive NeFeB magnet of the RAVENs is five times more powerful than a conventional magnet. The result: the RAVEN R1 is capable of 118 dB peak with no measurable distortion (R2: 120 dB). At 10WRMS, that is continuous power now, R1 reaches 105 dB with less than 1% distortion, and R2, 107 dB. The RAVENs come with a specially designed matching transformer (very low distortion, low loss and wide bandwidth) for optimum coupling with your power amplifier. Now look carefully at the decay of these units !



RAVEN R1
KGS 1.14
LBS 2.5
92 x 80 mm
3.63 x 3.15 in.
Moving mass:
0.0061 g
0.0002 oz.
dB/W/m 95
2 KHz to 40 KHz



RAVEN R2
KGS 2.22
LBS 4.9
166 x 76 mm
6.54 x 2.99 in.
Moving mass:
0.013 g
0.0005 oz.
dB/W/m 98
2 KHz to 40 KHz



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Reader Service #32

side, however, comes from a point that is laterally offset from the direct sound, so you would expect spatial smearing.

The time smearing produced by either vertical or lateral reflection will be equal in both time and intensity. There is evidence, however, that the subsequent coloration is not perceived to have equal significance. The majority of people show a greater dislike for the laterally produced coloration than for that which is vertically produced.

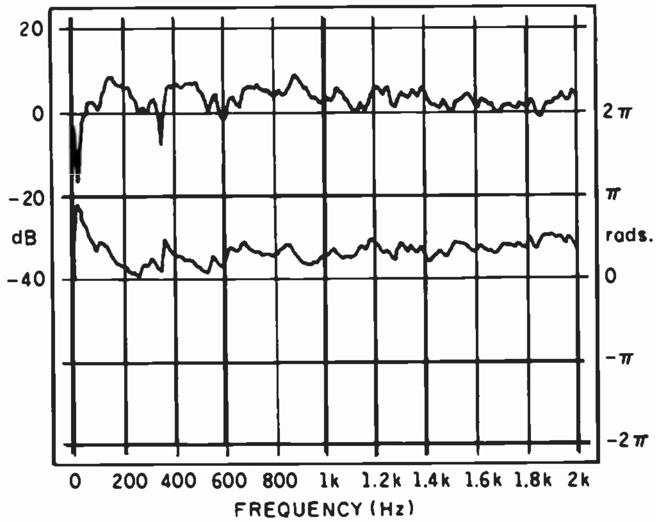

FIGURE 7: Response of a similar AX2 system through the 1kHz crossover region, using an electrically asymmetrical crossover.

Again, human inconsistencies do not make the life of a designer any easier. In general, however, the floor reflections are considered relatively innocuous. Based on the conclusion that floor reflections are largely ignored by most people, I used the Kinoshita concept as the basis for the monitor placement in my own versions of the new-style rooms (*Photo 3*).

Any of these reflections will cause overall frequency balance problems, unless the directivities of all the drive units match reasonably well. Reflections from any equipment placed in the room would also benefit from being timbrally neutral. Because I chose to use a horn as the only reasonable option in such a two-way, high-output system, the facts suggest that it should have an overall directivity pattern that corresponds well with the cone drivers used below the crossover point. This points toward an axisymmetric horn as the only realistic choice.

CHOICE OF COMPONENTS

Once an appropriate layout of drives is chosen, anything other than a two-way system is difficult to realize in practice. To help outline the actual decisions facing a designer, I shall use the Reflexion Arts 234 as an example of the thought processes that led to a practical solution to the problems.

The two-way design tends to dictate a crossover point somewhere in the octave between 600Hz and 1.2kHz when dealing with this scale of loudspeaker system. Anything smaller than 15″ for the bass drivers would have no chance of supplying the required low-frequency sound pressure levels. However, there is a very limited choice of 15″ loudspeakers that can respond at 20Hz, yet still continue to produce a smooth and linear response above 600Hz or so. The JBL 2235H is one of the few that can; and, indeed, its axial response is still respectable at 2kHz. The physics of cone loudspeakers predicts serious beaming problems once the diameter of the driving area reaches one wavelength of the highest frequency you're using. For a 15″ loudspeaker, this point is reached at around 1kHz.

Ultimately, you can determine which 15″ loudspeakers to use just by listening. I knew from extensive listening tests conducted for my earlier speaker-system designs that there was an openness and smoothness about the JBL 2235H that I did not find in other units. I knew of 15″ drivers that sounded slightly deeper than the JBL, but I knew of none that matched its overall performance from the 20Hz–1kHz range.

BOOSTING LOW FREQUENCIES

In order to assist the low frequencies, I used two 15″ bass units in parallel, enclosed in a 21ft^3 cabinet and tuned to resonate at 17Hz. It would have been possible to support the response around 30Hz by a higher cabinet tuning, but I chose 17Hz to give the smoothest amplitude/phase compromise by keeping the cabinet resonance clear of most natural musical frequencies. Two 12″ loudspeakers and a subwoofer below 60 or 70Hz might have been an alternative three-way option, but simplicity was one of my key design aims in the search for neutrality.

The midrange/high-frequency unit had to be capable of producing over 120dB at one meter. It also needed to have the desired axial frequency response so it could be used without equalization, and it had to have an off-axis response that met the criteria established for these rooms.

"Round The Horn," an article in the March '94 issue of *Studio Sound*, discussed at length the development of such a drive system. The combination of the TAD TD2001 beryllium diaphragmed compression driver and the AX2 axisymmetric horn seemed ideally suited for my requirements. Certainly its axial pressure-amplitude response was well within the design criteria for monitoring purposes, and its directivity characteristics had been the subject of a paper, "Axi-symmetric Horns for Studio Monitor Systems," presented to the Institute of Acoustics conference in 1990.

The use of this horn fixed the crossover frequency, as the horn itself possessed a 12dB/octave rolloff below its cut-off frequency of around 1kHz. The concept of this combination appealed to me both in the way that it naturally fell together and because of its simplicity. I could produce a total system in two-way form, with drive units that could respond within the design specification limits from 20Hz–20kHz, without needing any electrical contouring of the response. It was also based on units that had shown themselves capable of producing very natural and transparent sounds when assessed subjectively.

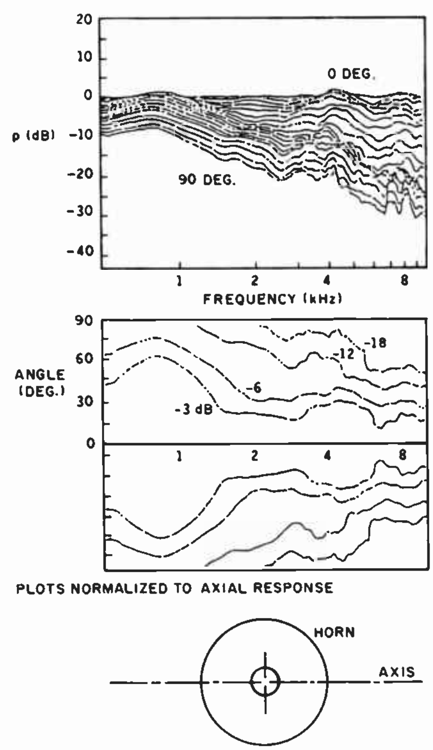

FIGURE 8: Excellent axial and off-axis frequency response performance of AX2 axisymmetric horn, and plots normalized to axial response. Felt diffraction rings around the perimeter can help to linearize the 7–9kHz region of the 70–90° performance, if required.

TRIED AND TRUE

The amplification system followed my long-time tried and tested concepts of using electronic low-level crossovers and multiple amplifiers, with the whole electronic system responding from DC to 100kHz to keep the transient accuracy, and the phase shifts well out of the audible ranges. The only interesting difference here was that although the system used a 24dB/octave, modified Linkwitz-Riley crossover, the inherent 12dB/octave rolloff of the horn below 1kHz was incorporated into the overall design. This meant that while the electronic crossover had a 24dB/octave low-pass section, the high-pass section was only 12dB/octave, allowing the horn itself to contribute the "missing" 12dB/octave of the slope (*Fig. 4*).

The enclosures were approximately 2′ × 3′ × 4′, with the front panels being 4′ high and 3′ wide. I designed the boxes to be constructed of two layers of ¾″ chipboard, with a layer of $5kg/m^2$ deadsheet sandwiched between the two adhesively bonded layers. Three-axis bracing helped further to reduce panel flexing, and the whole was lined with ½″ foam, glued to the side walls. There then followed another layer of deadsheet, and a final internal lining of either felt or Dacron. I bonded all layers with appropriate adhesives, and designed the cabinets for mounting in the heavy, dense, hard front walls of the nonenvironment rooms, which act as large baffle extensions against which the low frequencies can push. (For the deadsheet, it is possible to use a $4kg/m^2$ roofing felt.)

SUBJECTIVE & OBJECTIVE ASSESSMENT

Figures 5 and 6 show the throat impedance, pressure amplitude, and phase responses of the AX2 axisymmetric horn, compared to the corresponding plots of a good, professional rectangular horn. The superior smoothness of the AX2 is obvious. The asymmetrical 12/24dB/octave crossover as described above smooths the response through the 1kHz crossover region, as shown in *Fig. 7*. *Figure 8* shows the AX2 directivity, and clearly demonstrates the gradual off-axis rolloff.

What was evident from the moment I first tested the systems was the ease with which I could adjust them to optimum high/low balance by the use of a known compact disc. Subsequent measurement showed that the initial setup on music was within a decibel of the measured optimum. Certainly in the horizontal sense, the systems behave as coincident sources, with the "phantom" location of the bass-driver pairs being directly over the center of the mid/high horn. Therefore, no matter where the listeners are located within the room, the sources remain subjectively phase coherent, thanks to the listeners' relative immunity from positional instability due to vertical (floor) reflections.

For all intents and purposes, the monitors in the nonenvironment rooms behave like loudspeakers mounted in the outside wall of a building, pointing into a field. The fall off in sound-pressure level is much more rapid as you move away from the loudspeakers than you would experience in a conventional room. The monitor systems must therefore generate higher SPLs than normal usage would demand.

OFF-AXIS PROBLEMS

Another function of this "free-field" listening environment is that while the axial response is all important at the listening position (since no room reflections exist to upset this response), any off-axis problems are heard to a potentially greater degree than would be the case in a more conventional room, where some ambient smoothing would exist. In other words, whereas in average rooms off-axis irregularities can produce *on-axis* disturbances via the reverberant field, in nonenvironment rooms, such disturbances are clearly impossible because there is no reverberant field.

If a "hot seat only" priority exists in a nonenvironment room, then axial-response accuracy alone will suffice. If a "wider" coverage is required, while the on-axis response requirements remain, the off-axis response becomes even more demanding than in many conventional rooms. Under these circumstances, an axisymmetric horn in a suitable overall system configuration provides a simple and effective solution to what has proved in the past to be a considerable problem.

What all the nonenvironment rooms have in common is that the monitoring is essentially the loudspeaker system. If you position equipment so that reflections are deflected *away* from the listeners and into the traps, then rooms of different shapes and sizes are capable of achieving a high degree of room-to-room compatibility. Different types of monitor systems, providing they meet the special requirements, are not as different as they are often considered to be. Bad loudspeaker systems, however, are quickly revealed as such.

For smaller nonenvironment rooms, I produced a variant of the monitor system described in the previous paragraphs, using only one 15″ bass driver and a horn. Mixes travel between the rooms without any subjective changes in balance. Remember though, systems of this nature will perform optimally only if the drivers are kept in a vertical line. Placing *any* two-way system on its side will negate many advantages of directivity.

TABOOS DISREGARDED

Sometimes also, conventional taboos become irrelevant. When I designed the system described above, I was concerned at first that there were three signal sources at the crossover frequency of 1kHz: two 15″ loudspeakers and one horn. I thought that possibly I should curtail the response of one of the bass drivers above 100Hz or so, in order to prevent interunit interference above mutual coupling frequencies (above which the drivers would act as independent sources).

In practice, however, when you are listening several meters away in a large room, the path-length differences to the three drivers become an insignificant part of the wavelength. If you listen at fairly close range, the narrowing directivity of the bass drivers, as the response is taken over by the horn, makes it almost impossible to be in the direct field of all three drivers at once. After exhaustive listening tests, I decided to dispense with the planned rolloff.

On the other hand, in a more conventionally reflective room, the reflective surfaces would almost certainly in some areas be driven by very different path lengths to the individual drivers. This would be the case particularly for reflections originating from the floor or ceiling. Do not accept without question truths that relate only to more conventional circumstances.

STEREO IMAGING

On the audiological aspects of such monitoring conditions, a few points should be noted. First, in comparison with normal rooms, you clearly notice the differences in stability between stereo pictures built up from pan-potted images and those from recordings using stereo-microphone techniques. The stereo-microphone images, which contain positional data largely from time and phase discrepancies, support themselves well. On the other hand, pan-potted images, relying mainly on amplitude differentials, frequently collapse when you are listening well off-axis. This is exactly what *should* happen, but some who do not realize this often comment that the stereo can be less stable than under more conventional monitoring conditions.

From within the designated listening area, however, the imaging appears clearly defined and highly stable. It is not true that such rooms do not support a wide stereo listening area; it is just that the width of the area depends upon recording technique, which is precisely what human audiology would demand. In general, as soon as people become accustomed to what is happening, everything rapidly becomes "normal" to them. Once the subconscious takes control and uncertainties disappear, the rooms quickly become easy to work in.

It is clear that less acoustic confusion in the rooms leads to less clutter entering the ears. This highlights once again just how different human perception can be from one person to another, and is one more reason why what is "right" in monitoring remains such a subjective, individual decision. In general, the more sonically neutral the monitoring conditions, the more these aspects of individuality become apparent.

CONCLUSIONS

In principle, all good-quality loudspeakers should perform at their optimum in nonenvironment rooms, since their responses will be less colored than in conventional rooms. When the designer is freed from the constraints of having to take into consideration a reflective or reverberant environment, life is greatly simplified.

The only significant additional demands are the need for a greater output capability due to the lower subjective loudness in a room of very low reverberation time, plus the greater need for excellent transient /phase response, since anechoic listening environments can highlight time/phase artifacts that go largely unnoticed in more conventional surroundings.

Experience shows that in nonenvironment rooms, the majority of multiband, multidrive-unit systems (i.e., three-, four-, and five-way designs) often have difficulties in "knitting together" that are quite readily exposed when you move around the room. These are, however, systems that are entirely acceptable in many other environments.

Nonenvironment rooms demand a great deal from a two-way system, and a horn of axial symmetry has been a great boon in the design of appropriate systems for them. This is another example of a system that benefits from a "waveguide" of appropriate design—in this case, utilizing a compression driver of advanced design.

You now have a whole range of loudspeakers using axisymmetric horns/waveguides, from the midfield, free-standing designs such as the Meyer HD2, to the Reflexion Arts 234 described above. Yamaha has also introduced a range of sound-reinforcement systems using waveguides/horns of axial symmetry. Over the years, some people have considered me to be a proponent of the use of horns, but I am not. At one time in my life, I was strongly set against their use, but occasionally I found myself with no other option. In almost every good system where I encountered them, their drawbacks were usually counterbalanced by benefits.

Small, fast-flaring horns of axisymmetric design appear to be able to embody most of the benefits, with a minimum of the drawbacks, and hence can be a useful addition to the loudspeaker designer's options. The physical acoustics of this concept does tend to preclude their use below about 1kHz, but it is very useful to have available a drive system that can run smoothly, efficiently, and with good directivity properties from 1kHz to beyond 20kHz. What is more, the systems can be highly energy efficient, and you can now use noncompression drivers in areas from which they have been largely excluded. It would not surprise me to see an increasing use of smoothly flaring "waveguides" of axial symmetry, as they do provide practical solutions to some loudspeaker-design problems that have vexed us for many years.

THE CLASSIC ALTEC 604 REBORN

By Otto Doering

When I was building a large room for music in a new summer home, I had two Altec 604-B woofers, so I set out to plan an Altec system. I read back issues of *Speaker Builder* to see what I should do to get the best sound possible from these old-timers while still following the Altec corner-cabinet design. The whole process took about six months, during which I learned a number of valuable lessons (*Photo 1*).

HEAVY MATERIALS

Years ago, I was fascinated with Wharfedale's sand-filled enclosures, but decided that was too much to cope with. So I went to my local lumberyard in search of 1″ medium-density fiberboard (MDF). They didn't have any, but were willing to order some 1¼″ MDF. Why not? I went ahead. A friend's wife drove her truck to help me pick up the 4 × 8 sheets, and one of them slipped while we were loading it, mashing her hand. This was my first warning that I was dealing with a very substantial material.

Altec's basic design for the 606-A corner cabinet (featured in *Speaker Builder* 4/90 as a Vintage Design) is great. (It also follows Briggs' basic 1948 specifications for woofer enclosures.) However, I decided to simplify construction of the face frame and mounting board so that I could remove the latter rather than the right-rear side panel. The simplified design, along with some rounding of dimensions, probably added 5% to the interior volume.

I rough-cut all the pieces of MDF with a skill saw, using extreme care in measuring the pieces and cutting to the outside of my marks. Then I took them all to a friend's shop and recut everything to a final exact trim on his beautiful giant Delta saw with an oversized table.

This was the right approach, for the 1¼″ MDF is terrible to handle in big sheets. When they're reduced to 26″ × 36″ sections, you can do the precision trimming and cut an edge at an exact angle on a quality table-saw setup. At this time, I also cut all the braces and laid out or set up all the pieces to be sure everything fit. I found that some sets fitted together a little better than others, so I labeled pieces for speaker 1 or 2 accordingly.

PREFABRICATION MAKES SENSE

What I had at this point was a pre-cut unit. Its home was to be 800 miles away, so I loaded the pieces into my station wagon and we took off for the summer place. I did not have the precision cutting tools available at the speakers' final home, so my precutting was essential. My task on site was to drill, glue, screw, and otherwise assemble the monsters.

If some bracing is good, lots is better! Following hints from various articles, I braced asymmetrically to avoid sections of panel with like frequency responses (the knuckle-rap test confirmed this). I added an asymmetric brace to the top panel, and made all braces and glue strips a bit oversize (*Photo 2*).

If some screws and glue are good, more is better. I used tube after tube of Liquid Nails and hundreds of square-head steel screws. (I chose the screws after reading Wayland's Wood World column on fasteners—*SB* 8/95, p. 42.) Everything had to be predrilled and countersunk. The 2½″ screws are in to stay.

More damping is always better. The Altec design calls for stiff fiberglass, which is no longer a common material. I finally found some at an office-remodeling firm that uses it above dropped ceilings. It is

PHOTO 1: Author Doering with driver-mounted speaker.

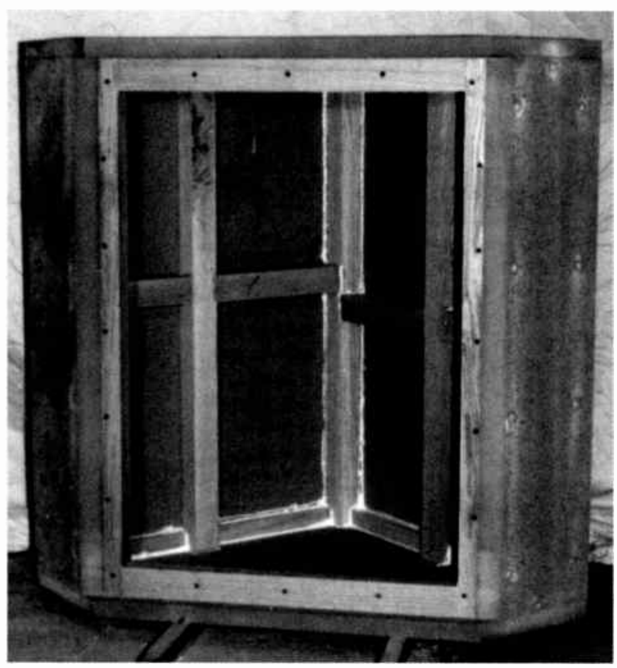

PHOTO 2: Inside the speaker with mounting boards attached.

wonderful for audio applications, having structural integrity and good damping properties. The Altec design called for it on the floor of the cabinet, on the narrow front sidewalls, and as a wedge inside to break vertical standing waves. I put the stuff on all surfaces except the top panel, and also used the wedge.

IMPROVISATION

One impressive result of my reading was the identification of asymmetry as a means of controlling standing waves, so I cut the hole in the mounting board a few inches off center. I didn't know what to use as gasket material for mounting the speaker, so I ended up using sheet cork. I did use a modern gasket in fastening the mounting board to the speaker box, which worked beautifully.

I fastened the board with many T-nuts (all tightened within an inch of their lives) to oak framing integrated into the box. On the inside, I caulked all the joints. I used the best quality binding posts I could find to go through the cabinet rear, and cut the port on the tight side of the Altec specifications (you can more easily make it larger than smaller).

TEST CONDITIONS

I now had about $600 invested in the speakers and materials. The cutting required about two days, and it had taken four days to assemble the first box and three for the second. Finally, I wheeled the finished speakers up to their room, a two-story space measuring 18′ × 40′ (12,000ft^3), with few parallel walls, a vaulted cruciform ceiling up to 22′ high, few large glass reflective surfaces, a stiff floor, and heavy wood panelling with wide beveled edges to diffuse sound.

It's a great room, just large enough to have its own sound, with good diffusion and no apparent aberrations. My sound sources were a turntable and CD plus a small AVA rebuild of an old Dynaco 120—an extremely sweet-sounding small amp with a big heart.

The first thing that astonished me was the return to extremely efficient speakers. With the volume on the preamp turned all the way down, faint sounds were still coming from the speakers. Quarter volume was near my ears' limit, yet both amp and speakers were loafing. Listening a bit more, it hit me—transient response! In the tube era, when I would pore over equipment catalogs with friends, we always looked for transient response and the graph of the tone burst. This was correlated to magnet weight.

I don't know what the magnet weight is on the big 604s, but the total unit weighs over 50 lbs. (I had trouble holding them at times when fitting them into the cabinet.) Transient response is not something that has been talked about much in recent years, but the old Altecs have it in spades, and it adds tremendously to the sense of realism.

Finally, the cabinet actually was dead as a doornail! At ear-shattering symphonic volumes, a hand on the top panel feels little or no vibration or resonance. Some of this is just inertia, for each unit weighs slightly over 200 lbs.

The sound was awesome—virtually no distortion at any volume I could stand. At one point, I really cranked it up, opened all the doors and windows, and my mother and I sat on the lawn over 100′ away and listened to Wagner overtures—glorious sound. My tentative concerns about interior volume and port-size adjustments were forgotten. There have been no tweaks to see if I can make it sound better, for I might make it worse!

TECHNICAL SPECS

Using a Radio Shack meter, I found in the room a remarkably good ±5 or less dB from about 35Hz to where the meter becomes wuzzy just under 10kHz.

Technical specs aside, the Altec horn is absolutely unforgiving with shrill CDs. The room cuts a little of this, but eventually I am going to try a small tube amp, which is what the speaker was designed for. The old Altecs in the big boxes do not give me the imaging others say they get from new speakers; part of the problem may be that our piano is placed between them. However, there are some basic design parameters—tweeter dispersion, for one—that prevent the old-timers from imaging as well as they might.

THE MORAL

I have found that taking one of the great old speakers, using the old cabinet design that was carefully thought out and tested by this wonderful manufacturer, and adding what new knowledge I had gave me a marvelous product. Careful work also paid off, especially when building large boxes. The high efficiency and transient response of these old-timers are a revelation.

Finally, this is great therapy for those of us who dream of being creative. You also get great speakers that were "digital ready" 35 years ago.

ABOUT THE AUTHOR

Otto Doering is a Professor of Agricultural Economics at Purdue University. From 1959–1963, as a Cornell University student, he ran University Recording Service, making reference and master tapes. He still uses plain commercial-quality large-gauge stranded copper wire for speaker cable and commercial-quality low-capacitance shielded cable with high-quality plugs for interconnects. He agrees that many new expensive cables do sound "different," but prefers neutral transmission. The money saved also buys better speakers.

Ask SB

IMP MERGE FUNCTION

I am using the IMP analyzer and have unsuccessfully tried to produce a complete frequency response using the merge function. I guess I need a step-by-step example, so let me set a scenario. I am trying to develop a response from a ported box with a 10″ woofer mounted in it, with the response of the port combined with that of the driver. Since it's taking a near-field measurement, the SPL function is not used and the response is referenced to zero. After taking a response from the woofer, then one from the port, I have two frequency responses covering the same area on the spectrum that need to be summed. Do I paste the woofer first, then try to sum the port to it?

Also, when calibrating, should I set the gain so the response lies exactly on the 0dB line? And if so, does this mean that the frequency response will be in the proper spot vertically on the graph? If I recall correctly, adjusting the gain setting of the mike causes the response to move up and down the graph. I suppose I would just adjust the gain after the pulse is FFT'd, moving the response where I think 0dB should be.

My next question is how to take into account the step of the frequency when only a near-field measurement is taken in the lower region to, let's say, 300Hz. For example, say I have a 3-way speaker that needs to be crossed over at about 300Hz. A near-field measurement gives a response to about 300Hz. I now must match sensitivities of the midrange driver to that of the woofer. I can take an absolute SPL reading of the midrange, but I don't know exactly where to paste the woofer to it.

I thought of importing a near-field measurement of the woofer into CALSOD, and adding a step boost to it. Would this be correct? I guess I need a step-by-step procedure for developing a complete frequency response of a multiway speaker that is ported, and where the port response and woofer response must be taken separately.

Mark Giglio
Hammond, LA

Bill Waslo responds:

Using Merge in IMP can indeed be difficult and confusing. A major problem is determining the relative levels to be used when summing a driver and a port (or two drivers). A good article that discusses this (along with much theory on how to combine near-field and far-field responses) appears in Vol. 42, No. 6 of the JAES: "Simulated Free Field Measurements," by Struck and Temme. In short, the driver or port with the smaller radiating area S1 should have its near-field level reduced by the value $20 \times \log \sqrt{(S2/S1)}$ *[dB] before summing with a driver with larger area S2.*

With Merge, you work with one data set at a time, pasting or summing it into a Merge workspace. You must fix it into the workspace before getting the next data set to work with, or its effects on the workspace will not stay. The data sets are in memory (not data files), so you need to load each from disk (if that's where they are) individually, before you move them onto the Merge workspace. The Merge workspace will accumulate the effects of any Sum or Paste operations performed on it, which is how different data becomes combined in this progressive operation.

You can preview the effect of a Sum or Paste into the Merge workspace, but after you choose Fix, you can't undo the Sum or Paste, short of Transform Merge Clear, which is used for starting over.

You can adjust the gain or delay of data being newly Summed or Pasted into the workspace (by using the Preview options), but not of the data already Fixed there.

For your example, in which you have two .FR2 files you want to paste together, the following is one way to proceed:

1. Do Transform Merge Clear to erase anything you might already have in the Merge workspace. You can skip this if you're just starting.

2. Load in your LOWEND.FR2 file. You want to paste the entire data set into the workspace, so put Marker 2 at the high end of the data (around 650Hz). This is so you can use Below_marker 2.

3. Choose Transform Merge Paste Belowmarker2. IMP will report the frequencies for informational purposes ...press a key as directed.

4. Choose previewGain and set the gain to position the curve where you want on the displayed Merge space.

5. Choose previewDelay and set the delay for the desired phase curve changes (usually near level, but inclined slightly downward—if it's from a near-field measurement, it is probably OK as it is).

6. Choose Fix. This pastes the data LOWEND.FR2 (the part below marker 2), with the selected gain and delay changes, into the workspace.

7. Load in the file MID.FR2 using [File Retrieve...].

8. Put marker 1 at the frequency that you want to have as the seam between the new data and the data that is already in the Merge workspace. This is so you can use Above_marker1.

9. Choose [Transform Merge Paste AboveMarker1] and press a key, as requested.

10. Use previewGain and preview Delay, as before. This time, the display will show both data sets in the workspace as it would look if Fix were used at this point. Set the Gain and Delay, as appropriate, to properly join the data sets at the seam frequency (which you chose with marker 1). Since you are Pasting, the data from MID.FR2 will replace all the data (in this case, originally from LOWEND.FR2) above the frequency of marker 1 in the merge space....if you were Summing, it would be added to all the existing Merge space data above that frequency.

11. Fix the new data into the merge space to make it permanent. If you have more data sets to include, continue with them in a similar manner.

12. If desired, print out the data in its current high-resolution form.

13. Since we are done combining files in this example, press [Esc] once to back up to the Merge submenu which includes [To_freq_resp]. You can use this to make a single-frequency response data set in the "Acous" memory space, which can be saved, IFFT'd, and so forth. However, the data will always lose resolution at the lower frequencies and will always result in a high sample rate equivalent data set—that's why it's best to make a printout from the Merge display.

14. Set Gain and Delay back to 0 to see the new combined frequency-response data set as it exists in memory (which should coincide with what you saw in preview).

Your second question is "Where is 0dB?," and the answer is "Wherever you want it to be." Decibel notation indicates a level's relative value with respect to some reference level. For instance, 2V is +6dB relative to 1V, or +12dB relative to 0.5V. For IMP measurements of a speaker's frequency response, you are looking for a sound pressure relative to an input voltage, which are two different physical entities, so the dB reference isn't easy to define.

There is such a thing as a defined value for 0dB Sound Pressure Level (SPL), which can be a reference for the pressure. And a voltage level that equates to 1W into 8Ω is often taken as a voltage reference level. So you can measure "dBSPL with 1W input," but then you also must specify the microphone distance used, because *sound level normally drops off with distance. You must consider this and all measurement gains if you are using the "absolute SPL sensitivity" format and procedure in forming far-field data.*

But usually, when working with speaker crossover design and driver or baffle placement evaluation, you are concerned only with relative response flatness and don't need to bother with all the tedious scorekeeping needed for SPL sensitivity measurement. For relative response, you want to see only whether the response varies from other parts of the response curve by more than some dB value (or with some target characteristic). So you choose 0dB to be whatever gets you the best (most easily viewed) picture of the curve on your display, usually when most of the curve is centered around the line marked "0dB." The easiest way to move the curve up or down to accomplish this is to use the Gain parameter ([F8]), which adjusts the dB vertical placement when a frequency-response curve is being displayed.

Your question about the baffle response step is not easy to answer. This effect can be quite problematic, as the baffle step comes into play quite near in frequency to where the floor echo usually does its damage. So if you remove normal room echoes in mid- or far-field measurements by truncation, you can't see the step because your measured response can't go that low; if you make a near-field measurement, you are too close to the driver to see the effect of the baffle edges, which are much further away.

If you want free-field measurement of the step, you may either have to resort to a large high-ceilinged room for measurement, or move outside and hoist your speaker well above ground. Or as you suggest, you could also use a simulation package to compute the step response. To get a good idea of the step trend, you could make a nonanechoic measurement about 1m or so away from the speaker and apply smoothing to the resulting hashy-looking response. But you will probably also see the effects of all the nearby boundary reflections getting into the act.

Determining a full-range anechoic frequency response on a single curve is usually possible, but is a difficult and very time-consuming task. Generally, unless you need a full "pretty" curve for marketing purposes, you can make do with a set of separate measurements of the high, mid, and low frequencies, perhaps using Overlay to show them simultaneously.

CORRECT EQUATION FOR Q_{MS}

In *Bullock on Boxes*, Robert Bullock gives the equation

$$Q_{ms} = \frac{\sqrt{r_0 f_s}}{f_2 - f_1}$$

Ray Alden's book, *Advanced Speaker Systems*, gives on the equation as

$$Q_{ms} = \frac{f_s \sqrt{r_0}}{f_2 - f_1}$$

(pp. 19 and 20). Obviously, someone is wrong. I suspect Bullock's equation is correct because it gives a more realistic answer. But, I ask, who is correct?

John J. Huff
Stockton, CA

Robert Bullock responds:

*f_s should not be under the radical sign. It is incorrect on the worksheets (*Bullock on

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Boxes, *pp. 16* and *17), but is correct in formula 9, p. 14. Thus, Mr. Alden's book has the correct formula.*

FREELINE ADVICE

Referring to "The Freeline: An Open-Pipe Transmission Line," by John Cockroft (*SB* 5/95), I have several questions.

1. I believe the objective of the article is quite clear. However, could Mr. Cockroft be more specific about the power handling of the speakers? What kind of SPL can the Freeline generate? My Radio Shack catalog says that the 40-1022 has 84dB/W SPL and can handle only 10W continuously. With the modification of the cone, I bet that the efficiency may be even worse. How bad is it?

2. What is the performance of the Freeline? How low can it really go? Is the speaker response flat?

3. Mr. Cockroft seems to prefer the Simpline to the Freeline in terms of sound. How good is the sound of the Freeline compared with commercial units? Is it better or worse than the Simpline, and in what respect (e.g., bass extension, distortion, timbre, or image)?

Bing Yang
Houston, TX

John Cockroft responds:

Thank you for writing to SB *and expressing your interest in my Freeline article. As with most of my designs, it is intended for use in small apartments where high SPLs—by landlord decree—aren't allowed. From experience I know that my speakers play loud enough to excite the landlord resonance factor, and that is enough satisfaction for me.*

My sole interest is to develop a sound that is natural and allows listeners to become as spiritually and emotionally involved as they would with live music. That, of course, is probably a goal no one will ever totally achieve, but it is what keeps me going.

Efficiency isn't something good or bad, better or worse—it depends on the situation. If you had to fill a great hall with sound, you would consider high efficiency good. If you had to design a system with greatly extended bass response, you might consider low efficiency good.

My concern with efficiency is how much of it must I give up to obtain the bass quality I desire. The quality of the music is my only concern. Of course, if, to achieve the quality, the efficiency must be so low that very little music is produced, then the quality isn't there, and I scrap the project. Articles I submit to Speaker Builder *contain projects that have a wide appeal.*

To give you some idea of how the Freeline and the Simplines sound compared to commercial speakers, I'll relate a recent experience. A reader who lives near me wrote that he had purchased a pair of Fried Studio V systems ($3700/pr), with which he was very pleased. He wondered why my design of the Freeline was so simple compared to the Frieds, and was curious to hear the Freeline.

We arranged for a mutual listening session at his home. I took the Freeline and my Simplines, which had by this time become Super Simplines (SB 1/96) with the addition of a tweeter. We listened to the Frieds first, and I was impressed with their solidity and clarity. We then set up the Freeline. He and I were both astounded at how closely, from top to bottom, the Frieds and the Freeline sounded in his living room. Of course, the Frieds are much larger and more powerful speakers, capable of considerable sound around 20Hz, and undoubtedly could have played at levels that would have fried (no pun intended) the voice coil of the Freeline. We both felt the Freeline offered most of the spiritual and emotional involvement of the Fried speakers. The pitch, timbre, and coherency of the two were substantially the same.

We then set up the Super Simplines. Both the Frieds and the Freeline are forward-facing radiator speakers, so a direct comparison was possible. The Simplines are reflective speakers and sound much more open and airy than the other two. However, the bass and midrange were very much of the same cut. What really amazed me was the volume that my systems put out. I think it was probably due to the acoustics of the living room we were in, with its high vaulted ceiling, hardwood floors, a minimum of drapery, and relatively unadorned walls. I kept waiting for the woofers to blow. Somehow they didn't.

The Freeline efficiency was reduced about 6dB by the cone modifications, making it about 78dB/W/m. For 10W input, that should be about 88dB. Loudspeakers can produce momentary peaks considerably higher than their ratings would indicate, but even so my systems are not designed for really loud playing.

Software Review

NETCALC

Reviewed by Jan Didden

NETCALC crossover calculation and optimization. Price—US $149. Authors—Rien den Boer and Rolph Smulders. Distribution—US: ARIBA, Baroniehoeven 32, NL-5244 HZ Rosmalen, Netherlands, FAX (+31) 4192 15841; UK: The Speaker Company; rest of Europe: Audio Components, the Netherlands.

Anybody who has tried to design a speaker crossover network knows that many factors influence the final result. Although most, if not all, of the mathematical relationships are known nowadays, the design process still is lengthy and complex because of the many iterations resulting from parameter interdependencies. Therefore, this is an area ripe for software implementation.

NETCALC is one such implementation to streamline the design process. It lets you enter the amplitude and phase versus frequency response of your drivers, your desired "target response," and your choice of crossover configuration. The program then does the number crunching. It will either optimize the crossover components for a specific channel (low, low/mid, mid/high, high) towards a previously specified channel response, or towards the total system target response.

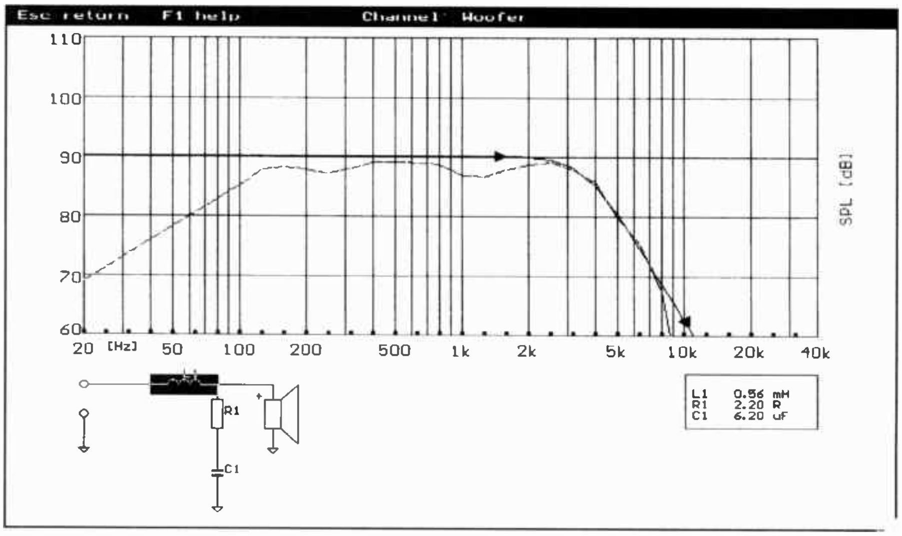


FIGURE 1: Sample screen display (interactive optimizer mode).

OPERATION

In calculating the acoustic response, the program accounts for specified geometric parameters, such as interdriver distance, driver "depth" on the baffle, listening position, and axis. You can also specify optimization parameters, such as the number of iterations, frequency interval to be optimized, and the acceptable error value between the target response and the optimized response. This error value is always calculated as the "average value of the RMS of the difference between the target and actual reponse." I think, however, that designers would also be interested in the *maximum* error values, since a close average match to a target response is not very useful if there are one or two large gaps.

During optimization of the crossover, you can "lock" certain component values, which are then not affected by the optimization algorithms. Alternatively, the program offers the option of optimizing the crossover network interactively. This works quite well.

In this mode, the program displays the channel or system response, along with the crossover circuit and the component values. You can then use the cursor keys to select any crossover component, and modify its value by stepping through the E12 (or E24) standard value ranges.

During this process, the program automatically updates the responses real time! It's quite fascinating to ramp up a parallel inductor value, for instance, to see the response straighten up. This is an excellent way of dividing the workload: you do the thinking, and the machine crunches the numbers. This immediate feedback also provides a direct, realistic feel for what is possible with a certain configuration.

PROGRAM FUNCTIONS

The program contains several "auto" functions, but also gives you the option to take charge manually, as with the automatic versus interactive optimization described above, for example. Another example is the entry of driver-response data. This can be done automatically by "importing" a data file, which can be from most popular speaker measuring systems, such as MLSSA, MLS, TEF, and some European systems (Kemsonic, ATB); IMP is excluded, however.

On the other hand, you can quickly manually specify a response through menu choices on a screen display similar to a graphics equalizer front panel with 1/3 octave "slides." It's easy to adjust the "slides" with the mouse to any driver response.

The program supports one- to four-way systems, and can also handle (indirectly) D'Appolito configurations. For the crossover configuration, you can select from a list of "standard" setups such as second- and third-order Butterworth or Linkwitz-Riley. You can also specify an attenuator in a network to match driver sensitivities and impedance correction components. You produce a custom filter by interactively specifying parallel and series elements (R, L, C) and actually "draw" the network on the screen. Very simple and very effective.

SYSTEM REQUIREMENTS

The program requires only modest resources: an XT or later computer, ½MB, 1MB hard disk space, and an EGA or VGA (or Hercules) adapter. Of course, a faster processor and a coprocessor speed up calculations. Surprisingly, this program (with its $149 price) includes copy protection.

You can install the program from the distribution disk to another system twice. If you then wish to install it on another system, you must "recall" (de-install) it from one of the first two systems. So, you can use the program on only two systems.

You make selections through the program's consistent system of drop-down menus and dialog boxes. Graphs are clear and choices are available for amplitude, phase, and impedance responses, as well as off-axis responses (up to ±15°). For documentation purposes, you can print responses and crossover networks on a wide range of output devices.

The program comes on a single disk, with a very sparse 15-page user manual, which is basically a short tutorial for a typical design cycle. It is enough to get you going, but I am sure there is more to the program than is covered in the manual. The author notes that he avoided a "heavy manual" to keep the price low. However, once you start using the program, questions that are not addressed in the manual may arise.

For instance, you can specify a crossover frequency for the target response and for each separate crossover network. What are the implications of this during optimization? Some important information (for example, why you should do channel optimization before system optimization) appears under "Hints" at the very end of the manual. The manual does include context-sensitive help for the menu selections.

SUMMARY

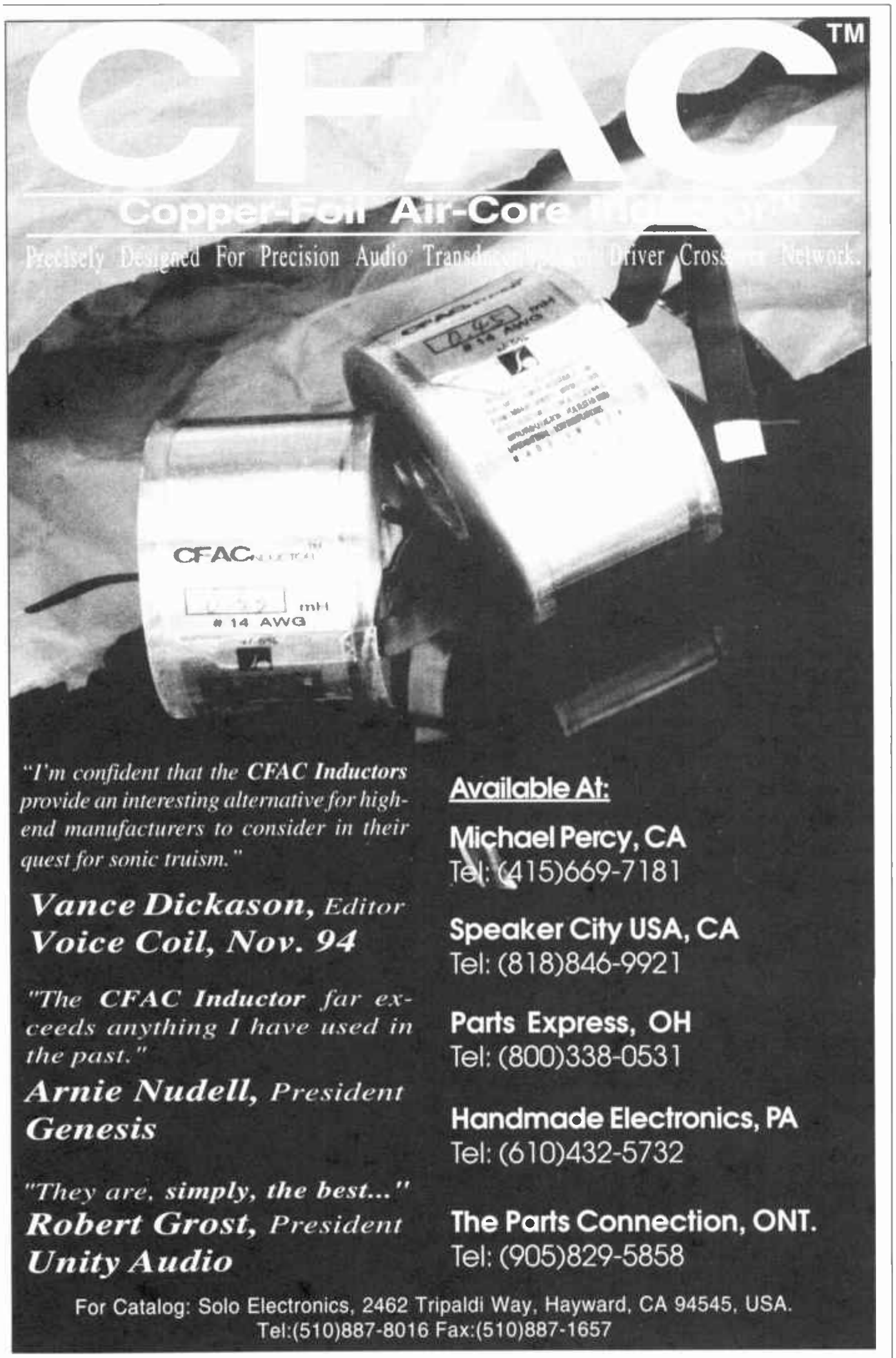
This program presents a surprisingly efficient mix of auto and manual crossover optimization features. It's easy enough to use for those starting to explore crossover configurations, yet provides all the necessary functions for the old hand. At $149, it is well worth the price. The ability to import IMP data files and to set the error calculation to "peak" values instead of average RMS (see above) would be welcome additions. Because of the sparse documentation (which should not be a problem for *SB* readers), a good text on crossover techniques would also be handy.

to page 59

SB *Mailbox*

AUDIO AND GENDER

It is my opinion that the pages of *Speaker Builder* have, over the past year or so at least, contained too many references to "Wife Acceptance Factor" and similar comments belittling women. Although these comments are clearly not meant in an overtly negative manner, they do little to advance the audio hobby, and nothing to promote increased acceptance of women in an overwhelmingly male-dominated field.



Reader Service #40

Furthermore, publishing articles containing these types of references to women shows acceptance of and perhaps concurrence with a rather juvenile writing style in your otherwise commendable magazine.

Speaker Builder is a one-of-a-kind publication. I advertise in it, and doubtless it provides good value in that respect. However, in the interest of providing not only a decent technical forum for speaker hobbyists, but also a positive contribution to the image of audio engineers and hobbyists, I believe it is long past time to look harder, much harder, at your articles before committing them to print. I welcome your reply, and still more would I welcome changes in your editorial policies.

Philip Jones
Owner, Driver Design

Ed Dell responds:

We have given a great deal of thought to the gender issues in the audiophile community. The dearth of women among audio enthusiasts, and the ignorant prejudice among many, especially sales persons, regarding the knowledge level of women, are deplorable and unwarranted.

The WAF term has at least two components: aesthetic and territorial. I dislike the term as an affront to males, since it often assumes that females are the exclusive keepers of aesthetic issues in the home. I don't see

OF NOTE IN
Audio Electronics

Issue 4, 1996

- A New Outboard DAC: Part 1
- The Dynamic Range Potential of the Phonograph
- Home Theater—A Playground for the Audio Amateur: Part 2
- Digital Projects for Musicians: Book Report

either gender as having a monopoly on good taste or design sense. Too often the enthusiast simply neglects the question of whether the new audio project is pleasing to look at.

The territorial matter seems part of the American idea of the marriage relationship where the living room is the province of the female. The level of control varies from the exclusive "my living room" to "our living room."

The root of the problem is in the word "acceptance." Women certainly love music as passionately as do men. But women are less often likely to be audiophiles. This raises a lot of questions. Is this another "guy thing?" Have men kept it that way, consciously or unconsciously? Women who try to get involved as audiophiles constantly run into the typical prejudice about females because they are supposed to be incapable of doing "mechanical or technical stuff."

I for one devoutly wish there were no such thing as WAF. That would mean that no audiophile ever wanted to put anything ugly in the living room. It would also mean that both spouses would attend concerts regularly, discuss music, listen to recorded music together and decide together whether some balance between visual and aural aesthetic could be worked out in the place where the family listens to music.

I cannot know what happened to the women who have become avid audiophiles, who love to solder, and get a thrill out of assembling a kit or building a speaker. But there are such people. We have published the work of one of them.[1] We will shortly publish another of her pieces on how to build a toy soldering iron. Her rationale: she'd like to raise a son who will grow up to be a speaker builder.

I believe a little thought and well-placed propaganda could result in recruiting many women as fellow enthusiasts. Failing that, perhaps your daughters and sons can be introduced to the pleasures of building a speaker with you—and eventually for themselves.

WAF is a symptom of a cultural problem being discussed more and more these days. As one author has put it men and women come from different planets. This is but another symptom of the unresolved value issues separating them. Music is surely one value that can bring men and women closer together. There is only one thing better than listening to good music, listening to it with someone. And better yet, with someone you love.

1. Nancy MacArthur, "Kit Report," SB 2/95, *p. 42.*

FIGURE CORRECTION

In Part 3 of "A Modest-Cost Three-Way Speaker System" (*SB* 8/96, p. 34), the graph in Fig. 42 is incorrect. The correct figure, showing acoustic response of enclosure B at 60″, with third-order crossover and Audax tweeter, is shown in *Fig. 1*.

AFTERSHOCK OBJECTIONS

I read Philip Abbate's article about his "Aftershock Subwoofer" (*SB* 6/96) and found a few problems with it. I modeled the enclosure using BassBox 5.1 and TopBox, both with the published specs and his mea-

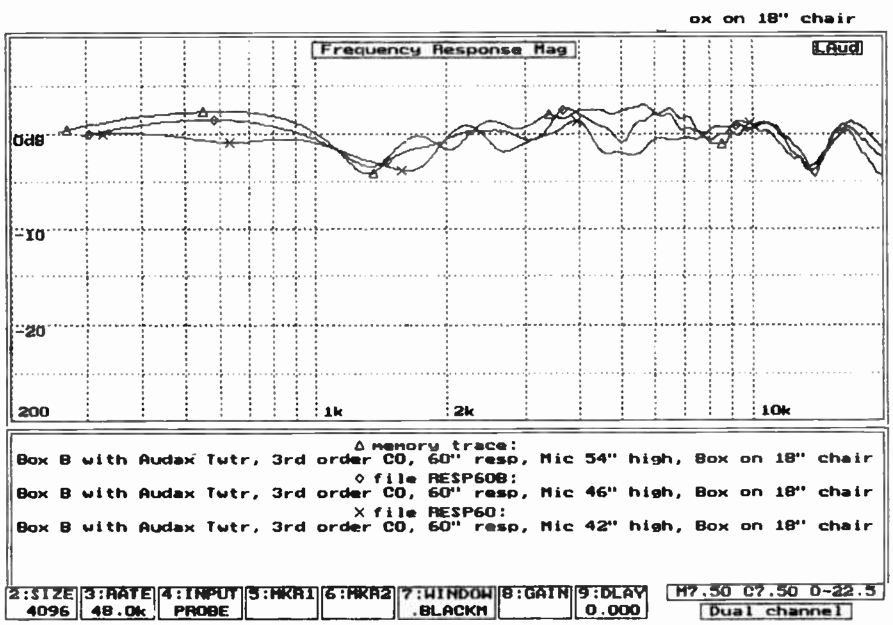


FIGURE 1: Corrected Fig. 42 for Koonce/Wright three-way speaker article.

sured specs, for the Altec 421-LF woofers. In both cases, I cannot see how the SPL or power capacity can be anywhere close to the article's claims.

Noting that Xmax is 1.3mm for the 421-8H and 421-LF (not the 13mm shown in Table 1 of the text), TopBox predicts the maximum excursion-limited input power to be 50.8W at the tuning frequency of 25.5Hz. Maximum SPL also peaks at this frequency as 102dB. At all other frequencies (assuming a sub range of 20–60Hz), the output is less than this, and the power capacity is less than 15W. This is disappointing for a cabinet of this size.

The problem is the Xmax for the 421 driver. The underhung coil sits in a very short gap in order to maximize efficiency at the expense of low-frequency capability. It was optimized for full-range response and horn-loading. The vintage Altec 411-A would be a better choice for bass-reflex subs, since its Xmax is about five times greater. Modern woofers, especially multiple 10″ or 12″ ones, would yield even better performance, due to their incredible strokes.

Regarding the differences in the measured parameters versus Altec's published specs, at first I figured that the magnets had lost some strength over time, which is normal (especially for woofers that were overdriven and blown). But then I recalled that the 421 woofers used ferrite magnets (not Alnico), which last virtually forever and cannot be degaussed in use. The differences in parameters are probably due to aftermarket voice coils, spiders and cones—those old Altecs have not been built in many years, and most original parts are likely long gone.

Matthew Honnert
Alumapro Metal-Cone Speakers
Addison, IL 60101

Philip Abbate responds:

Thank you for the opportunity to respond to Mr. Honnert's letter. I am delighted to have an objective technical critique, for one reason I published the memoir of this project was for validation. Speaker Builder *articles, Vance Dickason's book, some old test equipment, and my ears were the only resources I had on which to base my decisions when I designed and built Aftershock. Luckily, I did not realize that I was off by a misplaced decimal point in my conversion from inches to mm for Xmax when I ran the first Boxmodel plots. I may have erroneously concluded that the Altec 421-8Hs my friend gave me would not make a satisfying subwoofer.*

I spoke to Steve Upchurch at University (Altec) again to verify the Altec specs. It turns out that Altec used both ferrite and Alnico during the life of the product, and mine is ferrite. He was unable to help me with details on Xmax, but he told me how the Altec engineers designed the 421 series for rock-and-roll bass guitar while Small and Thiele were still in diapers. Their only criteria for the 421s were that they played loud, sounded good, and didn't break.

The specs Steve gave me were measured once, on only one speaker, to satisfy the curiosity of an engineer. According to Steve, the 421 specs, other than f_s*, were never published. I wondered myself if my measurements were in error, but now I suspect Altec's, too.*

My curiosity is piqued. I was just looking at the speaker cones moving way in excess of 0.05″ without distress or obvious distortion. I had a blown 421-A, which is identical to the 421-8H except for the basket shape, so I dissected it. I am not sure whether this speaker had original Altec parts because, although the voice-coil is edge wound, the wire is copper rather than the aluminum I recall in the 421s.

The top plate (gap) is 10mm and the voice coil windings are 8mm. The Xmax would therefore be 1mm. The geometry has a 10:1 gap-to-Xmax ratio. Dickason analyzes the Bl responses with respect to air gap-to-Xmax ratios in the first chapter of the Loudspeaker Design Cookbook. He concludes that the higher the ratio, the slower Bl drops off as the voice coil travels out of the gap. This may explain why, when I connect a 9V battery to the 421, the cone deflects ±5mm.

I put a second 9V battery in series with the speaker to see how much further the cone would move. Lo and behold, a static-like sound emanated. I suspect this noise is caused by the cone assembly free-wheeling out of the gap from inertia, and then springing back in, only to be pushed out again. The distance from the top of the pole piece to the bottom plate is 30mm, which may explain why I get such a long throw without bottoming out.

Mr. Honnert is right; the speaker can take only about 15W before the voice coil rides out of the gap and causes an excursion-limited distortion. The power is so low probably because of the 10dB, 20Hz, 6th-order equalizer. I am not surprised that the excursion limiting has not bothered me, because the meters on Aftershock's power amp rarely reach an average of 10W per channel with the system playing music at a nominal 90dB.

I listened to Aftershock alone with the amp peaking at 140W per channel as measured with a scope. On kick-drum and other percussive impulses, I heard a noise shifted to higher frequencies. This noise is masked when the rest of the system is playing. With the Isobarik configuration and Boxmodel, I am still confused as to what I should use for power input. If 15W, is it 15W per voice coil, or 7.5W per voice coil? I believe it is 15.

Since writing the article, I have built several more speaker projects and moved to Georgia, where there are no earthquakes. Aftershock is now the low end of a triamped dedicated stereo system located in a 15′ × 14′ listening room on the concrete floor of my basement. The crossover and amp remain the same, but the top-end breakpoint for Aftershock is now 40Hz. Frequencies above that are handled by a revived vintage JBL LE14, which crosses over electronically at 100Hz to a Focal 7V513, and then passively at 2.2kHz to a Focal T90TDX. Aftershock sits 4′ out in the room midway between the left and right speakers, with the back of the exposed Altec facing me.

I used the same method I detailed in the article to find a reinforcing node location. This system's output for the listening position is 1dB less than that of Aftershock for the listener in the same position. The Stereophile test CD2's lowest 1/3-octave warble tone still produces a reading of 108dB on my original unweighted Radio Shack sound-pressure-level (SPL) meter. Aftershock's acoustic and musical output has been anything but disappointing. My audiophile friends agree.

The seating area in the home theater is now about 15′ away and one floor removed from Aftershock. I have a 12″ powered sub in the theater room. When I want to wow my friends, I pipe the theater signal downstairs to Aftershock. The sensation upstairs is awesome. I still recommend that if you're inclined to revive an old stage speaker, you do a little experimenting similar to mine before you buy a new driver. You may be pleasantly surprised.

XVR-1 PC BOARDS AVAILABLE

Thank you again for the opportunity to share our work on the XVR-1 Electronic Crossover Platform (*SB* 2/96, p. 18).

Since publication of that article, we have received many notes about it from all over the world. We are happy to report that many have successfully constructed the XVR-1, either from the article or from the complete kit we offered. We even learned that XVR-1 is being used not only for home-audio biamping, but also for professional and live-sound applications.

Several readers have asked us about obtaining PC boards and other components to build XVR-1. Up until recently, we had few boards to spare, since they were all needed for the complete kit offering. We are now pleased to say that you may obtain the boards separately, and, due to economies of scale, we have reduced the cost of a complete XVR-1 kit. We hope that this will encourage more of you to build biamplified loudspeaker systems using XVR-1.

Readers interested in XVR-1 PC Boards, components, or complete XVR-1 kits should contact Audio Arts, RD 2, Box 3502, Wernersville, PA 19565, (610) 693-6740, or E-mail at audioarts@prodigy.com for more information.

Fred Janosky
XVR-1 Designer

DV12 WOOFER WIRING

I enjoyed tremendously Tom Perazella's article describing his subwoofer project ("True Bass," *SB* 5/96, p. 10). I also plan to use Audio Concepts' DV12 drivers in an Isobarik alignment, with a pair of drivers face-to-face, but I am unsure of the wiring that will make it come out as an 8Ω system. Please indicate the correct driver wiring per cabinet. And do continue writing about such excellent projects, as it makes my *Speaker Builder* subscription a terrific investment.

Bert Lopez
Wesley Hills, NY

Thomas Perazella responds:

Thanks for the kind words on my article, "True Bass." I'm glad you enjoyed it, for it has been one of my most rewarding projects.

When using the

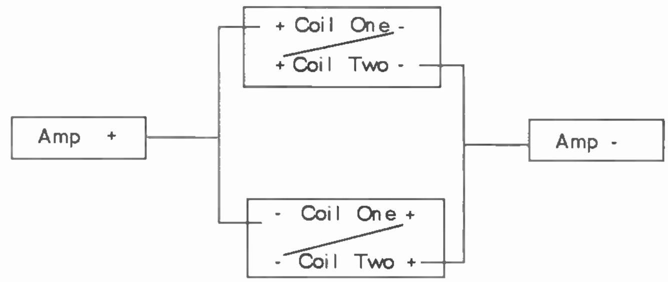


FIGURE 1: Wiring the two coils on each driver in series and then the drivers in parallel.

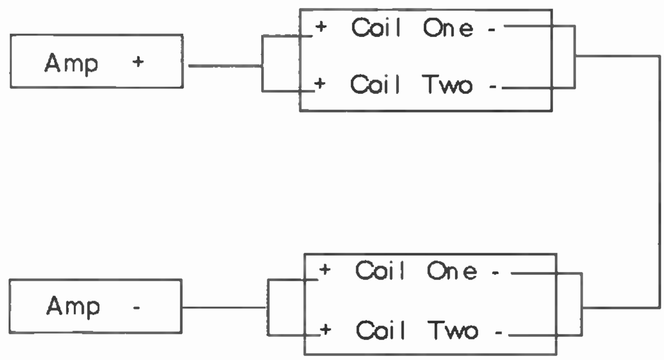


FIGURE 2: Wiring the two coils on each driver in parallel and then the drivers in series.

DV12 woofer, since it has two voice coils of 8Ω each, you have the choice of using the driver as either a 4Ω or 16Ω device. However, you can also leave one coil open, short one coil, put a variable resistor in series with one coil, or put a variable resistor in series between one coil and the amplifier, all with various effects on driver parameters. For this discussion, I will assume you use the more conventional series or parallel connections, and that you will not use the separate voice coils to sum two signals.

Your arrangement, with the two woofers facing each other, requires that you wire the drivers out of phase with each other. To achieve the 8Ω impedance you desire, you can 1) wire the two coils on each driver in series and then wire the two drivers in parallel (Fig. 1)*, or 2) wire the two coils on each driver in parallel and then wire the two drivers in series* (Fig. 2)*.*

Both the above methods will result in an 8Ω impedance. I prefer to use the first because I don't like to wire drivers in series. In the first case, although the coils are wired in series, they are physically tied together and behave as though they are one 16Ω coil. In the second case, the two drivers are wired in series to get an 8Ω impedance. The separate drivers will almost certainly have different characteristics and not behave the same at all frequencies and power levels.

Another way to look at the first case is to visualize two 16Ω drivers driven in parallel from an amplifier. Each driver is directly connected to the amplifier, with only the cable and amplifier-output impedance to interfere with driver control.

In the second case, you have two 4Ω drivers in series being driven by an amplifier. Each driver is connected to the amplifier with the cable, with amplifier-output impedance and the other driver impedance in series. Also remember that the driver impedance is nonlinear and that drivers act as both motors and generators, further complicating the issue of damping and driver control.

I hope this information helps you with your project. Good luck and have fun!

TAKING EXCEPTION

I read with great interest the "Tools, Tips, & Techniques" column by John Day (*SB* 6/96). Mr. Day has a number of interesting comments on wire and cable.

One statement he makes is on capacitive discharge. Capacitance between conductors is determined by the dielectric constant (DC) of the insulation material. While he claims that Teflon® is the lowest, that is not strictly true. Teflon® has the lowest dielectric constant of any plastic in its *solid* form, but there are other plastics which can be chemically foamed, or foamed by gas injection. Foam insulations have much lower dielectric constants, and lower capacitance.

Many Belden cables are made with foamed insulations. All our digital-audio cables, such as the 1800/A or our new flexible 1800F, use foamed insulations. And all our digital-video coax cables, such as the 1694A, 1505A, or 1865A, have foamed interiors. We have twisted-pair designs as low as 8.5pF/ft (part 9688). You can obtain a short sample by calling 1-800-BELDEN-1.

Mr. Day also talks about the impedance of speaker cable, and I believe that here, too, he may have gone astray. The critical nature of a cable's impedance is related to the wavelength of signal that travels over that cable. At very high frequencies (hundreds of megahertz), where wavelengths are short, the impedance is a critical number. But at audio frequencies, the wavelengths are extremely long. The electrical wavelength of 20kHz is 15,000 meters (48,750 feet).

This is so much longer than any cable you are likely to run that its impedance has no measurable effect. Only the phone company is concerned with these numbers, for it *does* run telephone lines many thousands of feet, and therefore needs to compensate for even a slight mismatch.

Impedance is a complex subject. The formula for impedance takes into account magnitude and phase of a signal. At audio frequencies, the phase is negligible, in effect making the magnitude components (i.e., the resistance) the impedance of the cable. But, of course, the resistances are cumulative. Eight ohms of resistance in your wire would cut your power to an 8Ω speaker in half. Not a very good deal.

The major effect in speaker cable is resistance, which has been well documented to affect a number of criteria, most importantly slew rate, also called "damping factor." In fact, the damping factor, which is the impedance of the speaker divided by the output impedance of the amplifier, can be dramatically affected by wire resistance. That's why big wire sounds better than small.

Finally, Mr. Day goes through a laborious exercise of combining plenum Category 5 conductors to "get to 8Ω impedance." Even if his technique was correct, you could use non-plenum wire, such as Belden 1583A, which is less costly. It has the same capacitance as the plenum Category 5. Both must meet the same industry specifications. Also, the impedance of a standard Category 5 (or any other category of cable) is 100Ω, not 75Ω, as the article stated. This is also an industry specification.

The article continues with a description of combining conductors to "get to 8Ω impedance." However, you cannot just combine wires to change impedance; it is not like combining resistors. I guarantee that the resulting capacitance (and inductance) of combined pairs will be some very different number from what you'd get by simply adding or dividing them.

The capacitance of a pair is determined by the space between the wires and the dielectric constant of the material used on them. But the space between pairs is air, and as the cable is bent, this space changes. So, in fact it is impossible to predict the resulting capacitance, and, therefore, the resulting impedance.

On the other hand, combining conductors and figuring out the equivalent gauge size is easy. Simply add up the circular mil area of the conductors involved. For 24 AWG Category 5 wires, this is 404 circular mils.

Eight of them are therefore 3232 circular mils, very close to 15 AWG (3260 circular mils) as Mr. Day claims. So, lest you think I offer nothing but negative suggestions, I do agree with that!

I think he's also a bit off in pricing. Our list price for 1,000′ of plenum Category 5 is $290. Non-plenum Category 5 is only $120 per 1,000′. Maybe your readers should be calling their local Belden distributor! If they can't find one in the phone book, they can check out our Distributor Locator on our Web Page, www.belden-wire.com.

Thanks for your always excellent magazine.

Stephen H. Lampen
Technology Development Manager
Belden Wire and Cable Co.
Richmond, IN

LOVE THAT BOOM!

I just received *SB* 7/96 and read the article by Charles Pike, "9Hz in a Barrel." What a wonderful and unusual idea he had! This is the sort of article I prefer to see in the magazine. I liked it so much that I will build one or two of these speakers for my media room, currently under construction. If we audiophiles put them in our cars, we could blow out those other drivers who are so proud of their "boomers." I am also interested in model railroading and have a Diesel sound generator that produces the sound of an Electro Motive SD-40 locomotive. Hmmmm, the possibilities!

Warren Bain
Bristow, VA

OPEN-BAFFLE SOLUTION

Some rooms are worse than others, but all have a devastating effect on the sound we hear from our hi-fi systems. If you have a test CD, just play the track containing the sine-wave sweep and listen to the enormous peaks and troughs in the sound level as the frequency rises from 20–350Hz, the range over which the room has the greatest effect. As well as this unevenness in the response, the peaks in the low frequencies have a masking effect on mid and high frequencies, causing a loss in clarity and imaging. And the response off-axis will likely be even worse, giving a change in the tonal balance you hear.

One partial solution is to listen in a room that is much larger, to lower the frequencies and the effect where the room problems appear. But the problems of the off-axis tonal balance will still occur with conventional speakers, because they radiate sound below about 300Hz, depending on baffle size. Another partial solution is to use miniature loudspeakers with very little low, mid, or bass output, and so avoid the problem altogether. That's what I do in the kitchen, where I use a little tranny, but that's not hi-fi. Unfortunately, we cannot make massive changes to our rooms, so we have to look elsewhere for a solution.

Fortunately, there is a solution to the problems of getting accurate reproduction in the normal-sized, conventionally constructed living rooms most of us live in. You need a speaker that radiates a lot less sound off-axis, to reduce the reverberant sound level and hence lower the effect of the room acoustics; a speaker that has the same response shape but is lower in level off-axis.

You can achieve all this with a properly designed open-baffle loudspeaker, that is, one having an open baffle down to the lowest frequencies. The reverberant sound diminishes to just one-third of the level of a conventional box speaker. You also lose the colorations caused by internal acoustic and box resonances. You gain clarity, openness, and better imaging, and, with well-recorded material, the speakers just disappear.

I can now play and enjoy recorded material with a lot of bass, and hear it not only at the correct level but also in every detail, down to the lowest notes without false emphasis. Sounds like magic, but it is true and confirmed by measurements. The in-room response using pink noise with third-octave analysis is very flat, without those nasty peaks and troughs that conventional speakers give.

I only wish I had discovered the properties of open-baffle loudspeakers seven years ago when I started to design, build, and develop speakers as a hobby. Then I could have avoided spending money on many of the 60 or so designs I constructed. On the other hand, I would not have gained all that experience in developing crossovers and finding the best ways of measuring the speakers I have built. I am so pleased with the sounds from the open-baffle loudspeakers that I think I will never again build any other kind.

This is not any pioneering effort on my part. The theory and the factors involved were all published in an AES paper by Siegfried Linkwitz entitled, "Development of a Compact Dipole Loudspeaker," AES reprint #3431(N-5), Oct. 1992. All I have done is to prove and endorse the work that Mr. Linkwitz did.

Tony Seaford
Burnley, Lancs., England

OF METAL DOMES

I read Ralph Gonzalez's "A Revised Two-

Way Minimonitor" (*SB* 3/94) with some interest, as I am considering renovating a dead pair of Smaller Advents and have a spare pair of Dynaudio 17W75 woofers. First, you did not actually contrast the sound of the Elac-equipped speakers with the original D28 setup. Granting that preferences are subjective and audio memory fallible, what do you believe is gained—or lost—by switching to a metal dome? Do you think these quality changes would be apparent with a less expensive metal dome? Have you any experience or anecdotes to relate regarding using the SEAS/MB/Vifa in this or a similar application?

Hugh Stevenson
Petaluma, CA

Ralph Gonzalez responds:

First a disclaimer: audio memory certainly is fallible, and it has been several years since I contrasted the Elac tweeter with the Dynaudio D28af in my minimonitors. Moreover, this comparison was rudimentary—no double-blind panel tests!

As I recall, the Elac tweeter sounded more detailed and "lush" than the D28. At low volume levels, the D28 seemed to make the overall sound too "tinny." I recall that D28afs have a mild peak around 12 or 14kHz, but on the other hand my subjective comments could also be attributable to different frequency-response interactions between the respective tweeters and the crossover.

I haven't done an extensive comparison between metal domes, and in particular, I'm not familiar with any drivers that are less than three years old. I obtained very nice results from the inexpensive SEAS 19-TAF/G (their ¾" metal dome using a phase plate instead of a protective screen). This driver had an exceptionally smooth and "high-end" sound quality. However, it may not be suitable for combining with the DYNA 17W75 woofer, because this woofer should be crossed over no higher than about 2.5kHz. That's a risky proposition for a ¾" tweeter unless you use a well-integrated high-slope crossover.

DISTORTION FACTORS

This letter concerns Roger Russell's article, "Quality Issues in Iron-Core Coils," (*SB* 6/96).

My interest was initially aroused because I supposed that he was going to show how the performance of a first-order, low-pass filter is influenced by the design of its inductor, and by power levels. Instead, he reported on the distortion products appearing at the resistor terminals, and asserted that this is caused by something he calls "coil distortion," as if it were an inherent characteristic of a coil, such as inductance, resistance, "Q," and so on.

A coil is, however, a passive element, incapable on its own of causing distortion. What it *may* do, when connected to an active element, is to cause this now active system to exhibit undesirable behavior. A fair conclusion is that this happens because the active element is intolerant of the passive element. This leads me to suggest that what Mr. Russell has demonstrated is that his amplifier is very load sensitive, much like many amplifiers, especially single-ended triode units.

Incidentally, the voltage required to deliver a given amount of power to the load resistor is not, as he states, the square root of the product of power times the resistance, because this implies that the full value of the amplifier voltage appears across the resistor. At 254.68Hz, for example, the 5mH inductor/8Ω resistor combination will produce a voltage across the resistor that is 0.707 times that at the amplifier terminals. From this, the voltage/power relationship can be shown to be the square root of twice the product of the power and resistance.

It would be interesting if Mr. Russell would present another set of curves illustrating the behavior of the first-order filter, more specifically, a plot showing the ratio of the fundamental of the voltage across the resistor to the fundamental of the voltage at the amplifier terminals versus various power levels.

David J. Meraner
Scotia, NY

Roger Russell responds:

Mr. Meraner seems to be questioning the internal impedance and distortion of the amplifier and its effect on the measurements. This is a good question. The damping factor for the amplifier I used is 40, which corresponds to an internal impedance of 0.2Ω. The amplifier distortion is rated at 0.005% at any power level from 250mW up to rated power. The quality of this amplifier greatly exceeds the minimum that is needed to make the coil-distortion tests. Most amplifiers will work well.

The voltage across the load resistor represents the current flowing in the circuit. To simplify, you can think of the amplifier as a generator with a series resistor of 0.2Ω prior to the output terminals. The same current flows through the amplifier, iron-core coil, load resistor, and hookup wire. Whatever array of fundamental and harmonics appears across the load resistor will also appear across the internal resistor in the amplifier, but at reduced amplitude. However, the iron-core coil in this circuit is the nonlinear element. If the amplifier had been sensitive to inductive loads, distortion would have shown up in the air-core-coil test shown in Fig. 15.

Mr. Meraner would like to see the same current in the load resistor at all frequencies, as if the amplifier were a constant-current source. However, most power amplifiers are excellent voltage sources. Perhaps the curves should be more clearly labeled "Amplifier Output Power Re: 8Ω."

In the crossover passband, about 95–99% of the amplifier voltage appears across the load resistor. The loss is due to crossover coil DCR. The same effect occurs at a woofer voice coil for a speaker system using a passive crossover.

At 250Hz, the reactance of the crossover coil is almost 8Ω, making a total load impedance of almost 16Ω. Less current is flowing compared to an 8Ω load. The same current reduction occurs in a woofer noise coil when a crossover coil is used. The current does not stay constant as frequency increases. By doing the tests this way, a better comparison can be made with what actually happens in a speaker system. I feel this is a more realistic and useful approach for the reader than simulating the conditions of a constant-current amplifier.

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EDITORIAL

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If response curves were made for a first-order network to show differences between the various coils, they would be very similar to the differences between coils G and H shown in Figure 19. This is a maximum of 0.5dB. Most of the curves would literally be on top of each other. If the load impedance is higher, as shown in Fig. 20, there will be essentially no noticeable loss (the DCR will be extremely small compared to the total impedance).

TURN OF THE SCREW

Bob Wayland's excellent article, "Screws and Other Fasteners" (*SB* 8/95), is one of the best I have seen in any consumer publication. I would, however, like to clarify three of his statements.

He states that "sheet-metal, or tapping screws, have about 10% greater withdrawal resistance than other types of the same size...." This is true of normal hardware-store–variety screws, but not for the deep "particle board" threads found on most Square Drive screws. In fact, tests prove the deep threads provide significantly greater withdrawal resistance than the Type-A tapping threads used on sheet-metal screws.

The rather casual mention that "using soap or glue—or even better, beeswax—to facilitate insertion has little effect on the withdrawal resistance" may lead readers to believe that soap is a suitable lubricant. Because soap is hygroscopic (attracts moisture), soaping screws can lead to severe corrosion and, perhaps, fastener failure. Beeswax, paraffin, or even non–silicon–containing paste waxes are vastly superior lubricants.

Finally, the comment that "a tapered drill produces a pilot hole that profiles the shape of wood screws, thus increasing their load resistance" is only partially true if the screws themselves are tapered. Today's manufacturing processes produce screws that are tapered only at the tip, with straight shanks. Using a straight, brad-point, or standard-point drill bit produces an ideal pilot hole for most modern screws.

These are perhaps nit-picky comments, but I believe the overall excellence of the article justifies these minor clarifications.

James C. Ray, President
McFeely's Square Drive Screws

Bob Wayland responds:

Mr. Ray makes some interesting points. The remark that his company's Square Drive screws' withdrawal resistance is higher than that of Type-A tapping screws may be true. Until I have seen results from an independent testing laboratory, I don't believe that I should recommend his screws to our readers over other brands.

The remarks about the hygroscopic properties of soap should be put in proper perspective. If the screws are made of bare steel, there is a possibility that some rusting might occur. However, there are a number of other considerations that the speaker builder should keep in mind. Most modern screws are coated with rust-resisting compounds or are made from non-rusting materials, e.g., stainless steel. The Encyclopedia of Wood *also recommends the use of soap. I suspect because it realizes the advantage of using soap far outweighs the small problems Mr. Ray mentions. Personally, I like a 50/50 mixture of tung oil and beeswax as a lubricant applied to the first third of the threads.*

Mr. Ray's remarks about tapered drill holes for tapered screws is misleading. Speaker builders usually buy screws from the local hardware store. My experience has been that, aside from some drywall and a few "particle board" specialty screws, the modern wood screws are tapered. If the screw you are using has a straight shank, of course you should use a straight-shank standard-pointed drill bit. I doubt that using a brad-pointed drill bit for the pilot hole will improve a screw's load resistance. My remarks are in no way a criticism of McFeely's Square Drive Screws, which I think are very high quality and deserve the speaker builder's consideration.

SEEKING ALTEC 412C ADVICE

I wonder if you have any information on the Altec 412C speaker. I am planning to use it with a Lineaum Tweeter like those in the Radio Shack LX5 speakers. If anyone has done such a project, I'd appreciate their advice.

Is there any chance of my using an Altec 515B as a woofer for that type of system? I understand that both drivers require a port, so it could be a complicated enclosure.

Please advise if you can assist me.

John Gresh
JSG <clash@redrose.net>

SPIKES

By now, everyone knows that spiking your speakers and/or stands can improve the sound quality of your system. Your speakers will sit more solidly and squarely on the floor and will not rock as they would on carpet. Commercial spikes will work, but speaker builders who like doing *everything* for themselves can make these with a minimal number of tools and materials.

For our purpose, a spike is a hard piece of material, usually metal, that is long enough to penetrate carpet or directly contact the surface on which the speaker rests. The spike can be a nail of sufficient diameter to support the load without bending. The nail is force fit, pressed into the "foot," cut, and filed flush on the top or back side. You can attach the foot to the stand with double-stick carpet tape (*Fig. 1*).

CONSTRUCTION

First, cut the feet from your choice of material approximately 1¼″ square and ¼″ to 3/8″ thick. The harder the material, the thinner the foot can be. For example, acrylic and aluminum can be ¼″ thick, while wood should be at least 3/8″. You'll need four feet (or maybe three if the base is large enough) for each stand and one extra for testing to get the right fit. Sanding or routing the wood edges is a nice touch, but not necessary.

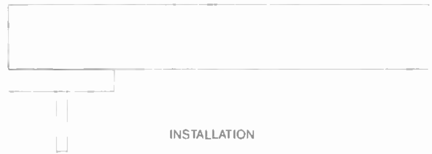


FIGURE 1: Spike installation for your speaker or stands.

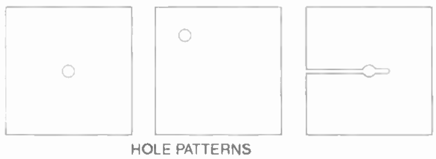


FIGURE 2: Hole patterns (centered, off-center, and slotted for aluminum).

Next, obtain some nails. For my 30-pound speaker/stand combination, I used tenpenny nails which proved strong enough. After the manufacturing process, the nail's point is slightly larger than the shaft, so it needs to be reduced and smoothed to obtain the proper fit. Otherwise, the hole will be too large and the spike will fit loosely. You might consider knocking off the sharp point of the nail, since sharpness is not a real concern for this high pressure area.

Drill a hole—with a drill press if available—through the center of the foot about 0.001–0.003″ smaller than the nail diameter. Or, you could drill your hole off-center close to a corner for maximum footprint size and more stability with small stands. But be sure to leave a distance of at least three diameters of the nail to any edge to maintain structural integrity.

The nail must fit tightly. My tenpenny nail measured 0.128″, so I used a 0.125″ drill bit. This worked fine for acrylic, but for aluminum I had to cut a slot starting from one side of the foot and continuing through the hole and a little past so the material could "spread" slightly when the nail was tapped through (*Fig. 2*).

INSTALLATION

To ensure the spike lengths are the same, use a piece of scrap wood the thickness of the length of the spike (in my case, ¾″). Drill a hole in the scrap piece a little larger than the spike diameter. Position the hole in the foot over the hole in the scrap and tap the nail through until it makes contact with the supporting surface, such as a drill press table. Cut off the excess nail close to the foot top and file flush. Or, instead of filing the nail flush, face it off using a lathe. Do this with all the pieces.

Stick a piece of double-stick carpet tape on the top of the foot. The foam-type tape may work, but the mating surfaces won't have solid contact. Trim as required using scissors or a knife.

Position and press the feet on the stands.

Flip the stands over and set them in place. You may need to rock or wiggle them a little so they penetrate the carpet and contact a solid surface.

With the variety of materials available and a little ingenuity, you can create something unique and functional. But best of all, it's homemade.

Ron Gallus
Vacaville, CA

THIRD-ORDER CROSSOVER FOR VMPS SUBWOOFER

When I changed the satellite speakers I used with my VMPS Smaller Subwoofers to Mach 1 Acoustics M-Twos, I found that there was too much overlap between the speakers to provide good imaging. The passive crossover supplied with the VMPS was billed by the manufacturer as a quasi-second-order design.

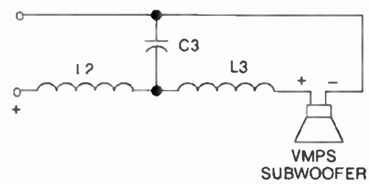
One immediate recourse was to use the M-Twos by themselves, retaining the VMPS as stands for them, but as the M-Twos lack the lowest octave, I went in quest of a fix to the VMPS so they would again be usable. I studied several books seeking information about crossover design, and found *The Loudspeaker Design Cookbook* (*LDC*)[1] to have the most complete information about third- and fourth-order network design.

THIRD-ORDER COMPROMISE

I selected a third-order network (18dB per octave) as the best compromise between a fast rolloff and maintaining sensitivity. The third-order network described in the *LDC* has a Butterworth filter response with a Q of 0.707. According to this reference, it is both a constant-power crossover (maintaining flat power transfer) and an all-pass crossover (maintaining flat frequency magnitude).

I chose to omit the original L-pad attenuator from the new VMPS crossover, as the sound without it had more clarity than with it, and I didn't think attenuation was needed with the M-Twos.

The *LDC* shows formulas for third-order components for a two-way network, but as this crossover was to be for one loudspeaker, and not a two- or three-way system, the values for capacitors C1 and C2 and inductor L1 were not needed. *Figure 1* shows the formulas and calculation results, as well as the schematic for the new crossover. I used some capacitors and inductors taken from the original VMPS crossover and purchased some others from Parts Express and Radio Shack.



f = 150
RL = 8

$$L2 = \frac{.2387\ RL}{f} = \frac{.2687\ (8)}{150} = .0127H = 12.7\ mH$$

$$L3 = \frac{.0796\ RL}{f} = \frac{.0796\ (8)}{150} = .0042H = 4.2\ mH$$

FIGURE 1: Schematic and calculations for third-order design.

I originally planned to fit the crossovers into metal kit boxes outside the enclosure, but as I accumulated parts, it became evident that the components' sizes precluded that plan. Instead, I mounted the parts to a piece of pegboard using silicone rubber and wire ties (*Photo 1*). Note that the inductors are installed at right angles to one another to minimize the likelihood of mutual inductance. This is addressed in detail by Mark Sanfilipo.[2]

I then mounted the pegboard, with components installed, to ¾″ plywood, using silicone rubber and drywall screws. I fastened the completed crossovers to the inside surface of the cabinet side walls, again using silicone rubber and drywall screws.

NEW BINDING POSTS

The original crossovers had been mounted on the inside surfaces of plastic inserts that also held the binding posts for connecting the speaker wires from the amplifier. To replace these, I cut two ¾″ 6″ × 9″ boards, drilled holes for the binding posts ¾″ on center, and installed new, longer posts. I fastened these new boards to the inside surfaces of the back

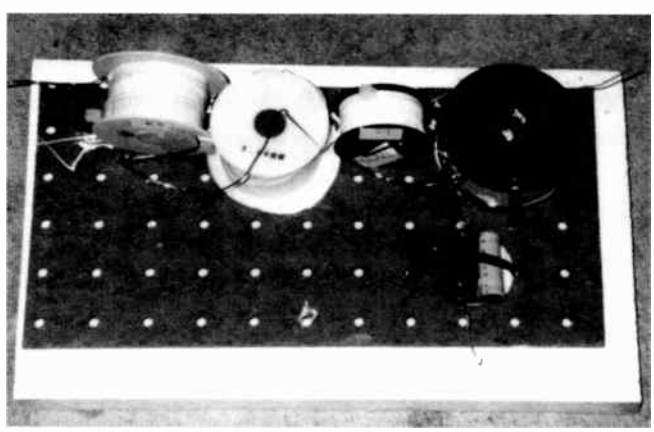


PHOTO 1: Crossover mounted on pegboard.

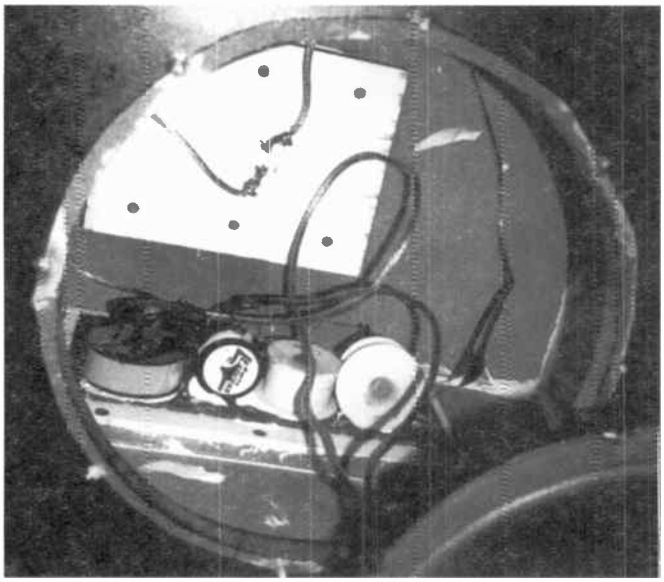


PHOTO 2: Crossover placement in cabinet.

walls of the cabinets, covering the openings in which the former plastic inserts were mounted (*Photo 2*). I applied a bead of silicone rubber around the peripheries of these openings and used drywall screws to fasten the boards to the cabinet back walls.

While the drivers were out of the cabinets, I also applied a bead of silicone rubber around the inside seams of the cabinets and around the passive radiator juncture to the cabinet (*Photo 3*). I did this to minimize the possibility of air leaks through the cabinet, which, when they exist, can degrade sound quality.

I wired the 12″ drivers to the new crossovers, using 12ga. stranded copper wire with clear plastic insulation, and then verified the polarity and functioning of the system using a 1.5V battery to apply voltage to the binding posts. This check ensured both electrical continuity of the circuits and that the systems had the same correct polarity, i.e., that the cones moved outward when a positive voltage was applied to the positive (red) binding posts.

I reinstalled the fiberglass batting I had removed, keeping it away from the passive radiators. I also plugged all of the speaker mounting screw holes in the front baffle with sections of match stick coated with yellow wood glue.

REINSTALLING THE DRIVERS

When I reinstalled the drivers, I applied a bead of Mortite™ flexible caulking rope to the sides of the driver flanges that would be in contact with the front baffle. When setting the drivers in place on the front baffle, I rotated them a few degrees from their former orientation so that I would have new baffle material into which to drive screws.

I installed the drivers using 1″ drywall screws, tightening them in a diametrically opposite pattern, going back and forth across

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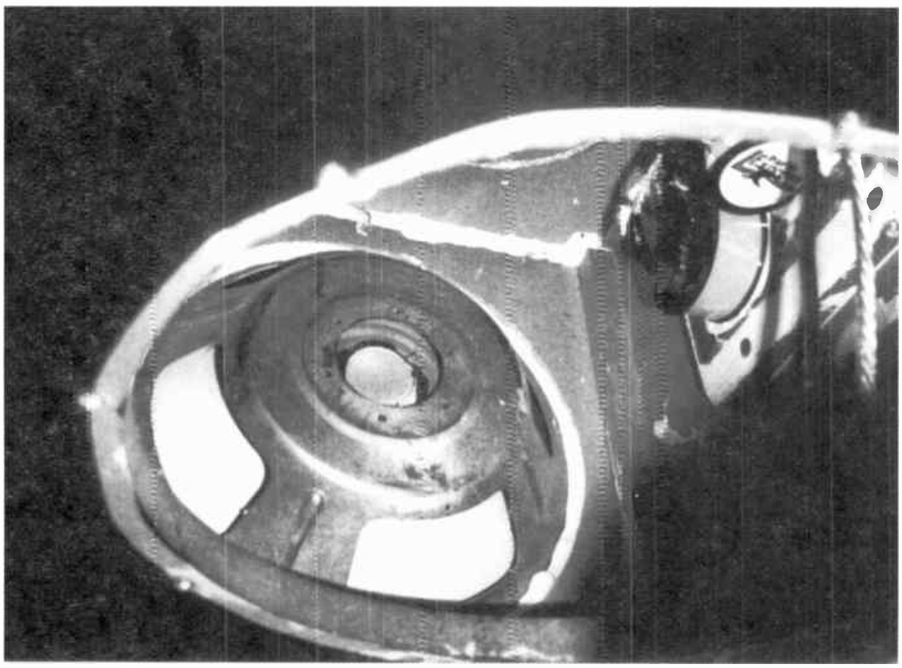

PHOTO 3: Passive radiator, crossover, and caulked cabinet seams.

Book Report

Build Your Own High-End Audio Equipment, compiled by Elektor Electronics, is available as part #BKEE38 for $29.95, plus $7 s/h in the US, from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458, (603) 924-6371, FAX (603) 924-9467.

In May of 1964, I bought my first copy of *Popular Electronics*, which in those days was *the* place to go for construction articles on all kinds of neat projects. In many ways it was the continuing historical journal for the art of electronics. November 1965's cover story, for example, was "Transistors Take Over Hi-Fi."

Later came construction articles for an electronic calculator (no gears or ratchets!) and the first personal computer sold in significant quantities. *PE* introduced the universally respected Ampzilla hi-fi amplifier as a construction project. And so it went, until people's interests gradually wandered elsewhere and the magazine was no longer a meeting place for electronic hobbyists.

It goes without saying that the Audio Amateur publications have taken up the audio end of the hobby. It seems that the *Elektor* magazine has laid claim to all other miscellaneous audio projects. A look at *Elektor*'s back issue listing uncovers all sorts of interesting topics, including hi-fi.

CONTENTS

Elektor's new book, *Build Your Own High-End Audio Equipment*, neatly compiles a series of audio construction projects. The cover includes those subjects that the basic audiophile would clearly like to know about: "...construction projects for solid-state and valve preamplifiers, active crossover filters, an active subwoofer, a mono/stereo compressor, and a headphone amplifier."

Perhaps, then, you would expect to find a nice discrete phono stage or line driver included? Or, certainly, a single-ended triode amp, since everyone's building one these days. And, of course, a pair of mini-monitor speakers, which are also quite common. Surprisingly, none of the above appears in the book; and, there are no tubes, either. Apparently, *Elektor* decided to eighty-six the "nasty" high-voltage stuff.

What you *will* find is a tidy collection of construction projects for what would be considered the exotic high reaches of the general-interest market. Where the standard audiophile routine seems to require stripping hardware designs to their most elemental form, this volume presents beautifully designed,

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thoroughly thought-out, fully featured apparatus, which generally use op amps and the occasional integrated power-amplifier modules. There are, for instance, three preamp projects, starting simply and expanding at each subsequent level.

Preamp I is a stripped version with a rather nice jack panel/circuit board/rotary switch assembly. Long shafts on the input selector switches run to, and through, the front panel. This is a board I would use. Preamp II adds a phono stage with several switchable cartridge loads, multiple dual-input transistors with LED-controlled current-source bias, and a DC servo. Inputs are

AD INDEX

relay-selected, and the power supply has extra transistors to improve the performance of the already passable LM 317/337 regulators. Preamp III is even more intricate.

GADGETS GALORE

The reader is encouraged to use high-quality parts, and these projects require the use of many. Op-amp types are discussed as they relate to the various positions and functions in the circuit. This displays a level of awareness that is not universal among designers, and provides good insights regarding the processes involved in these choices. By the time you come to Preamp III, *Elektor* considers the AD712 op amp, which was praised not too long ago by audio heavyweights Walt Jung, John Curl, and Richard Schram (*Stereophile*, September 1994, p. 189).

The power-amplifier section contains some promising circuits, including a sleek and trim 60W MOSFET. In Power Amp II, the circuit is upgraded to a 90W IGBT configuration. The top-of-the-line is a 200-Watter using bipolar outputs and elaborate protection circuitry. A mono chassis for this amplifier contains four circuit boards. If you build a pair of these machines, you've really accomplished something.

The book continues with headphone amps with three circuits, each optimized for a particular interest: niftiness (pushbutton volume and balance controls), audiophilia (a mini power amplifier with very upscale specs), or cuteness (a circuit board barely bigger than the input and output jacks mounted thereon). There is discussion about Dolby surround-sound and a project surround-sound decoder with integrated power amplifiers for center and surround channels. An active three-way speaker system using an internal electronic crossover and a combination of discrete and integrated amplifiers is presented. It includes a Linkwitz "bass correction network" with equations for your adaptation elsewhere.

Loudspeaker driver specifications for this vented cabinet consist of "Woofer, 100W, 200mm, 4Ω"—a rather scant set of specs for an audiophile project, but par for the majority of happy amateurs who do put speakers into boxes.

WORTH A LOOK

More promising, perhaps, for those craving precision, is an "Active Phase-Linear Crossover Network." While it uses 13 op amps per channel, the waveforms in scope photos are certainly in-phase. Nonetheless, some summed-pulse or square-wave pictures should have been provided. The "Active Mini Subwoofer" is a pretty involved project, but it looks like a welcome addition to the many systems currently using mini-speakers.

Although the loudspeaker driver make and model are not specified, a set of driver parameters assists you in choosing a proper woofer. The book closes with a discussion of hardware and software test instruments and the circuit board patterns for the book's projects.

So, finally, is the book worth it? I'd say so. It contains some circuits you'd probably like to see, and some very useful circuit-board functions to incorporate into your own projects. It is worth mentioning that the book is beautifully made, with clear print, bright paper, supple binding, and lots of photos of very nicely made gear. Even if your tastes run to more esoteric projects, it is worthwhile looking at comprehensively designed home projects that resemble top-notch commercial equipment.

Phase Distortion

from page 20

enough to make the door to the room vibrate. But I have a pair of Linaeum minispeakers with their highly acclaimed 360° ribbon tweeter and first-order crossover. With violins and some horns, I think they sound a little better than my home-builts. (Don't you hate that?)

For the ultimate, I guess we'll have to wait for someone to invent a perfectly continuous 3′ spherical electrostatic!

NETCALC

from page 41

MANUFACTURER'S RESPONSE

Thank you for a thorough review, which gives a good idea of the concept behind the program.

Please note that manual input of driver responses through the graphics equalizer screen is also possible in 1/6-octave increments with the "double resolution" option [not documented in the manual—JD]. Also, after looking at IMP data files, I am pleased to report that the US version will have the ability to import IMP files.

Finally, the suggestion to do the optimization relative to maximum error values instead of average RMS is an interesting one, which we will consider for possible future implementation. It should be noted that the maximum error values can still be determined by visual inspection of the graphs.

Rien den Boer
ARIBA Software and Hardware

Tools, Tips & Techniques

from page 55

he driver. I tightened all of the screws until the force they applied was even, evidenced by both the feel of torque and the driver flange being parallel to the baffle surface.

Those tasks complete, I was ready for a listen and for some measurements. I collected the revised subwoofers in parallel with the satellite speakers (for a resultant nominal impedance of 4Ω). The sound was much improved, as the subwoofers' energy was spread over a smaller range, with most of the sound coming from the satellites. The subwoofers were evident only (as they should be) where low bass content was present.

MEASUREMENTS

I placed a Radio Shack digital sound-level meter 1′ in front of one subwoofer's driver, on its centerline, and used 1/3-octave center-channel frequencies from the Carver Corporation & GRP Records' "Amazing Bytes" CD (GRP-Z-9907) as the signal source. I turned the other amplifier channel off during the test. While the 25Hz level was the same, the 31.5–63Hz levels were 2–3dB higher with the modified crossovers. Above 125Hz, the levels of the modified VMPS decreased more rapidly than did the stock unit, being down 3–4dB at 400Hz and high frequencies.

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1. Dickason, Vance, *The Loudspeaker Design Cookbook*, Fifth Edition, Audio Amateur Corp., 1995.

2. Sanfilipo, Mark, "Inductor Coil Crosstalk," *Speaker Builder* 7/94, p. 14.

I found the calculating, designing, and building of this crossover modification to be a lot of fun, and the end result was certainly worth the effort with my system.

James T. Frane
Orinda, CA

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Pair Lowther Accousta 90s, mint, $850; pair Tannoy Berkeley 15s, mint, $1,750; pair Tannoy DMT System 12, excellent, $1,500; pair Tannoy SRMIS, near mint, $1,000; pair Tannoy DMT8, mint, $500; pair AR9LS, mint, $1,200; pair JBL C-37, mint, $750; JBL D38 (single) excellent, $300; pair Boston Acoustics A-150, mint, $400; pair KLH6, excellent, $350. Heerlein, (314) 579-0088.

Pair Altec 515E-16 (recommended in Stronczer/*Sound Practices* article), $450; pair Altec 288KA, $450; 828 cabinets, 805B horns; pair Altec 421-8LF for "Aftershock" subwoofer, $375; pair JBL 2425 drivers, $275; pair 2344 bi-radial studio monitor horns, $175; pair Community PC494 2" horns, $200. Kent Elliott, phone/FAX (913) 677-1824, E-mail KentElltt@aol.com.

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The Audio Revolution™
An informational magazine on the Internet designed for audio enthusiasts.
http://www.audiorevolution.com

B+K Precision Catalog
Engineering specification guide of the company's test instruments.
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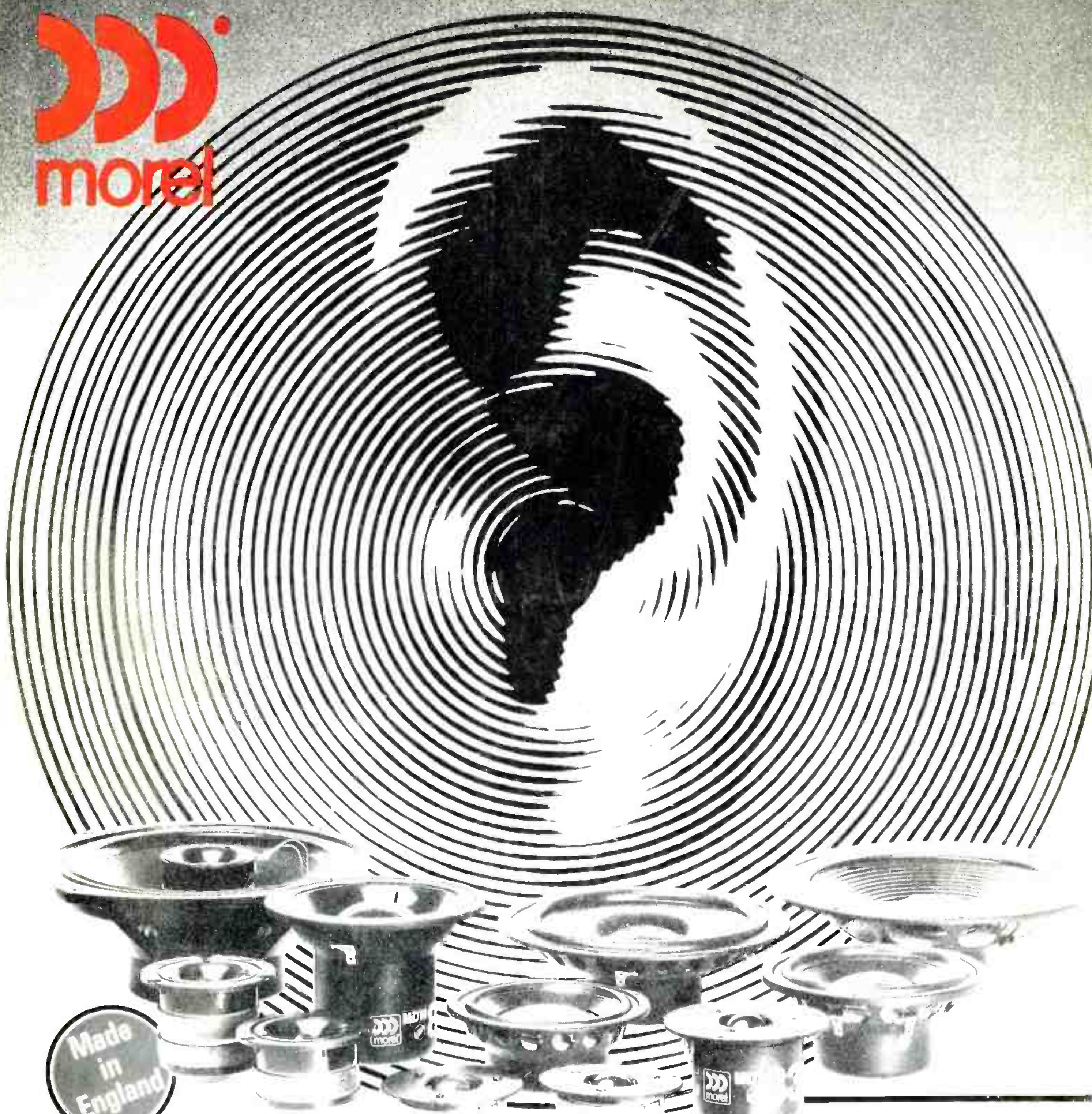

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