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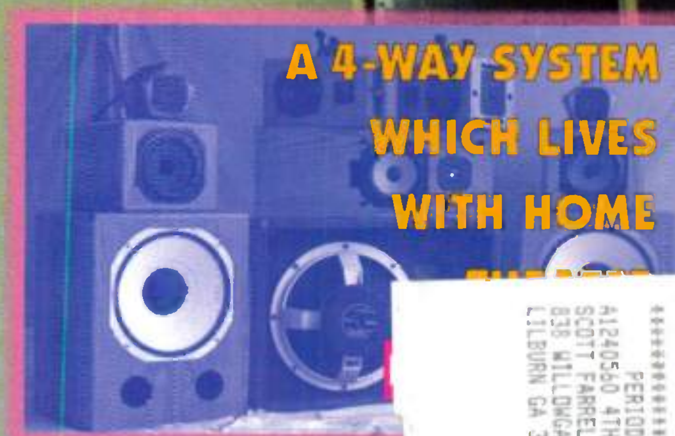

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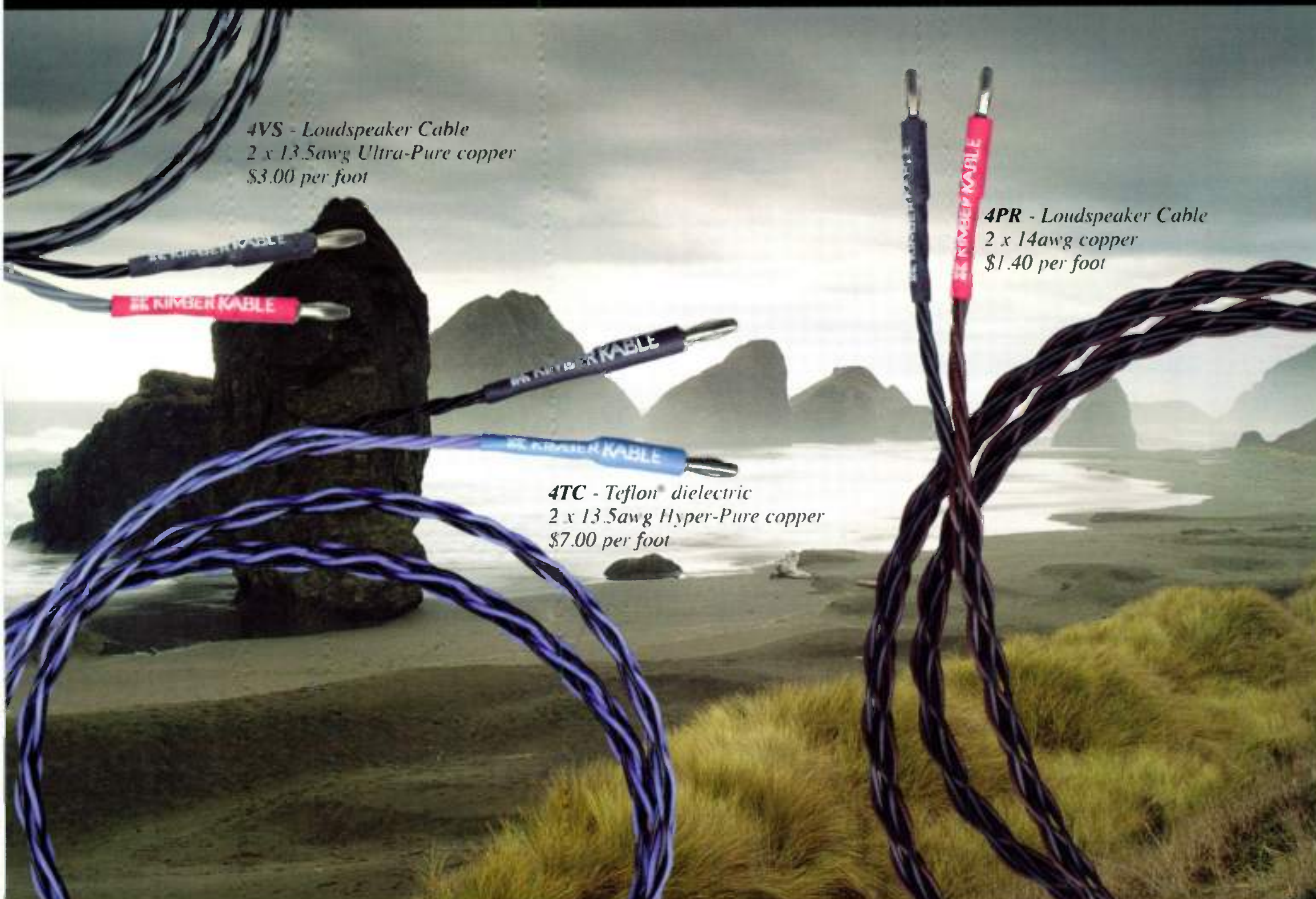

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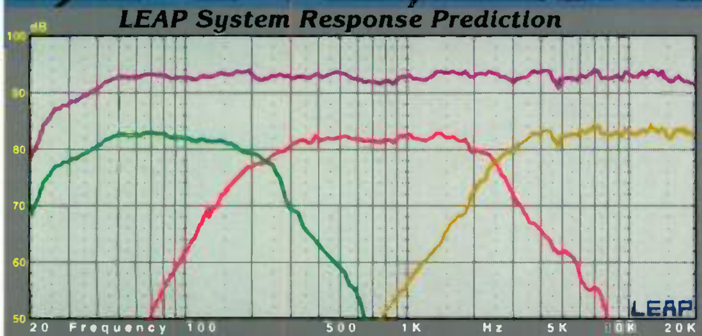

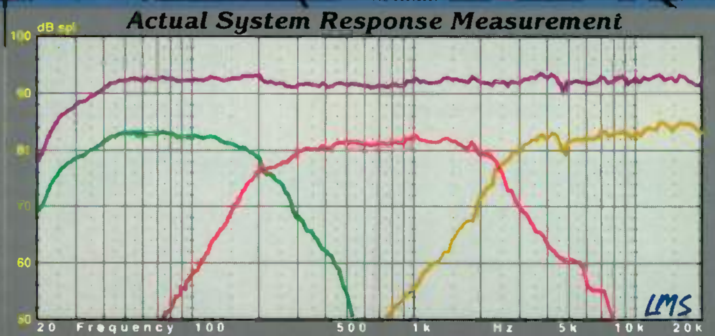

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■ A MUST-READ

Introduction to Loudspeaker Design, by John L. Murphy, was written for hobbyists, technicians, and engineers seeking an introductory overview of the technology of loudspeakers. Starting with a brief history of audio developments, the book begins by introducing the concepts of frequency, pitch, and loudness. In addition to discussing closed- and vented-type loudspeaker enclosures, the book covers bandpass enclosures and includes a complete design for a dipole loudspeaker. It also deals with advanced topics, such as loudspeaker design tradeoffs, spatial loading, diffraction loss, cavity effect, enclosure construction, and crossover design. True Audio, 349 W. Felicita Ave., Ste. 122, Escondido, CA 92025, (760) 480-8961, (800) 621-4411, Websites www.trueaudio.com, www.amazon.com.

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About This Issue

In his continuing quest for accuracy in sound reproduction, **Bill Waslo** experiments with the focused-array design concept—this time with a loudspeaker using electrostatic elements ("FAE—Focused-Array Electrostatic," p. 8). The author invites readers to improve upon his unusual design. Any takers?

Philip Abbate admits that his many modifications to improve his audio setup may be irksome to some. But, judging from his latest configuration, we'd have to say his efforts have paid off in an earth-shattering way ("The Seismic Stack System," p. 18).

When it comes to motivation, we were impressed with the energy and enthusiasm of the **Tim Sandrik**-led group from Michigan Technological University in tackling a dipole design as part of a college research, design, and development project ("Designing the Dipole Monster, Part 1," p. 26). Aided by some strong support from the industry, these collegians watched their original simple plans develop into a monstrous, but successful, dipole design.

Felt, foam, fiberglass? What's the best material to dress up your front panel? Don't go by aesthetics alone. Covering your speaker's front panel is not simply window dressing; the materials you choose can damp the effects of delayed echoes added to the signal. **G.R. Koonce** measures the effects of various materials to determine which one is best ("Testing Front-Panel Damping Materials," p. 32).

Do you need help with the proper setup of your unit in the home? **Bill Waslo** reviews the TACT RCS 2.2 Digital Room Correction System to determine how this system can help you with speaker placement ("Product Review," p. 42).

Industry insiders **Mike Klasco** and **Steve Tatarunis** reveal why speaker manufacturers are turning to ferrofluids as heat conductors to keep units cool ("Trade Secrets," p. 52).

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JOHN STUART MILL

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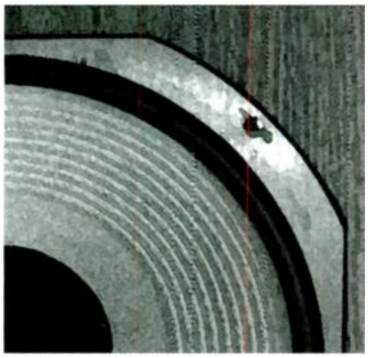

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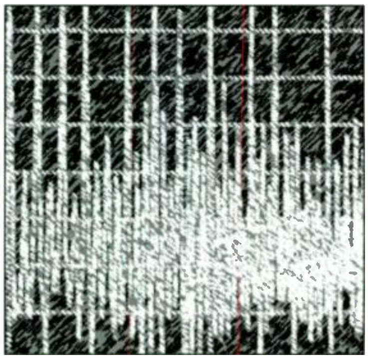

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DEPARTMENTS

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Part 1

FAE—FOCUSED-ARRAY ELECTROSTATIC

By Bill Waslo

I named this very unusual loudspeaker system "FAE," which stands for "Focused Array Electrostatic." The "Focused Array" part refers to a loudspeaker design concept in which I became interested several years ago (see *SB* 4/95, "Focused Arrays," and *SB* 6/95, "Testing a Focused Array"). The "electrostatic" portion refers, of course, to the transducer principle of (most of) the loudspeaker. The frame and electrostatic elements of the focused array system are shown in *Photo 1*.

A major goal of this design was to be able to generate more accurate representations, at the listening position, of the audio signals which were originally acting on the recording microphones during the actual musical performances. In short, I wished to produce an acoustical illusion for the listener of being transported to the original recording locations—as opposed to an attempted simulation of the original performers appearing in the listening room.

I recognize that either approach (listener in original hall, or performers in your home) is valid, and listeners may prefer either (or neither), depending on a given recording or mood. But I consider the more enticing of these possibilities to be the former, which is also more related to accuracy in recorded sound—a symphony orchestra has never been in, nor could it feasibly be presented as, performing in my listening room! I hoped to determine by this design whether this kind of accuracy in sound reproduction is a worthwhile goal, given two-channel recordings and existing recording practices.

My primary approach was to try to minimize the effects of near reflections and echoes, which form a signature sound of the listening room, by maximizing the ratio of direct sound relative to the reflected sound. By this approach, I assumed the use of original recordings already containing reflected sound components of the original recording venue (usually quite different in nature from those sound components that might result from speakers in a typical listening room).

I also assumed that these spatial components are best reproduced by minimizing contamination from the acoustics of the listening room. I had to give practicality a relatively low priority in this design. My primary interest was to attempt an experimental system for the accurate two-channel reproduction of recordings rather than to develop a commercial product or home-speaker project for general use.

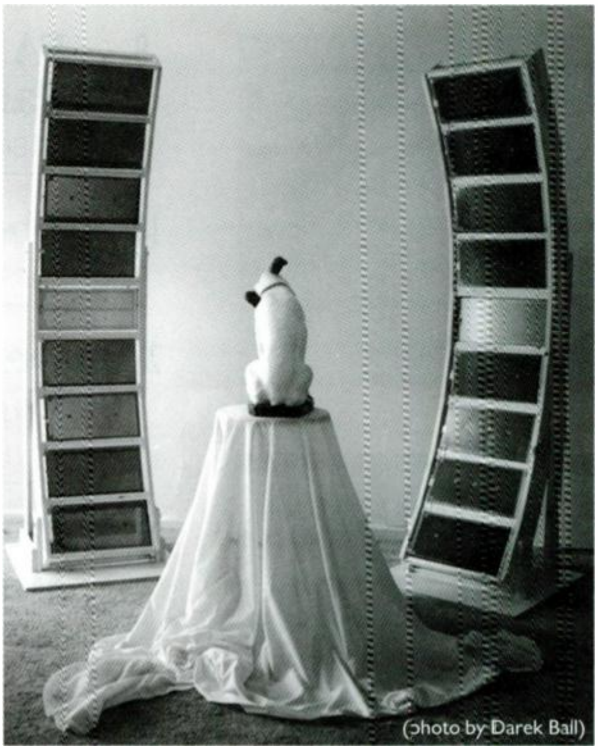

PHOTO 1: Nipper critiques the focused-array electrostatic.

FOCUSED-ARRAY CONCEPT

I previously presented the basic concept of the Focused Array loudspeaker, but will briefly discuss it again for those who may have missed the earlier articles as well as to expand on a few characteristics or limitations which perhaps were not emphasized sufficiently last time.

An excellent demonstration of the principle can be found in the COSI museum (Columbus, OH), where two parabolic reflectors are set far apart but aimed toward each other within the building (*Fig. 1*). You can climb a ladder to put yourself at the focus of one of the reflectors and converse in a normal voice to someone situated far away at the other reflector. The voices are directed and recollected by the curved reflectors, giving startling clarity and good volume.

But if you move away from the focus, the other person's voice becomes inaudible. The transmitted sound components, when combined by the reflectors so they arrive coherently in phase, reinforce each other. Away from the focus, where they can combine only noncoherently, the sound components are lost in the background clutter. In effect, the near acoustic environment at the focus of each reflector is transferred to the other reflector.

A focused array is a set of numerous similar drivers arranged such that the wave fronts radiating from them arrive in phase and constructively combine at a specified listening position. The simplest way to provide for this situation is to have all the drivers driven in common and placed at an equal distance from the listener. This scheme operates similarly to the way a concave mirror or satellite dish antenna focuses light or microwave signals. As opposed to a point source (for which the signal is radiated equally in all directions) or a line

FIGURE 1: Operation of the parabolic sound reflectors at the COSI museum.

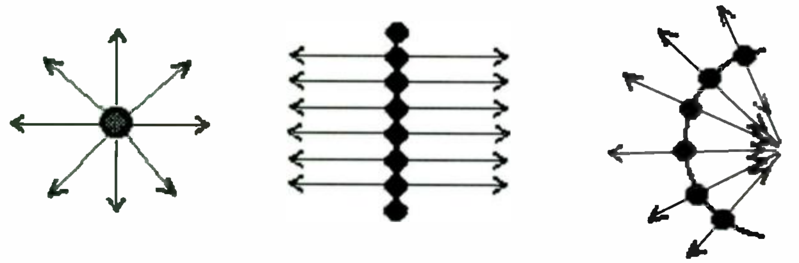

FIGURE 2: Directivity or pressure patterns for three speaker types.

array or planar source (for which the signal is concentrated into preferred directions in one or two dimensions), a focused array generates a broadband intensity peak at a single region in three-dimensional space (*Fig. 2*).

With the Focused Array, there is not so much a radiation pattern or a "beam" that projects sound in a certain direction (where the level changes, but the response curve is constant at any distance). Rather, a concentration of sound at a point occurs—when using a focused array, the listening distance from the drivers is also important for proper frequency response. Another way of thinking of the Focused Array is that it is designed for use in the near field (the array is very spread out relative to the listener's distance from any part of it).

At first glance, you might assume that the "point source" characteristic is the ideal situation for a loudspeaker. After all, doesn't that give equal radiation toward the listener wherever he may be? And doesn't that seem like the most "transparent" situation—a sound emanating from a point in space toward all directions?

The problem is that sound waves which are projecting out do not just stop when they arrive at boundaries (walls, ceilings, floors) or objects in the room. They bounce off with only slight attenuation and will eventually find their way back to the listener, delayed in time and filtered in their spectrum, to interfere with the waves from the direct path so that the received signal becomes extremely irregular in frequency-response variation and confused and imprinted with characteristics imparted by the listening room (*Fig. 3*).

You can imagine how distracting it might be to view a theatrical performance in a small room with mirrors for its walls, floor, and ceiling. A similar thing happens when generating a sound field in a typical listening room with omnidirectional or predominantly reflecting loudspeakers. The effect may be interesting or even pleasant, but it has little or nothing to do with the recorded sound you are trying to reproduce. It is very much like having a reverb unit connected to your system, but one which you cannot switch to "bypass."

The omnidirectional characteristic is not ideal even if you are attempting to create a "performers in my listening room" effect. The reflected-sound characteristics are those of the room, but only of the specific speaker location within the room. The sounds from the speaker (or of a performer in your listening room) will change, and quite dramatically, depending on distance from the room boundaries. This can be easily demonstrated by auditioning a loudspeaker in different locations, or just by listening to your own voice as you move around the room and near or away from the room surfaces. The induced effect of the room can be as if all the instruments and performers were concentrated at that single speaker location.

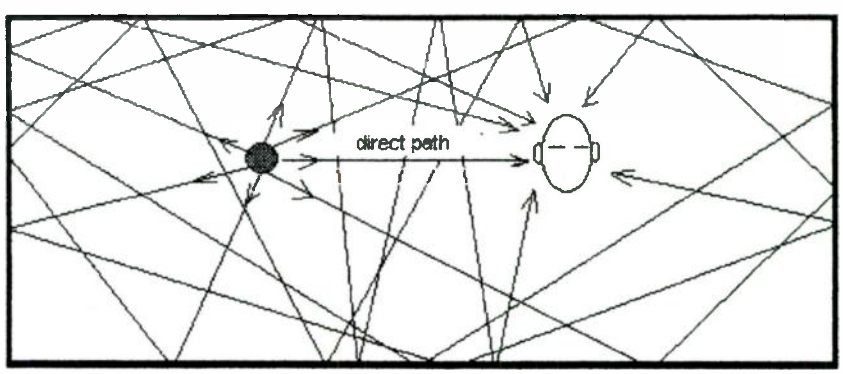

FIGURE 3: The direct path and a few of the reflected paths of sound going from a point source toward a listener.

A BETTER WAY

The planar source or the line source or array restricts radiation of the sound toward preferential directions. This can be quite effective in minimizing room effects, as fans of electrostatic speakers, ribbon drivers, or of multidriver towers will attest. These systems have reduced vertical (up/down) radiation because of combination effects. Pressure components from the separate drivers or from distributed radiating regions of the long single-driver surface combine in-phase (to constructively reinforce each other) in the forward or backward direction, but in staggered or smeared phase (because of different path lengths) in the vertical directions.

A dipole driver, in addition to directivity due to its length, also has very steep radiation nulls to either side because the front surface produces pressure which is the opposite of that from the back surface. At the sides, these opposite pressures exactly cancel, so there is no net pressure or radiation in these directions. This is very effective at preventing first reflections from the side walls. A consequence of obtaining directivity by these methods is that the optimum listening position becomes smaller—the sweet spot becomes sweeter, but more closely defined.

The radiation from the planar drivers, at mid and high frequencies, tends to move forward somewhat like a "plane wave." Again, only a small portion comes directly to the listener's ear; most of it goes past him to reflect from the walls or objects behind him. The dipole type also radiates equally and oppositely in a direction away from the listener (toward the wall behind the speakers) and most of that energy will also reflect back toward the listener after a delay.

As compared to a point-source driver, these speaker types generate more distinct but generally more delayed reflections. The greater delay helps the listener to isolate the perception of the direct sound, and it minimizes broad variations in frequency response in the midrange region which would result were there stronger near reflections from the side walls, floor, or ceiling. A side effect of the greater "coherence" of the reflections is the creation of phantom images, in which a listener at some locations in the room will sometimes hear what appears to be the sound emanating in whole from strange places in the room.

If using a planar driver can be considered as making a tradeoff from the number of usable listening positions in favor of better quality at the remaining positions, the focused array might be considered as taking

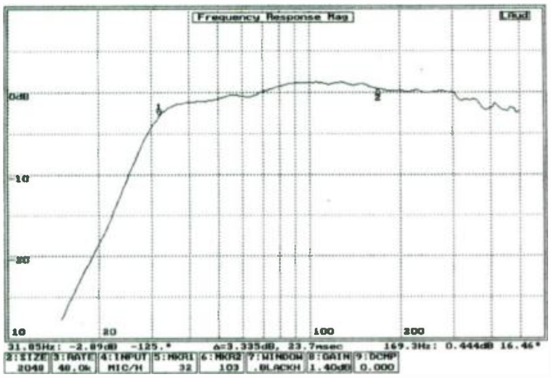

FIGURE 4: Combined cone and port near-field response of sixth-order woofer. Response is measured relative to the input signal to the woofer high-pass equalizer.

that tradeoff to an extreme. There is one listening position (or at most two or three with some compromise), and all sound quality elsewhere in the room is sacrificed or ignored toward optimizing for the choice seat. I do not consider this a problem for solitary "serious" listening. Most audiophiles, after spending many hours and/or dollars on a system, always sit in the "sweet spot," anyway. But Focused Arrays are not appropriate for use as party speakers, nor for casual listening as you putter around the house.

In the FAE design presented here, the individual drivers for the midrange and tweeter are planar or curved planar electrostats, to give further reduction of radiation to the sides of the speaker. Rather than using entire circular cup-shapes such as the parabolic reflectors at COSI, I made the FAE arrays as simple curved slices—I needed to be able to fit them into a room. And while arranging the slices to be inclined would have afforded better echo minimization, my listening setup did not afford that luxury.

Each channel features eight midrange electrostatic panels, each about 7″ × 14.5″. The forward waves from the midrange panels concentrate for increased intensity at the listening position, but the back waves are splayed outward, so that the arrivals of their eventual reflections back to the listener are randomized and thus reduced relative to those from the coherent direct paths. The reflections that occur are diffuse and dominate at most locations in the room, but are comparatively much lower than the direct energy at the target listening position.

DESIGN CONSIDERATIONS

Of course, there are tradeoffs and limitations with the Focused Array and with the electrostatic loudspeaker driver, which I considered when making the design tradeoffs (or later, when making fixes after problems were uncovered). Although I intend to include sufficient detail for a builder to make a system like this, I am not writing this as a "Build This Speaker!" article. I hope that readers use portions or principles of this system as elements in one of their own designs.

One major consideration which may not be obvious from the discussions above or from previous articles is that the focused-array effect is quite frequency dependent. While "same distance" means "same" at all frequencies, the effectiveness of the off-focus signal randomization is reduced as frequency decreases. This has two consequences.

For very high frequencies, the principle works way too well—the sound changes dramatically (between good sounding and dead sounding) for head movements on the order of less than an inch! To solve this, I used a separate single panel for the high frequencies above 3kHz. But even this still gave too limited a lateral listening region for the treble frequencies, for the panel I used was only 14″ wide.

After discovering this problem, I substituted slightly curved tweeter panels to spread the high-frequency energy across several feet around the listening position. Fortunately, the higher frequencies are usually more attenuated when reflected, so that echo reduction is probably needed less at these frequencies, and can be accomplished by use of absorbing surfaces in the room.

At low frequencies, the difference between a focused array and a line array begins to disappear. For example, for the end panels in the six-foot FAE frames, the distance to the listener is only about 10″ different than if the array was constructed in a straight line. At 100Hz, this corresponds to a path difference of only 0.074 wavelengths, so the curvature of the panel contributes little in this frequency range or below. While this factor is unfortunate in terms of reflection control, it provides a bit of freedom in fixing a problem with the electrostatic panels (i.e., poor bass output capability).

BASS

Due to cancellation, the dipole radiators have a natural frequency-response characteristic at low frequencies (and when still above the diaphragm resonance) of 6dB per octave. To obtain a flat response, the electrostats must somehow be equalized, which requires greater drive and much more diaphragm excursion at lower frequencies, putting a very real limit on how loud an ESL will play without extreme distortion. Below the diaphragm resonance, even greater drive would be required if a flat response were to be maintained. And at the high-Q resonance frequency itself, the panel cannot handle even moderate drive without slapping.

For the panels I made (using David Lucas foam dielectric and heat-stretched Mylar diaphragms), the resonance point was at about 90–100Hz. For a while, I used the curved arrays full-range, which worked better than you might expect. I was surprised how effective the apparent bass could sometimes seem (a bit of room lift from judicious placement relative to the back wall helped here). But high-level program material with any considerable real bass (below 100Hz) would cause a noticeable hashiness in the sound, when not making outright pops and snaps from the ESL elements. Use of an electronic crossover and separate boxed dynamic cone drivers, arranged in a tall line array, allowed for very good bass performance and high output level, without really losing much of the curved array's performance.

Adding the bass cabinets was not a difficult enterprise in my case. I already had the cabinets available from a previous focused-array-design attempt. That system consisted of eight 6″ cone drivers per side, arranged into an arc, with a 6″ axisymmetric horn driver at the center of each array to provide for frequencies above 2kHz. The system sounded quite good, but not as good as some commercial systems of less oppressive bulk.

Even with the increased directivity of the focused-array arrangement, there was still a noticeable difference in sound character between the horn and the cone speakers. The bass quality was very impressive, however. There was a very effortless quality to the bass and no boxy sound at all.

There are four separable bass cabinets for each channel. Each cabinet consists of two isolated $0.7ft^3$ chambers, housing one 6″ paper cone driver (MCM part number 55-1470) per chamber. I tuned each chamber, using a 4.5″ port length of 1.5″ diameter PVC pipe, to a box resonance of 32Hz. The alignment is sixth order, that is, a ported cabinet with a second-order electronic high-pass (peak at 28Hz, with a Q of 2). This alignment is excellent at utilizing small drivers in limited cabinet size down to low frequencies, while still playing cleanly when fed program material with infrasonic out-of-band signals.

The measured near-field –3dB point of the woofers is at about 32Hz (*Fig. 4*). The simple first-order electronic crossover diagram (f_C = 340Hz) is shown in *Fig. 5*. The eight total woofers per channel are arranged in a line to minimize vertical radiation and floor/ceiling reflections. In my setup, they were wired in series-parallel for an overall (nominal) impedance of 4Ω.

Swans M2 kit

The Swans M2 is a floorstanding model that features several technological achievements and sound quality distinctions.

The speaker system is a two-way bass-reflex design with MTM driver configuration. The front baffle is very narrow with rounded edges to reduce cabinet diffraction for better clarity and imaging. The internal panels and corner reinforcement bars substantially suppress unwanted cabinet vibrations. The bottom part of the cabinet is sealed and can be filled with sand or lead shot for better stability and further performance improvement. A port is mounted on the rear panel.

The drivers used in the Swans M2 represent a new high performance design from Hi-Vi Research. The 5-inch paper/Kevlar cone bass-midrange has a rubber surround, cast aluminum frame and a magnetically shielded motor system. This driver utilizes a central phase plug to avoid air compression, improving frequency response and dispersion The extremely rigid cone is hand coated with a special dampening compound to further maximize its performance. The cone is coupled to a selected grade rubber surround, this provides break-up free operation and very low distortion even at high power levels. These key features greatly contribute to the Swans M2's clear transparent sound and effortless dynamic performance. Swans M2 delivers amazing bass without runing in "doubling" or Doppler distortion problems.

The tweeter is a high-tech planar isodynamic design that employs Neodymium magnets and extremely light Kapton® film, with flat aluminum conductors.

The vibrating element of the tweeter is almost weightless in comparison to a conventional dome driver. This unit provides an immediate and precise response to any transients in original signal, and gives the Swans M2 an exceptional ability to reveal the true dynamics of instruments with a complex high frequency spectrum.

The crossover is a second order Linkwitz-Riley type resulting in an in-phase connection of the drive units. The crossover frequency between the two drivers is 3.3 kHz and only high quality polypropylene capacitors are used. Each filter has it's own dedicated board mounted on a special rubber interface to reduce vibrations and microphonic phenomenon. The filter boards are spaced inside the loudspeaker with the inductors positioned at right angles to minimize the interaction.

Swans M2 provide very even acoustic power dispersion. The important horizontal early reflections that create spatial impression and add to the overall presentation have the same even spectral balance as the direct sound. these are crucial features of a good loudspeaker.

On the contrary, the vertical dispersion is well controlled in the midrange and high frequency domain in a 15° arc symmetrically to the reference axis. While 15° create adequate room for adjusting a listening position, the floor and ceiling reflections are well down in amplitude. This feature greatly contributes to the clarity of sound and imaging of the system.

Swans M2 kit includes:

- 4x F5 paper/Kevlar bass-midrange drivers,
- 2x RT1C isodynamic tweeters with sealing gaskets,
- 2x dedicated tweeter crossovers,
- 2x dedicated bass-midrange crossovers,
- two ports and two Swans logos,
- two pairs of heavy-duty gold plated terminals.

Cabinets are not included.

For those who are interested in a home theater set up, the instructions and parts for correspondent central channel speaker are available.

The drawings of the cabinet shown here represent general dimensions required for optimum bass performance. Rounded corners are advisable as they improve imaging and clarity. Actual finish and appearance is a matter of personal taste. The system should be installed on adjustable spikes and slightly tilted back to aim tweeter axis at listening position.

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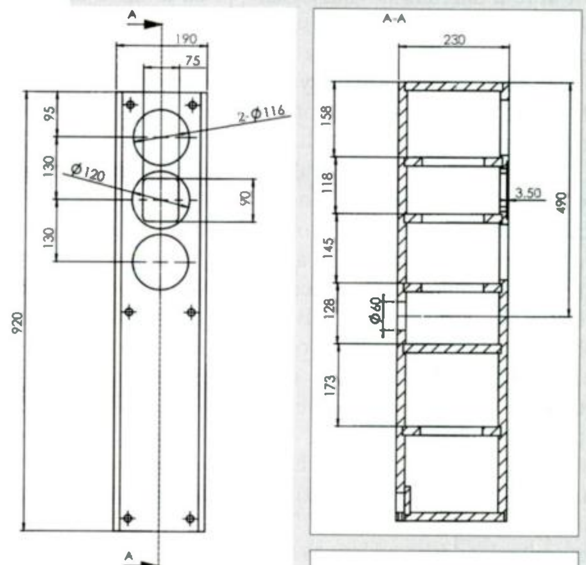

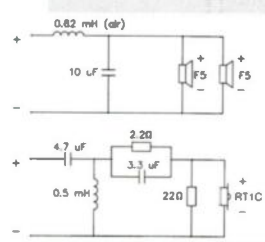

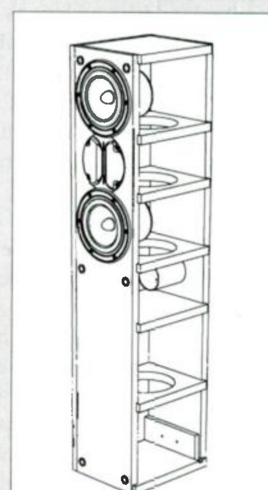

SPECIFICATIONS

Frequency response (1m,half space)	53Hz-20kHz,±2.5dB
Sensitivity,1W/1m (100Hz-8kHz averaged)	87.5 dB
Nominal impedance	4 ohms
Harmonic distortion At 90dB SPL, 100Hz-10kHz, 1m	THD less than 1%
Power handling	80W nominal, 150W music
Dimensions,HxWxD (without spikes)	920x190x230 mm 36 1/4x7 1/2x9 inches

Amplifier requirements:
30W recommended minimum.

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Reader Service #89

The big tradeoff in using these woofers was the increased bulk and size: the two sleek curved arrays now became only part of a wall of stuff which now makes up the full-range speaker system! In terms of room dominance, this setup is intimidating. It is not a good choice for those who must use a listening room for other purposes, nor for those who wish their speaker systems to visually recede into the background.

DEALING WITH THE ELECTROSTATICS

The natural slope of the electrostatics' mid- and low-frequency response requires equalization in some form if a flat response is to be obtained. One approach is to provide this equalization ahead of the amplifier that drives the midrange panels of the focused array. This is a good technique under normal circumstances, but since the tweeter in my system must have a different driving signal response, this would require use of a separate tweeter amplifier. I was expecting that some local audiophiles might want to try different single amplifiers with the ESL portion of the system, so I rejected this approach.

Instead, I chose to use a passive equalization method, similar to that used in the original Quad ESLs. Since the impedance of an electrostatic element is simply that of a capacitor, it is a simple matter to produce an electrical response that increases at 6dB per octave with falling frequency (or, in other words, which falls at 6dB per octave with rising frequency). You need only drive the capacitive panel through large resistors.

The total resistance R and the capacitance C act to form a first-order low-pass filter, with the corner frequency at $1/(2\pi RC)$. This corner frequency should be as low as you need for the equalization to hold (about 350Hz, in this case). Since the capacitance of each panel of the array of eight was 200pF, the total feed resistance I eventually used to drive the eight electrostatic midrange panels (parallel connected) was 300kΩ. This was divided into two sets of resistors (5 × 30k, in series, per set), one to drive the stators at the front and the other to drive the stators to the rear.

An electrostatic speaker requires a high-voltage push-pull audio signal to drive the stators. This is normally supplied from a stepup transformer. Most ESL designs are limited by the tradeoff between transformer turns ratio (how much ESL drive voltage you can get for a given amount of nominal 8Ω amplifier power) and the usable bandwidth of the transformer. Also lurking in the background is the potential problem of transforming the ESL's capacitive impedance into a dangerously low impedance at the transformer's primary side where the amplifier connects.

Most power amplifiers become quite unhappy when directly driving loads that look like large capacitors. My use of resistors to provide the slope equalization caused a need for higher drive-signal voltages than usual because the resistors equalize by attenuating the higher frequencies. Hence, even greater stepup ratios are needed from the transformer (more on that later). On the other hand, the equalization resistors isolate the midrange-panel capacitance from the transformer and from the power amplifier. And the focusing effect of the array provides some additional gain via higher sound pressure at the focus.

CROSSOVER TRICKS

Since the EQ resistors come after the secondary of the transformer, the signal at the transformer itself is unaffected, and could be used to drive the tweeter panel. The tweeter's response within its operating band is essentially flat, since the panel dimensions are sufficient to largely prevent front-back cancellation at high frequencies. In fact, when driven with a flat response voltage, the natural high-pass corner of the tweeter panel is essentially first order at around 3kHz. This allows for a rather cute crossover trick that I used for a while—a crossover network using only resistors.

If the midrange and tweeter panels are of identical construction, as they were initially in the FAE, they both have the same natural rolloff: the response is essentially flat above 3kHz and rolls off at 6dB/octave at low frequencies until the 90Hz panel resonance is approached. This characteristic, very similar to a first-order high-pass, can be used as is (no network required) for the tweeter. For the midrange, application of a –6dB/octave slope from the equalization resistors flattens the region between the equalization cutoff at $1/(2\pi RC)$ and 3kHz, and gives a downward tilt to the otherwise flat response region above 3kHz to provide a first-order low-pass response. All that is needed then to make the tweeter panel and midrange panels combine almost perfectly is to adjust their relative levels, which I did by driving the tweeter from lower-ratio taps on the transformer secondary and by setting the midrange panels' RC equalization corner such that the levels matched.

I eventually abandoned this scheme, however, because of drive considerations. I was connecting the transformer in nonstandard ways to achieve higher turns ratios, which allowed for good volume levels using amplifiers of 40W and under. A problem in using a very high turns ratio for driving the midrange panels is that, while the resistors isolate the panels' capacitance, the reflected stray capacitance from the transformer windings becomes a problem.

Even with the secondary windings disconnected, the primary impedance dropped to under half an ohm at the point where the reflected secondary winding capacitance became series-resonant with the transformer's series inductance. I blew out several homemade amplifiers in the course of these tests and decided that the setup was just not practical.

Adding a series resistance to the primary could prevent the horrendous impedance dip, but would also drastically roll off the frequency response of the tweeter. Eventually, I opted to drive the tweeter panel from a separate transformer and to use separate cross-

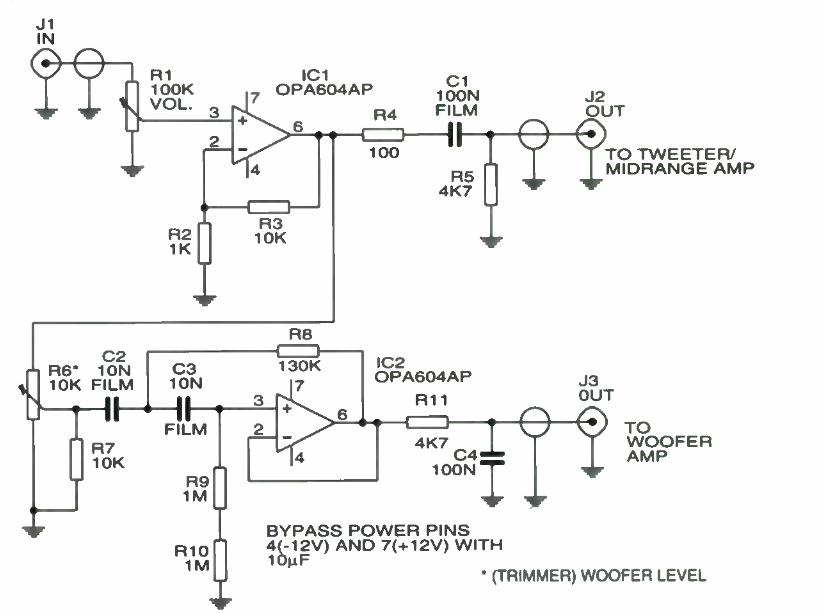

FIGURE 5: Schematic diagram of woofer electronic crossover and high-pass equalizer.

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18000 pieces **Panasonic 3.3 mfd polypropylene** cap., 10%, 200 volt, dipped with long leads, 33mm x 11mm x 20mm, only **60 cents each!**

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5000 pieces **nichicon 8.2 mfd Mylar Capacitor**, 250VDC, 10%, axial leads 35mm long, yellow jacket, 13mm x 38mm x 22.5mm, AS series. **Price each $1.10.**

36 pieces **Audax TM025A7 Micro Series 1" textile dome tweeter**, 4 ohm, Neodymium magnet, only 37mm diameter flange, no screw holes, Fs 1100 Hz, 87dB, 70 W, good choice for automotive applications of limited space, low Fs allows for crossover point below 3kHz, textile dome offers smoother sound, **Price each $19.00**

850 pieces **Audax TW51A 10mm poly dome tweeter**, 8 ohm, Fs 2900Hz, 91.5dB, 40 watts at 5kHz, 2" square flange ½" depth. Good choice as a supertweeter add on to an existing system or for use in small 2-way systems. **$3.50 each**

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100 pieces **Peerless 821385 4.5" paper cone midrange**, die-cast housing with non-resonant rear loading chamber and basket in one piece, foam surround, 8 ohm, Fs 224Hz, Qms 3.35, Qes 1.88, Qts 1.20, 90dB, 150W, very smooth response, natural sound, 122.5mm square basket, 105mm depth, **$29.00 each** (regular price $44.00)

400 pieces **Famous Maker 10" woofers**, polypropylene cone, foam surround, inverted dust cap, 2" Ø VC, Kapton former, 8 ohm, Fs 22.7Hz, Qms 8.56, Qes .36, Qts .34, Vas 109 ltrs, x-max 8mm peak, 88dB, Re 6.4 ohms, Mms 69.3g, 150W, F3 of 45Hz in 1.2cf sealed or F3 of 33Hz in 1.8cf vented, 2.5" Ø x 8" long. **Price each $32.00**

13 pieces **Audax HM210Z2 Aerogel 8" Dual Voice Coil woofers**, 8 ohm/8 ohm, Fs 28 Hz, Qms 9.08, Qes .32, Qts .31, Vas 63.1 ltrs, 70W, 89dB, 40mm Ø VC, 4.25mm x-max, smooth response to 2000Hz, F3 of 68Hz in 20 ltrs sealed, F3 of 45Hz in 30 ltrs vented, 2" Ø port 4" long, use each coil independently for 2x 8 ohm, series for 16 ohm or parallel for 4 ohm. **$75 each**

80 pieces **Vifa D26SG05 Shielded Magnet 1" Textile dome tweeter**, Fs 1450 Hz, 92dB, 6 ohm, 4" flange with 3 1/8" cutout, 80W, usable from 3kHz to 30kHz, ferrofluid cooled VC **Price $17.50 each.**

This offer expires Sept. 30th 1998

54 pieces **Peerless 850112 CSC 7" Sandwich cone woofer**, rubber surround, inverted dust cap, 8 ohm, good response out to 3kHz, Fs 37.4 Hz, Qms 2.43, Qes .52, Qts .43, Vas 33 ltrs, 88dB, 110W, 14 oz magnet, Fs of 60Hz in .7 cf sealed or F3 of 40Hz in 1 cf vented 2" Ø vent x 4.75" long, good choice for 2-way system with decent bass response. **$25.00 each**

42 pieces **Peerless 850143 CSC-X 10" Sandwich cone woofer**, rubber surround, short circuiting ring in magnet system to reduce distortion, 4 ohm, Fs 26 Hz, Qms 4.17, Qes .61, Qts .53, Vas 118.8, 90.3dB, 150W, 5.5mm x-max, F3 of 35Hz in 3.5 cubic foot sealed box, with Qtc of .79, 2cf box has F3 of 37Hz and Qtc of .94 for autosound use, **Price each $45.00**

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65 pair **Europa 23 Wedge Mount Tweeters**, 4 ohm, 14mm poly dome with 6dB Mylar crossover filter for 4500Hz crossover, square shaped wedge mount with sloped front, 66mm tall, 50mm x 65mm foot print, very good sound quality, don't let the low price fool you, **Price per pair $10.00**

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1200 pieces **0.47mfd Mylar capacitor**, 400V, PC mount, 26mm x 17mm x 8mm, white jacket, **10 pieces for $2.00**

30 pieces of **Sammi Sound ME300B100** 12" professional woofer, paper cone, accordion cloth surround, good for guitar speaker, 8 ohm, 100 watt nominal, 98dB, frequency range 48-6KHz, 2" voice coil, 50 oz magnet, Re 6.6 ohms, Vas 113.6 ltrs, F3 47, Qms 6.57, Qes .276, Qts .27, all for the incredibly low price of **$25.00 each**.

48 pieces **Peerless 812774 1" Textile dome tweeter**, same as 812687 but with ferrofluid cooling of the voice coil, 8 ohm, Fs 1460 Hz, 90dB, 140W, max linear SPL 106dB at 60W, recommended crossover frequency of 3kHz, 105mm diameter aluminum flange, 79mm cutout hole, 23mm deep, replaceable voice coils available. **Price each $18.00**

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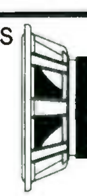

overs ahead of the transformers to provide an amplifier-friendly impedance (*Fig. 6*).

TRANSFORMER TRICKS

I mentioned that I used nonstandard connections to the transformers to achieve higher turns ratios. Before explaining this further, let me caution you to try this only if you are able to measure the resulting primary impedances at both low and high frequencies—these connections are going beyond what the transformer designers intended. Also, beware of using these connections with high drive levels (amplifiers much above the 50W level), because the secondary voltages could otherwise go beyond what the winding insulation can handle, and the transformer core can saturate more easily. **Use any of this information at your own risk.** And, as always, be very careful around the panel or transformer secondary circuit when the speaker is activated or when the stepup transformers are being driven—the voltages and currents could be lethal.

I have used both the transformers from David Lucas and some similar transformers from Roger Sanders (model TS-142A) in this design. In my application and in bench measurements, they proved to be nearly equivalent in performance (as well as in appearance) but the units from Roger Sanders cost only a little over half the price of the Lucas transformers.

A difference you may find relevant is in the winding configurations. The Lucas unit has a single center-tapped secondary (8kΩ CT), while the Sanders unit has an additional set of secondary taps (8kΩ CT and 2kΩ CT). That could make the Sanders unit more useful if you are going to try the "resistor-only" crossover trick. On the other hand, the Lucas transformer has an additional "2Ω" primary tap, which can provide for more choices and higher values of turns ratios.

Audio transformer windings are usually marked in terms of impedance values, which are related to the turns ratios between the windings. These impedances are meant to indicate in some manner the range over which the voltages and currents on either side approach ideal behavior over the operating band. For example, an ideal transformer with a fixed wideband drive level should draw no primary signal current if the secondary is left disconnected.

A real transformer will draw some current in this configuration due to stray shunt capacitance and to limited effective shunt inductance from the windings. This current should ideally be small compared to the currents being utilized by the circuit, and the circuit impedance levels dictate these. If you have a transformer with multiple primary taps and can further limit the high- and/or low-frequency range over which you expect the transformer to behave (as in the case of the FAE midrange panels), or if you can compromise on what you consider to be ideal transformation, you can sometimes push the transformer for higher turns ratios.

Transformer impedance ratios scale with the square of the turns ratios. For instance, the nominal 8Ω primary to 8kΩ secondary of the Lucas and Sanders transformers equates to a $(8000/8)^{0.5} = 31.623{:}1$ secondary to primary turns ratio. The turns ratio is also the voltage stepup ratio: if you apply 1V AC to the 8Ω winding of the primary (between the "0" tap and the 8Ω tap), you should get 31.6V AC between the 8kΩ wires of the secondary.

Since we are talking about ratios, for the sake of analysis, let's consider the winding between the 0Ω and the 4Ω tap to consist of one single turn and then find the number of turns in the other windings relative (or "normalized") to this. Between the 8kΩ winding and the 4Ω winding, there is a turns ratio of $(8000/4)^{0.5} = 44.721$. So, we'll consider the secondary as consisting of 44.721 turns.

We already know that the secondary has 31.623 times as many turns as exist between taps 0 and 8Ω. So there must be (44.721 turns)/31.623 = 1.414 turns between the 8Ω and 0 primary taps. If so, and if there's one turn between the 4Ω tap and the 0Ω tap, and if we know that common windings are shared, then there must be 0.414 turns between the 8Ω tap and the 4Ω tap.

Similarly, we can determine that there are 0.586 turns between the 8Ω and the 16Ω taps, and again a single turn between the 4Ω tap and the 16Ω tap. Driving from the 8Ω tap and the 4Ω tap, we then can get an overall major secondary-to-primary turns ratio and voltage stepup of (44.721/0.414) = 108.0. Driving between the 16Ω and the 8Ω, we can get a ratio of 76.34. With the Lucas transformer, driving between the 4Ω tap and the 2Ω tap could provide a ratio of 152.6.

For the midranges of the FAEs, I used the 8Ω taps and the 4Ω taps (ratio = 108) as the primary.

DRIVING THE TWEETER PANELS

Since I also wished to drive the tweeter panels with a very high turns ratio, lest they limit overall system sensitivity, I could not use the Lucas or Sanders transformers and still maintain a friendly impedance to the drive amplifier. I needed a high-turns-ratio transformer with lower shunt capacitance (fewer turns or greater spacing between them), but this transformer did not need to cover low frequencies and hence did not need a large number of total turns or a large iron core. Also, I wished to avoid having to use an expensive transformer for driving the tweeter alone.

The solution was, again, unconventional.

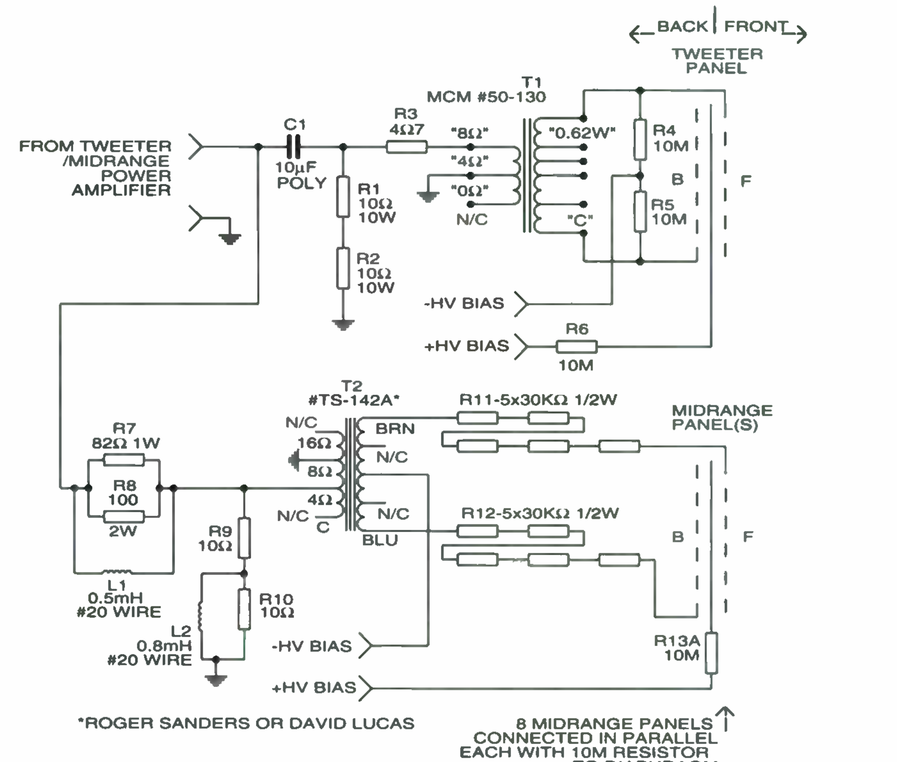

FIGURE 6: Midrange and tweeter stepup and crossover diagram.

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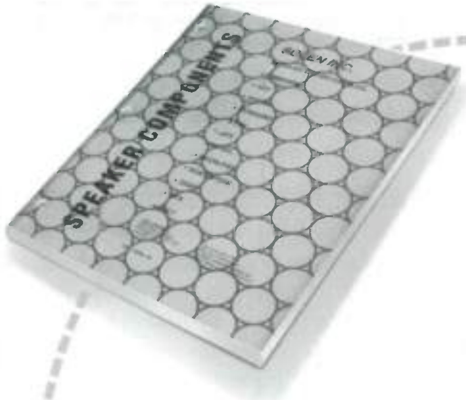
The Process of Design.



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We have all seen the cheap 70V public-address transformers used to pipe intercom or background music around businesses and institutions and would normally not consider these inexpensive units for high-fidelity use. These transformers are meant to be operated by connecting the low-impedance secondary to an 8Ω speaker and driving the primary from a drive signal connected to the taps, which give a desired output level.

The higher the turns ratio, the lower the output signal that drives the speaker's terminals, for the transformer operates in a "step down" mode. This allows for attenuation of the output signal without wasting power, for the amplifier then merely "sees" a higher-impedance speaker and provides it less current. These transformers cost only a few dollars, and can be configured to give the needed turns ratio for driving an electrostatic tweeter.

As before, let me warn that these transformers were not designed to be used the way I used them, and in particular, the insulation is not designed to handle the kinds of signal or bias voltages used by ESLs! Use at your own risk. But on the other hand, over several months of use, I have not had a single problem from using the PA transformer this way, and the sound quality when driving only the tweeter is very good—it may even be usable down into the midranges, but I have not tried this.

There is a measurable but not severe high-frequency rolloff from the transformer near 19kHz, but the input impedance at high frequencies does not dip as severely as it does with the larger transformers. The turns ratios obtained when you use a 70.7Ω transformer with various connections are given in *Table 1*. I used the highest available ratio, 106.9, to drive the tweeters in the FAE. For these higher turns ratios, you will need a transformer with both an 8Ω tap and a 4Ω tap.

TABLE 1

STEPUP TURNS RATIOS USING 70.7V PA DISTRIBUTION TRANSFORMER

SECONDARY CONNECTIONS (TO ESL), FROM COMMON TO	FROM 8–0Ω TAP	FROM 4–0Ω TAP	DRIVEN FROM 8–4Ω TAP
10W	7.83	11.07	26.73
5W	11.07	15.65	37.78
2.5W	15.65	22.14	53.45
1.25W	22.14	31.31	75.59
0.62W	31.31	44.27	106.9

Normally, when driving an ESL, you use a center tap on the secondary for connection of the bias supply for the diaphragm. However, since this connection does not supply current (other than leakage), you can instead connect a 10M resistor from each stator connection to a bias-feed point as shown in the schematic of *Fig. 6*.

HIGH-VOLTAGE BIAS SUPPLY

I originally purchased and built the "Battery Bias" supply kit sold by David Lucas, but did not have much success with it. I had difficulty obtaining higher voltages from it without playing with added capacitors on the transformer that was supplied, and then had problems with transistors failing in the circuit with the added capacitance. Also, the adjustment range afforded by the potentiometer in the circuit was small and irregular. More coarse adjustment could be obtained by changing connections on the diode multiplier ladder, but since I intended to do a fair amount of experimentation, I needed more continuous adjustment.

I designed a simple circuit (*Fig. 7*) using a backward-connected filament transformer driven by a FET and an IC intended for switching power supplies. The output voltage adjusts by varying the duty cycle of the current pulses drawn through the transformer's primary, and can give continuous adjustment from about 100V to 3.5kV at the output. You can change the number of diodes and capacitors in the voltage-multiplier ladder as required for other voltages. The circuit operates at 20kHz to avoid the possibility of any annoying "whining" sounds from finding their way to the panels or from the transformer body.

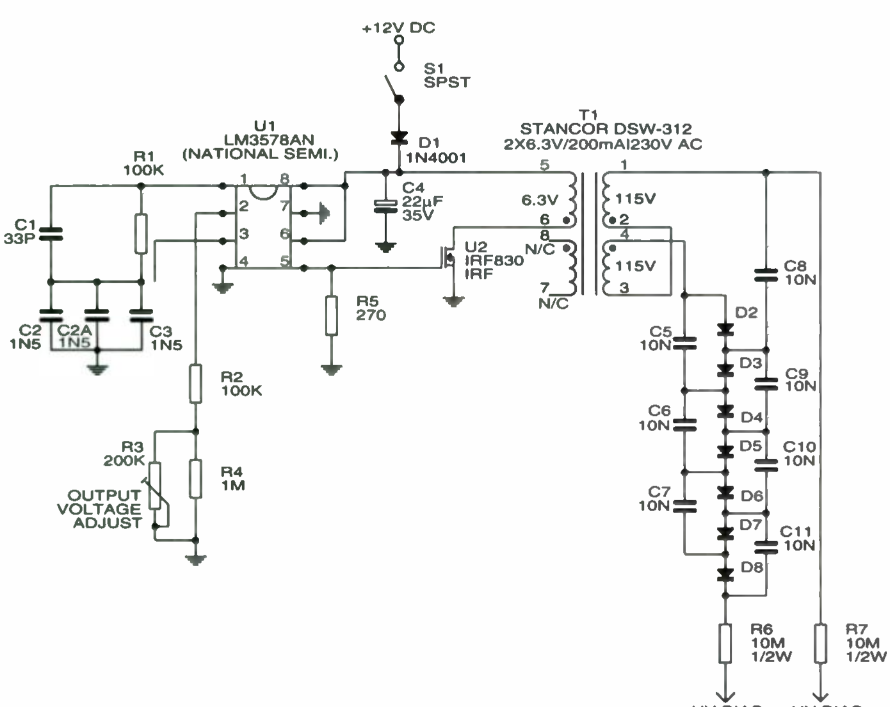

FIGURE 7: High-voltage bias supply.

The transformer is a Stancor number DSW-312 (dual 6.3V to dual 115V windings), with one of the 6.3V windings driven as the primary and the two 115V windings series-connected for the secondary. Other small low-voltage transformers may also be usable, but I suggest staying with the "split bobbin" types to help avoid voltage breakdown problems. Once again, I must remind you that this transformer was not designed to be used the way I am using it—"use at your own risk."

I suspect you can substitute the FET with a much lower voltage type, such as IRF530, 630, or 730. I used the one specified only because of easy availability. I obtained the DC input for the bias supply from a 12V DC, 100mA "wall wart" type power supply.

Electrostatic Loudspeaker Transformers, Mylar®, and information is available from Barry Waldron (1847 Country Club Dr., Placerville, CA 95667, 530-622-1539, E-mail esl@information4u.com), who is now marketing these items on Roger Sanders' behalf.

In Part 2, I'll discuss construction details.

Reader Service #45

THE SEISMIC STACK SYSTEM

By Philip Abbate

Back in my rock 'n' roll days I was sold on the benefits of active crossovers. I don't believe there is any better way to get realistic, dynamic, high-decibel, low-distortion sound without them. This technique is not unique to live sound reinforcement as it once was; many of the top-rated high-end systems use active crossovers and bass equalization.

One of the key reasons to sell partially powered speakers such as the Genesis and Infinity is to allow biamping. They come with the bass amps so they can slip in the equalization. The most prominent advantage is the synergy of the amplifiers as they work together to generate the equivalent of more power than their mere sum can provide.

CONVENTIONAL CROSSOVER COMPARISON

Suppose you wish to reproduce a complex signal consisting of three 28.28V sine waves. (28.28V is required to develop 100W into an 8Ω resistive load according to the formula power = volts^2 divided by load resistance in ohms.) Assume each wave is precisely at the center of the passband of one of the three drivers in a three-way speaker system. If you connect the low-level signal to a 300W amplifier, you would expect that amplifier to deliver 100W (28.28V) to each driver in the speaker system through its passive crossover. Right?

Fat chance. To produce that complex signal without distortion, that amplifier would need to generate a voltage equivalent of one (in-phase) sine wave on top of another, on top of another, or 28.28V × 3 = 84.84V. Into an 8Ω load, that signal would require an 899W (84.84V^2/8Ω) amplifier. On the other hand, triamping with an active crossover simply uses three 100W amplifiers to provide power to the various drivers.

Other benefits of an active crossover are: reduced intermodulation distortion as amplifiers operate over a narrow bandwidth; elimination of the transfer of harmonics to the tweeter as the bass notes clip the amplifiers; easy incorporation of multiple active equalizers before the amplifier; compensation for driver-efficiency differences; and a higher amplifier damping factor presented to the speaker modules, since there are no passive inductors raising the impedance of the signal path.

CROSSOVER TOPOLOGY

I started my Seismic Stack System project (*Photo 1*) with a scratch-built active-crossover topology (*Fig. 1*) based on the Baekgaard constant-voltage crossover technique, which ensures that the high-pass (HP) and low-pass (LP) outputs of a second-order Butterworth filter sum flat. I chose the second-order Butterworth because it is free from polar tilt, an important factor when the drivers are physically offset and pointed straight ahead.

The theory of the active implementation of this crossover is described by G.R. Koonce in "The Baekgaard Crossover Technique" (*SB* 2/95, p. 20). No schematic is provided because I regularly change the crossover.

If building an electronic crossover from scratch does not appeal to you, several kits are available from Marchand Electronics (www.marchandelec.com). You could also use commercial high-end units or pro sound models from the likes of DOD. The least expensive crossovers are those made for autosound, which should work pretty well if you have a clean 12V power supply. As your system improves, however, the less expensive crossovers may become the system's weak link.

The CD/home theater switch routes the L&R output of the Pro Logic preamp to the L&R line-level input of the crossover. The CD position puts the world's best preamp (a straight wire) between the variable output of my Marantz CD-67se CD player and the crossover when I am listening to music.

The active crossover's HP output feeds the mid-high modules. This output is the constant-amplitude summation of an adjustable-frequency-state variable filter's high-pass and bandpass outputs. Nominal f_3 sounds best at around 100Hz. The level control for the mid-high output is a stereo-ganged audio taper pot.

The mid-high module's passive crossover is a filter that results from the combination of the electrical response of a network of capacitors and inductors and the natural driver rolloff. It splits the active crossover's HP output and feeds the appropriate portion of the signal to the 7″ midrange and 1″ tweeter. Keep reading for details on how you can download a program that lets you input all of the following parameters and graphically view the predicted results.

PHOTO 1: The seismic stack.

METHODOLOGY

The methodology I use considers the natural or mechanical response of the drivers reaction as well as the electrical crossover. The overall response model therefore uses more of the real-world parameters that affect the sound, and models the higher-order acoustical filters that result. The combined acoustical filter is of a higher order than that supported by the electrical filter alone.

For instance, the tweeter and midrange in this design (*Photo 2*) have differently shaped baffles, and therefore begin to reinforce the sound of different frequencies at different rates. The baffles create diffraction loss for waves too small to reflect from them. This refraction

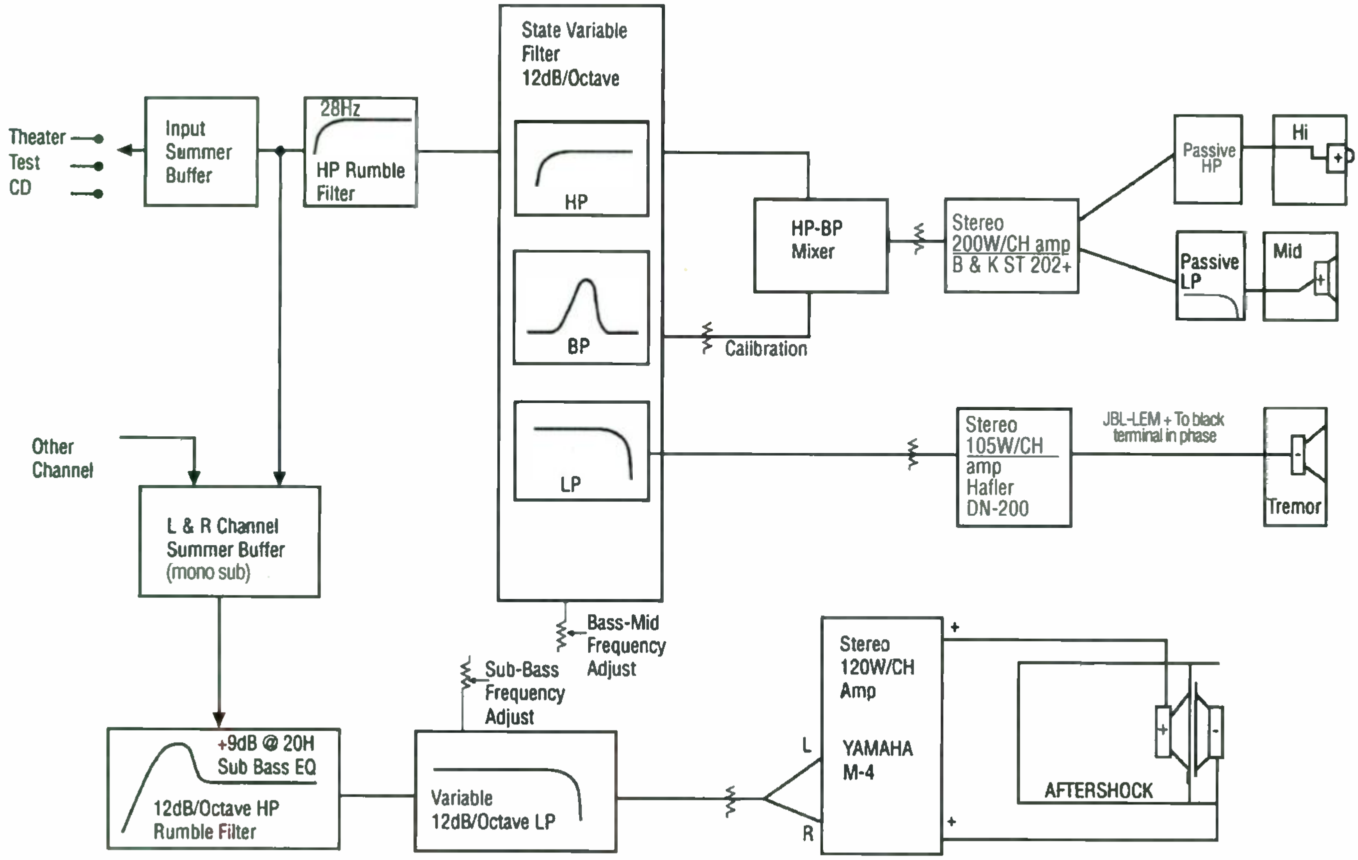

FIGURE 1: Triamp block diagram; one of the two channels shown with mono sub bass.

phenomenon affects the balance of the system, and should always be considered in crossover design. (See Joe D'Appolito's "The Swan IV Speaker System" [*SB* 4/88] for a detailed discussion.)

One of the undeniably audible differences you can hear and easily measure is the result of driver offset. Varying the spatial relationship of the tweeter and midrange affects the time the sound waves coming from each driver arrive at your ears. You need to consider this parameter in crossover design, especially if it has a fixed, rather than adjustable, offset.

I usually begin with some second- or third-order filter having textbook values, enter them into the program, and then start to play around until I find something that looks flat and phase-coherent that I can construct from the parts in my junk box. From that point, the tweaking is more ritualistic than analytical.

ABOUT THE AUTHOR

Phil Abbate resumed building speakers for friends and family four years ago. Tired of not being able to tell the difference between the infrasonic output of his stereo and the ground's shaking on its own accord, he recently moved from Los Angeles to Atlanta. Some of the friends Phil has met through the Atlanta Audio Society (www.mindspring.com/~chucksaudio) have diagnosed Phil as a terminal "tweak-o-phile" case.

MODELING THE DESIGN

Ralph Gonzales' "Real-World Two-Way Crossovers: A Design Method" article (*SB* 2/92, p. 18) explains how to use his Loudspeaker Modeling Program (SD-LMP) to make easy work of modeling the response of a tweeter's LP rolloff and a midrange's HP functions in conjunction with the electrical components to design a crossover.

(I downloaded a two-way-only demo version of Ralph's LMP from Madisound's BBS [www. itis.com/madisound]. The multiple-driver enhanced version is available from Old Colony [www.audioxpress.com].)

First I entered the tweeter's and midrange's natural high and low corner frequencies by looking at the manufacturers' spec sheets and guesstimating the locations of the drivers' corner frequencies, Qs, and orders of natural rolloff. Just put your guesses into LMP and look at the response graphs. Make a big change to get a sense for what happens if you lower the Q or raise it.

Once you achieve this, you have done a lot more than merely learn to iteratively enter parameters until LMP's plot for the crossoverless drivers approximates the specsheet response. You have developed a notion for what small or large differences in some parameters mean to driver response.

The efficiency of the tweeter (92dB) and the midrange (93.3dB) are closely matched because they share the same amplifier. Depending on the room I am in, the tweeter still needs about 1Ω to 1.5Ω of attenuation. The final results of this crossover will be printed in my obituary. You see, I never said I would stop being a compulsive tweaker.

BANDPASS-OUTPUT FUNCTION

The active crossover's bandpass (BP) output that drives the 14″ woofer consists of the LP section of the state variable and a 35Hz second-order HP filter that facilitates the sixth-order alignment

PHOTO 2: Tweeter and midrange.

of the bass modules. I used Boxmodel to determine the parameters for the HP.

The bass modules' low-end rolloff with and without the filter is essentially the same. I can tell the difference in the sound with and without the filter. I like it better without the filter, but when the system is cranking, the filter cuts down on the 14″ woofer's inaudible infrasonic excursion.

Boxmodel's driver-excursion graph shows how its cone will move when low-frequency drive voltage is applied, even if the signal is in the acoustic cutoff region of the woofer cabinet system. Inaudible woofer movement robs systems of power and produces distortion by modulating higher frequencies the woofer simultaneously produces. The level control for the bass output is a stereo-ganged audio taper pot.

The active crossover's sub-bass output is summed left and right mono. The extreme low end is a fixed, second-order, 20Hz subsonic filter with 9dB boost that affects the sixth-order tuning of the Aftershock subwoofer (*SB* 6/96, p. 34). The low-pass section is adjustable independently of the bass modules' low-end crossover output. This adjustment allows me to tweak the topside of the Aftershock so it seamlessly blends into the Tremor's low side. The sub bass has its own mono audio taper volume control, which is parallel to both of Aftershock's amplifiers. The subwoofer is a 111-ltr isobaric design implemented with two 15″ Altec Lansing 421LF bass speakers.

STEREO TRANSISTOR POWER AMPS

The system's amps have in common large, heavy transformers, huge power-supply capacitors, and substantial heatsinks. I used POOGE'd B&K ST202+ for the midrange and tweeter modules, POOGE'd Hafler DH-200 for the bass modules, and 120 WPC Yamaha M4 for the subwoofer.

All seven of the speaker modules in *Photo 3* are finished in matching red oak veneer and solid oak trim. The cabinets are stained with minwax #245 Golden Pecan and sealed with Minwax clear satin polyurethane. Connections are made with gold five-way binding posts. The basic methodology used to construct the cabinets is the same as I used on Aftershock.

TWEETER MODULES

The two tweeter modules house 1″ Focal TC 90 Tdx inverted titanium-oxide dome tweeters and their portions of the passive HP crossover.

These pentagon-shaped enclosures (see *Photos 2* and *3* for the finished module) are easier to make than they look if you have a radial-arm or table saw or are an expert circular or jigsaw operator. Each frame is made of five pieces of plywood or hardwood, measuring 3¾″ on each of the face sides. Set your saw to cut a 54° angle, and slice the ten equal pieces, running your stock through twice and keeping both of the longer edges of each piece on the same side of the board. The cross section of each piece has the shape of an isosceles trapezoid, with the longer base 3¾″, and the base angles 54°.

PHOTO 3: Home-theater speakers.

Match up the grain by using two strips of "blue" low-tack masking tape to hold the five wood pieces together. Put two strips of masking tape on the workbench and place five of the pieces with their ends adjoining so that they form the pentagon. When you have a combination of grain that works for you, put glue on the joints and then hold them together with a band clamp (or an old belt, or more tape).

The pentagon's ends are capped with solid oak front and rear. I used the glued-up pentagon enclosure to trace the outline for four caps onto an oak board. Before cutting the cap shapes from the board, I traced the cutout and recess for the Focal tweeter and binding posts in the center of each. I then used my tweeter template and router to cut the tweeter recesses. I cut the holes for the tweeter's magnet and binding posts with a jig saw.

Once I'd finished the sawing that required clamping down the full-size board, I cut out the pentagon ends freehand on the radial-arm saw. The jigsaw would work just as well, and you could forgo the recesses if you don't have a router. I tapered the front baffle with my belt sander using progressively finer grit. Once I'd matched all to size, I glued the ends and clamped them in place.

The unequal path lengths between the center of the dome and the edges of the pentagon, and the tapered baffle all combine to minimize the effects of baffle diffraction and to enhance dispersion, both of which affect imaging and frequency response. The tweeter modules' HP filter uses high-quality SCR poly-

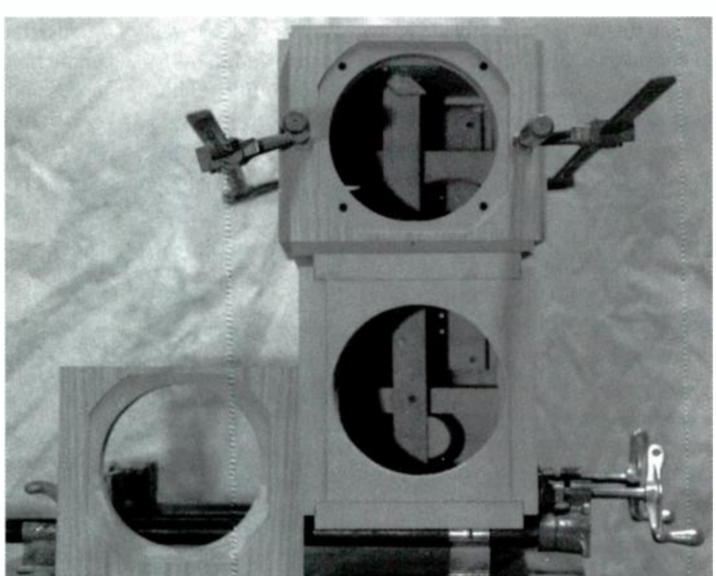

PHOTO 4: The mids ¾″ recessed veneered baffle is laminated onto the 1″ MDF enclosure.

propylene capacitors and 16 AWG CFAC foil wire inductors by Solo.

THE MIDRANGE MODULES

The midrange modules are Focal 7V-513 7″ polyglass bass drivers in a ported enclosure. This driver was custom made by Focal for an OEM who changed the design before they were delivered. Zalytron occasionally has similar gems on sale for a fraction of the cost of the Focal catalog drivers.

The 7V-513 has a substantial magnet and a flatwound copper voice coil that makes it very efficient. Its response is fairly flat to around 3kHz, where it begins to rise slightly. Once into the treble range, it begins to roll off naturally at about 24dB per octave starting at 4.5kHz. The natural rolloff of the driver, coupled with that of an LP electrical crossover, makes for a smooth, acoustic transition into the treble regions.

The box is constructed of 1″ MDF with a ¾″ MDF mounting baffle laminated to the front. Like all the other cabinets, the midrange box is finished with an oak door skin and oak trim (*Photo 4*).

BASS MODULES

The bass modules utilize a pair of circa 1969 resurrounded 14″ JBL LE-14a woofers. The bass module serves double duty as a speaker stand for the midrange and tweeter modules. These speakers fill the gap between the lean bass of the midranges and the tactile bass of Aftershock with that solid, JBL bass punch I fell in love with, back in my pro sound days. In fact the LE-14 looks like a member of the old pro sound D, or the current E, series family of JBL drivers.

The T/S parameters of the resurrounded speakers measured very close to the factory parameters. The factory Lancer 99 box was a slightly undersized ported design with a bump of a couple of dB at 50Hz. I guess the JBL folks knew how to put the pop into pop

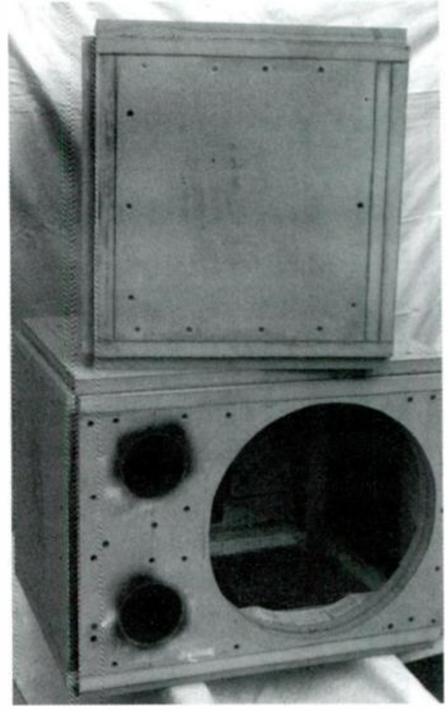

PHOTO 5: The double-wall bass-box enclosure.

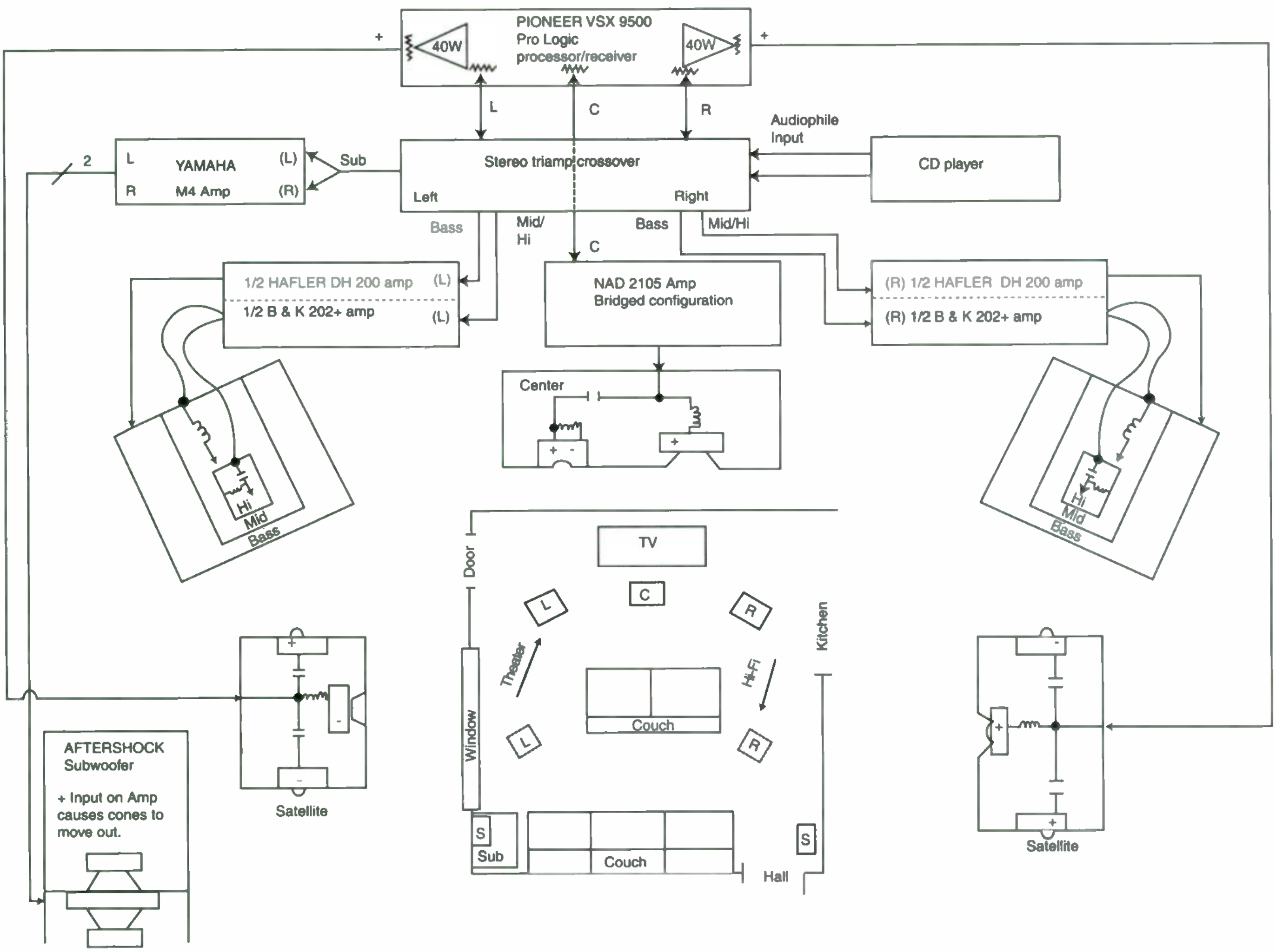

FIGURE 2: System diagram and room layout.

Reader Service #93 →

music 25 years ago. One of the speakers has the original flatwound copper voice coil and the other has a Waldom replacement. They both seem to play at the same loudness, sound the same, and measure the same. The electronic crossover works well to balance the output of the woofers with that of mid-high modules and the subwoofer.

I had to route the recess for the LE-14 after I veneered the oak sheet to the enclosure (*Photo 5*). The challenge was to attach the template to the box without damaging the box. To do this, I clamped 2 × 4s to the sides of the finished box and then screwed the template into the 2 × 4s (*Photo 6*).

The polarity of the LE-14 woofers is such that when a positive voltage is presented to the red terminal, the cone moves inward. To compensate for this, I left the woofer output on the crossover inverted. If your speakers are not inverted like the LE-14, you can build an inverter into the crossover or just wire the speakers backwards to reduce component count.

To make sure I could move the cabinet around, I designed grab points around the cabinet, consisting of the recess for the binding-post connections in the rear and the ports in the front. I also designed in the flexibility to operate the system with the ports up or down, or the cabinet on its side with midrange and tweeter modules on top. If you can't find an LE-14, the "cinder block" cabinet could be useful with another woofer, such as the Focal 10K617.

STACKING THE MODULES

The seismic stack is designed for time adjustment of the three drivers. My appreciation for time-adjusted speaker systems was spawned decades ago during my quest to locate the optimal position for HF horns stacked on top of the rock 'n' roll midrange and bass bins.

Trying this dramatic demonstration of the effects of positioning between the midrange and HF drivers will make you a believer in time adjustment too. Play wideband pink noise through a system while a friend varies the front-to-back position of your tweeter with respect to your midrange. You easily will hear the character of the noise change with an offset alteration as small as ¼″.

I believe there are many current alignments that are dependent on the room and the geometry of the listener versus speaker placement and crossover. (See "The Listening Arc Alignment," by Douglas Rauer, *SB* 2/89.) If you frequently rearrange your listening environment, a stack design will give you more flexibility in tweaking the time-adjustment of each new listening position. This one feature should satisfy even the most neurotic tweaker's basic compulsions.

I selected the offset method over the popular tilt-back method to time-adjust the seismic stack because I wished to point the midranges and tweeters in the same plane as my ears. The tilt-back method shoots the direct wave toward the ceiling, consequently placing the listener off the driver's axis.

The offset method allows the drivers to point directly on a level with the listening position, enhancing imaging and reducing image depth clutter caused by reflections of the direct wave shot into the juncture of the rear wall and ceiling. Keeping the tweeters level also helps imaging and timbre outside of the sweet spot, which is important to me since I usually have company.

One of the more popular ways to correct the effects of offset misalignment without physical or electrical delay is to overlap or spread the crossover points. Bruno Carlsson does an excellent job covering the subject in his *SB* 1/95 and 2/95 articles.

PHOTO 6: Locating the recess template on the veneered bass box.

DETERMINING ACOUSTICAL CENTERS

Since it is rare for manufacturers to publish the acoustical center of the drivers, I determined the acoustical centers of the drivers empirically. My starting point was physical alignment of the voice coil/cone/dome union of each driver as recommended by Vance Dickason (*Loudspeaker Design Cookbook*, p. 92).

I wished to determine the offsets to within half an inch prior to final cabinet design and fabrication so I could size the cabinets for minimum baffle dimensions that would maintain the optimal height of the stack. The original center of the tweeter I used was 32.5″ off the floor, which was at ear level while I slouched in movie mode on my couch. Now the system is up on an 8″ stand because that sounded right.

To minimize construction during this preliminary testing, I used the original JBL Lancer 99 cabinets in lieu of Tremor to house the reconed LE-14s, and I built test boxes for the Focal midrange.

I have never believed in quickly jumping to conclusions about the sound of a system modification, particularly when the modification would be subtle. Typically, I get used to the configuration for a while by listening to a variety of music and movies. Once your audio experience gets to a certain point with regard to both listening and equipment, the time it takes to decide what's "better or worse" will shrink.

If you are lucky enough to have a wife or daughter who is an audiophile, you could ask her opinion. Would luck be the word? I heard of a couple who nearly got divorced over a digital-interconnect choice.

I stopped soliciting the opinions of my family when they began to threaten to have me committed.

I usually focus on improving one area of the system at a time. I found that undisciplined tweaking of a variety of areas does not lead to a true understanding of what each tweak achieved, but I do it all the time because it sure is fun. By listening, measuring, and then confirming the selected offsets with the Loudspeaker Modeling Program (LMP), I believe I may never run out of options to try.

HIGH-QUALITY SPEAKER CABLES

I have tried everything short of buying expensive high-end speaker wire. Now I have the tri-amp system connected with nine pairs of shielded 22 AWG computer-control cable. The separate pairs of wire reveal inner detail that the 12 AWG "monster style" speaker wires I replaced never did. None of my audiophile buddies point to my speaker cables as the weak point in the system.

I tie all of the shields together at the amp end and cut them off at the speaker end. I tie one side of each pair to the amp positive and the other to the negative. I use three pairs for the tweeter and six for the midrange. The LE-14 has its own cable, using all nine pairs, and Aftershock has two cables, one per speaker.

SETUP AND TUBING THE ROOM

I frequently experiment with different speaker-placement and listening positions and have found a couple that are satisfying (*Fig. 2*).

For music listening, I like to keep my back to the wall and move the speakers out more into the room. This configuration has a deeper sound stage and a more laid-back high end. I believe this can be attributed to the reduction of time between the arrival of the sound wave emanating directly from the speaker and the arrival of the attenuated sound wave reflected off the back wall. Since the system is not lacking for bass, situating the seismic stacks in the center of the room does not degenerate the sound.

For movies, I sit in the middle of the room, and get up close and personal with the TV. Placing the speakers and your furniture properly will have a phenomenal effect on your room. I suggest you buy the Delos Surround Spectacular The Music/The Tests CD (Catalog DE 3179, available from Tower Records 800-648-4844) and tune away.

SOURCES

Focal-Orca
818-707-1629, FAX 818-991-3072

Marchand Electronics, Inc.
(716) 872-0980, FAX (716) 872-1960

Solo Electronics
(510) 887-8016, FAX (510) 887-1657

Speaker City USA
818-846-9921, FAX 818-846-1009

Zalytron Corp.
(516) 747-3515, FAX (516) 294-1943

DESIGNING THE DIPOLE MONSTER, PART 1

By Timothy E. Sandrik

This is the first in a series of articles describing an exciting adventure in loudspeaker design. The Michigan Technological University student section of the Audio Engineering Society (AES) has a reputation as a very active organization, especially in the area of undergraduate research-and-development projects.

Numerous electronic and loudspeaker projects have served to give members of the section some hands-on experience to supplement their education. The "Loudspeaker Design Project," as members call it, has been on our wish list for several years, since our reference system needs new speakers. The Western Michigan AES Loudspeaker Design Challenge II, which took place in May 1997, provided a deadline for our efforts.

PHOTO 1: The builders of the system: from left to right, Ryan Mihelich, Tim Sandrik, Robert Lawson, and Brian Zaremba.

PHOTO 2: A view of the system as set up at the show.

COMPETITION GUIDELINES

The basic rules of the competition were simple. Each loudspeaker of the pair must weigh less than 100 lb. Filtering is limited to a two- or three-way passive network. Judging would proceed according to "AES recommended practice for professional audio—Subjective Evaluation of Loudspeakers," *JAES*, Vol. 44, No. 5, May, 1996. These guidelines, which outline standards for setting up and performing blind listening tests, were developed by respected members of the audio industry, including two of the judges, David Clark and Tom Nousaine.

In September, the rules seemed to outline simple designs: perhaps a three-way with a good midrange and tweeter, and possibly a 10″ for bass, or maybe an MTM for some extra efficiency. Of course, we assumed the design would require a thick, heavy cabinet that would use up a large portion of the allowed 100 lb. So our adventure had simple beginnings, but it really took off as the first question of "Where will we get the drivers?" was answered.

PHOTO 3: A view of the entire system setup.

We decided to aim high in looking for sponsorship, since we had the facilities of Michigan Technological University and Liberty Instruments' wonderful AudioSuite measurement package to aid us in our quest. Kimon Bellas of Orca Design and Manufacturing designs and/or distributes what we believe to be the best loudspeaker drivers available. Orca supplies Zalytron and Madisound with Focal, Cabasse, Accuton, and Raven transducers, as well as various other necessities for designing and assembling a good loudspeaker.

We asked Kimon for several drivers for a simple design. He agreed to provide us with whatever supplies would aid us in developing the best loudspeaker we could. This certainly opened the floodgates of creativity, and many wild concepts were sketched from week to week as we realized the potential of such a proposition. With great support from our first sponsor, we applied the "aim high" principle to finding the rest of the necessary materials.

ABOUT THE AUTHOR

Tim Sandrik recently graduated from Michigan Technological University with a BSEE. At MTU, he was an active member and officer of the AES student section and a founding member of the 306WD engineering team. He is currently employed at Klipsch, LLC, involved in professional and consumer loudspeaker research and development.

FILLING THE WISH LIST

You might guess what else would be on a wish list for such a project. Orca also supplied us with Black Hole damping materials, Axon capacitors and resistors, and binding posts and floor spikes, as well as the opportunity to sample its entire stock of high-end drivers. North Creek Music supplied us with over 100 Zen, Harmony, and Crescendo capacitors to stock our crossover parts bin, and Renco Electronics, an inductor manufacturer, gave us a similar stock of air-cored inductors.

A visit to Bill Dudleston at Legacy Audio provided us with advice, encouragement, and a few big iron-cored inductors for dipole baffle compensation. Baltek Corporation supplied us with a space-age composite called Decolite, which is the primary structural component of the speaker, while Analog Devices furnished some of its newest ADXL150, ADXL50, and ADXL05 accelerometers to optimize its use. E-A-R Specialty Composites supplied us with various acoustical foam samples and a marvelous damping foam called Confor.

Finally, Wadia loaned us a brand-new model 860 CD source; Audire Electronics came up with a Parlando Class-A amp and a Diffet 5 balanced preamp; and Synergistic Research supplied balanced interconnects, speaker cables, and power cords to demo the system. The system's builders appear in *Photo 1*. *Photos 2* and *3* show the system as we had it set up in San Francisco's Westin St. Francis, the site of Stereophile's '97 hi-fi show.

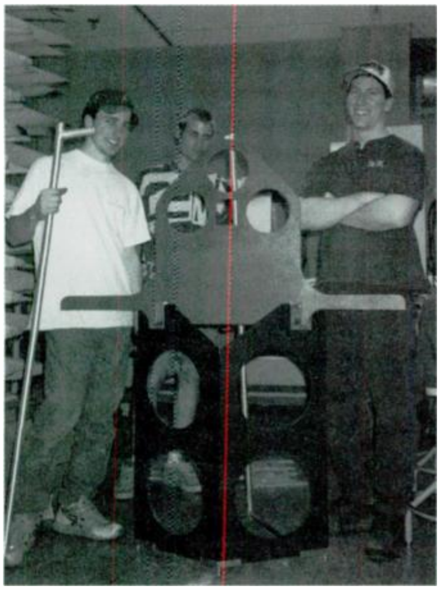

PHOTO 4: The finished baffle.

BEHOLD—A MONSTER

What we ended up with is a monster. The final iteration is a hybrid dipole design that measures four feet wide by nearly five feet tall at the extremes. *Photo 4* shows the finished baffle. The solid portion of the baffle is made up of a very light and stiff composite material called Decolite, which is used in the new Corvette, and a layer of Black Hole Pad to control vibrations.

The baffle also has a nonrigid portion that serves to increase low-frequency efficiency while adding very little weight to the speaker. The baffle does not stand unassisted. A welded and machined aluminum support framework completes the structure. The baffle, even with the support structure, weighs less than 12 lb, leaving a lot of leeway for the weight of drivers, crossover, and wire.

How could we even approach 100 lb with a baffle weighing less than 12 lb? Well, each speaker contained four Focal 12V726S 12″ woofers, three Focal Audiom 7k midranges, and a Raven R1 tweeter. Sound expensive? These drivers are valued at over $1,500 per speaker, and make up most of the weight.

The crossover (*Fig. 1*) is complex for a three-way, including Axon and Zen capacitors, Axon resistors, and inductors from

Renco Electronics to perform minimum-phase high-order filtering, dipole baffle compensation, and midrange attenuation and shaping. *Table 1* is the crossover parts list. *Photo 5* shows Brian Zaremba with one of the partially assembled crossovers. The result is very flat on- and off-axis response, with a highly optimized radiation pattern.

ABUNDANT HELP

How did we get from MTM to monster? We didn't do it alone; all of the mentioned parts and many more were donated by the generous sponsors previously mentioned. Without them and the opportunities they provided, we would never have explored some of the concepts. Even with such generous sponsors, there was a lot of work to be done.

We spent endless hours at the library with AES journals and acoustics texts. We developed models to simulate multidriver arrays of loudspeakers, and countless prototypes to test concepts. *Photos 6–9* illustrate some of these prototypes. The most dramatic step was abandoning the idea of a box, and researching the benefits of dipole designs and minimizing room interaction.

The project was not only a good research, design, and development experience, but also a good way to become acquainted with

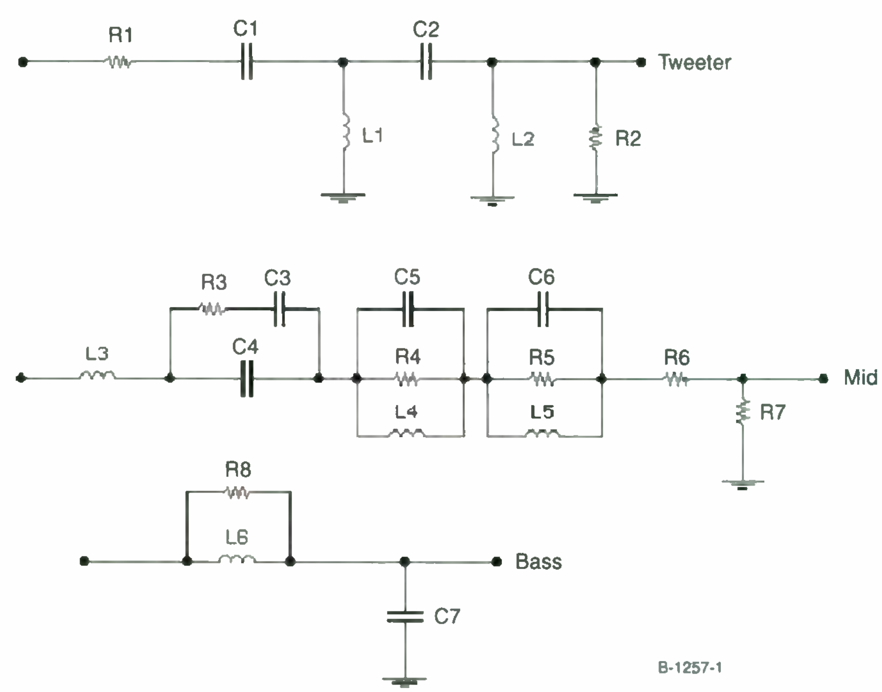

FIGURE 1: Crossover schematic.

the audio industry. As mentioned before, we not only enjoyed the help of many sponsors in the audio industry, but also had the opportunity to interact with other industry members during development, at the competition, and at the Stereophile '97 trade show.

TABLE 1

CROSSOVER PARTS LIST

Resistors	
R1, R2, R7	6Ω
R3	30Ω
R4, R5	15Ω
R6	2.5Ω
R8	22Ω
Capacitors	
C1	3.9μF
C2, C5	30μF
C3	10μF
C4	6μF
C6	4.7μF
C7	300μF
Inductors	
L1	1.5mH
L2	3.0mH
L3	1.0mH
L4	0.5mH
L5	0.25mH
L6	18mH

COMPETITION

The competition took place in Grand Rapids, MI, in a large conference room of the Van Andel Museum. The competing speakers were all within a "U" formed by a large theater curtain placed to partially absorb radiation to the wall behind and to the sides. We had the opportunity to set them up to ensure their best performance.

We then left the museum for lunch and an afternoon of anxious anticipation of the results. Our competitors had some interesting ideas, and we saw what we feared most—a well-executed conventional design with a big box. We were afraid of that because a key sponsor failed to provide us with promised air-cored inductors, which were necessary to solve a saturation problem our iron-cored inductors were having. At just over 100dB (at 2m), our speaker suffered saturation so bad it sounded like hard amplifier clipping. This is obviously undesirable in any speaker, let alone one with so much invested in materials.

PHOTO 5: Partially assembled crossover.

As it turned out, we came in second to the conventional system by two points. The score sheets reflected that more than seven points were lost due to a lack of dynamic range. Mr. Clark and Mr. Nousaine, the judges of the competition, also noted what seemed to be a phase problem at the top end.

We found later that a tweeter had been blown and a thick piece of felt, which normally lay beside the tweeter to shape the response and radiation pattern, had been misaligned, placing it directly in the path between the tweeter and the listeners. We suspect that these mishaps may have given the impression of a phase problem between speakers.

To tell the truth, we weren't that disappointed; take another look at *Photo 1*, which shows the speaker builders at the competition after we got the news—and we're all still smiling. It was a great learning experience for us all, certainly for myself. Luckily, our adventure still was not over.

OPEN YOUR GOLDEN GATE

Getting the speakers to San Francisco for the Stereophile show in the Westin St. Francis was an adventure in itself. One box did not arrive with the other four, sending three overdressed kids into a frenzy. All of the parts finally arrived, but broken and bruised, and we proceeded to collect another Parlando amp and Diffet preamp from our friends at Synergistic Research.

We returned to our room and literally tore

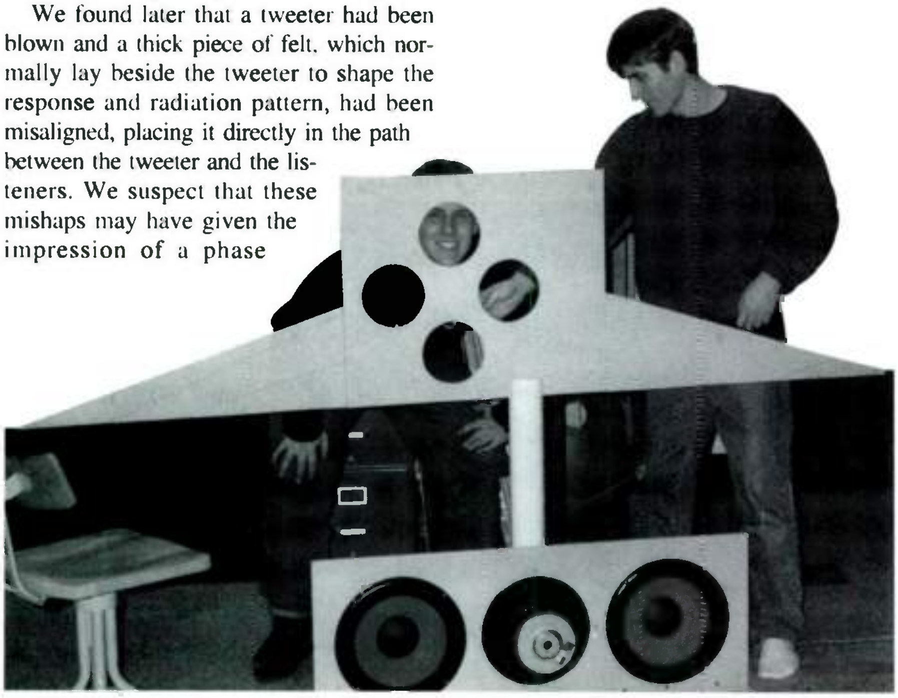

PHOTO 6: A unique concept prototype of the system.

PHOTO 7: A floor-model prototype.

it apart. Tables went into the shower, one bed behind the listening chair, and the other on the open wall of the room. Then we proceeded to unpack Wadia's model 860 CD source. We placed flyers and information about the project, our organization, and our sponsors by the door, and music filled the room. Wow, did it sound good (tweeter now fixed)! Ask anyone at the show.

In subsequent parts of this article, I'll describe the design of the driver array, the filtering and acoustical measurements, and the design and construction of the ultralight baffle. We'd like to thank our sponsors again, especially Kimon, Patti, Jesse, and Chris at Orca.

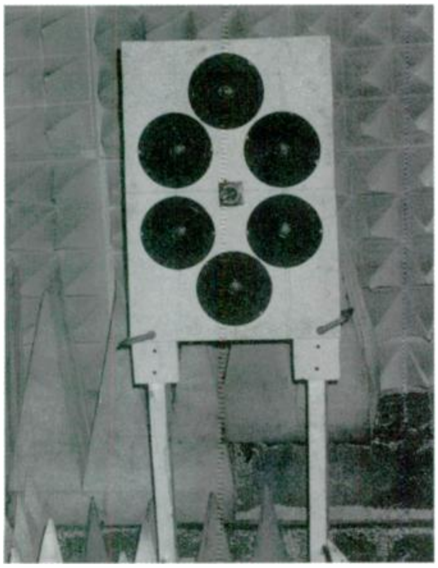

PHOTO 8: A six-woofer prototype of the system.

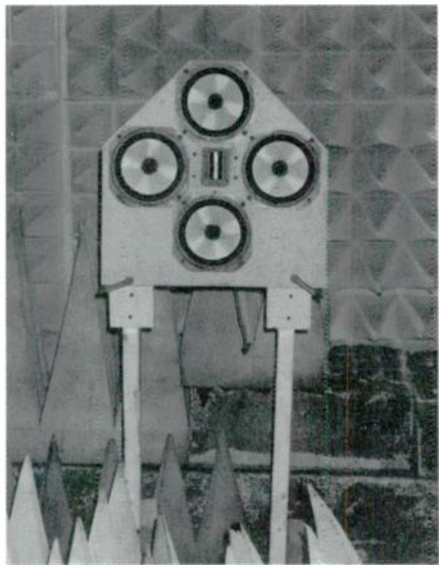

PHOTO 9: Another prototype of the system.

TESTING FRONT-PANEL DAMPING MATERIALS

By G.R. Koonce
with assistance from Kim Girardin

Many years ago I discovered by listening that adding a grille frame and grille cloth to a system could really mess up the high-frequency sound quality. The system would be clean without the grille assembly, but when the grille was added the highs developed a "distorted" sound. I'm sure many readers have noted this effect. At the time I had no test capability that would show the cause, but Liberty Audiosuite now has the capability of generating a signal's power cepstrum which will show the presence of delayed echoes in that signal.

Work showing the effects of a grille assembly on generating delayed echoes was included in "A Modest-Cost Three-Way Speaker System" (*SB* 8/96, p. 34). This work showed via power cepstrum plots that when a grille frame and cloth are added in front of a clean tweeter, delayed echoes are added to the signal. These echoes seemed to be mainly a function of the grille frame, but did become worse as the grille cloth was made more dense. In addition, the work demonstrated that adding a coating of some damping material to the front panel would greatly reduce these effects and, thus, preserve the quality of the highs.

It has been my practice for many years to cover the front panel with a layer of ½″-thick fiberglass. This material is available in widths of 3″ and 6″ for wrapping around pipes to insulate them. I had selected it because it was cheap, available, easy to work with, and—according to listening tests—greatly improved the sound quality of systems to which it was applied.

However, testing had shown that the fiberglass itself could generate echoes if it stood high enough that the tweeter output would "see" it. This raised the question of whether there was a better low-cost material to use on the front panel to solve the problem. This article reports the results of testing a variety of

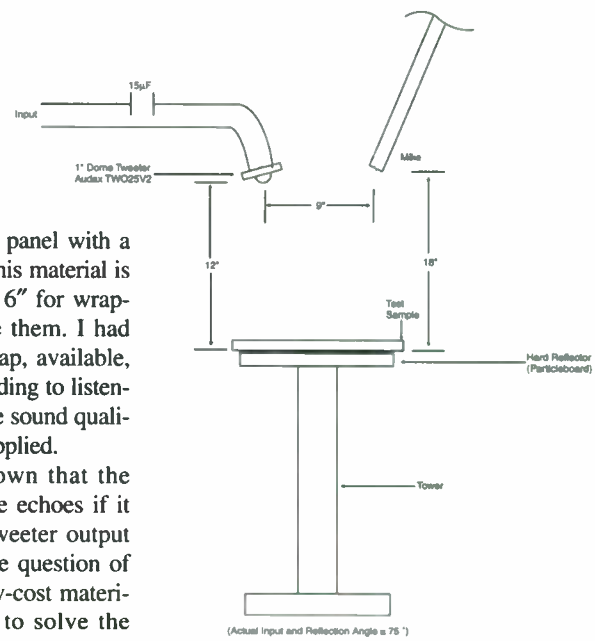

FIGURE 1: Test setup for reflection tests.

TABLE 1

DAMPING MATERIALS TESTED

#	PRODUCT	DESCRIPTION	SOURCE	COST (APPROX.)
1	Felt Gard	4½″ × 6″ self-adhesive felt pad about 3/16″ thick	B	$5.50/pr.
2	Mouse pad	rubber pad with fabric covering (blue) on one side	A (Catalog #130-382)	$2
3	Black latex-backed cabinet carpet	About 1/8″ thick	A (Catalog #260-762)	$5/36″ × 48″
4	Foam slip-on pipe insulation	For pipes, about ¾″ diameter, 3/8″ thickness	B	Cost not recorded, but low
5	Black automotive carpet	About ¼″ thick and made of polyolefin fiber	A (Catalog #261-700)	$6/36″ × 40″
6	Thick acoustic foam	Overall thickness is 1.5″	B (Catalog #260-316)	$5/24″ × 36″
7	Fiberglass	6″ wide × ½″ thick, for pipe wrapping	D	$7/35′ roll
8	Strange white foam material	Do not know what it is made of, but it is flexible, compliant, and really absorbs impact energy, but unfortunately not sonic energy	I've had it around for years	
9	Duct-liner material	About ½″ overall thickness composed of medium-dense fiberglass bonded to a 1/16″-thick layer of black backing that seems to be high-density fiberglass; used in heater/AC ducts to quiet them	Supplied by Kim	
10	Acoustical fiberglass	Stiff, very dense at 7 lbs/ft^3, 5/8″-thick, and smooth on one side with a dot texture on the other	E	
11	4 oz polyester batting material	A white fibrous material about ¾″ thick	C	$0.70/ft × 48″ wide
12	10 oz polyester batting material	A white fibrous material about 1¼″ thick	C	$1.35/ft × 48″ wide
13	Green high-density foam rubber pad	½″ thick	C	$1.80/ft × 24″ wide
14	White low-density foam rubber pad	½″ thick, of less weight per unit area than #13	C	$1.50/ft × 24″ wide

A—Available from Parts Express
B—Available from your local ACE hardware store
C—Available from your local Jo-Ann Fabrics store
D—Wrap-on Co., Inc., 341 W. Superior St., Chicago, IL 60810
E—The La Crosse Acoustical Tile Company, 1541 Miller St., La Crosse, WI (608) 784-3787

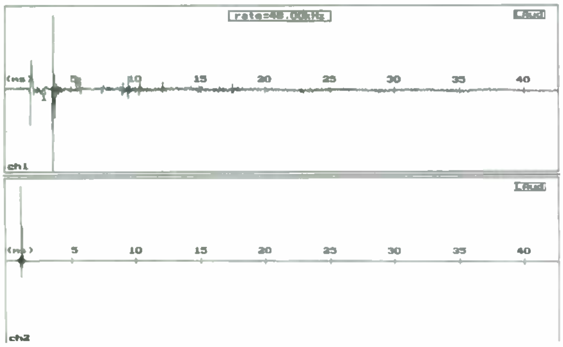

FIGURE 2: Audiosuite time responses.

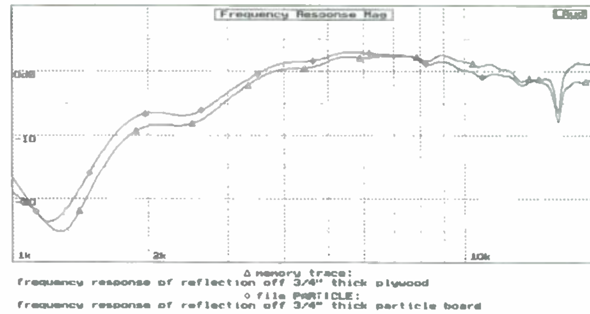

FIGURE 3: Plywood and particleboard reflections.

materials for possible front-panel damping application.

MATERIAL REQUIREMENTS

What is desired is a material that prevents the presence of a grille frame or grille cloth from generating delayed echoes. The material must thus prevent any sound energy reflected back from the grille cloth from being re-echoed back to the listener. It should keep hard objects, such as the grille frame, from "seeing" sound energy to diffract or echo back. This means that the damping material has two requirements. First, it must not itself reflect sound; i.e., it must absorb sound energy. Second, it must provide sufficient attenuation of sound energy passing through it to prevent hard objects (front panel, grille frame, etc.) from reflecting enough sound energy to cause trouble.

The first requirement should be relatively independent of the thickness of the material, while the second requirement will most likely improve as the material gets thicker. I believe you do not want to place the grille cloth any farther from the tweeter than absolutely necessary. I normally build with the cloth about ½–¾″ ahead of the front panel. Thus the damping material thickness is limited to this range.

As indicated in the *SB* 8/96 article, it is desirable to keep the front panel damping material relatively thin at least near the tweeter, or the edge of the material itself can become a source of destructive echoes. For that testing, I used 1″-thick damping material, which is thicker than I have ever used in construction.

During this work I compared notes with Kim Girardin, who was also interested in this topic. Kim also undertook some testing of various materials on his own. To ensure consistency in the results, Kim sent me samples of the better materials he had found so they could be tested in the same setup for a direct comparison with my materials.

TESTING APPROACH

Just how to test front panel damping materi-

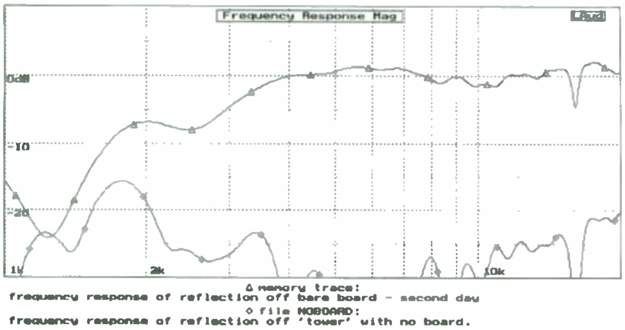

FIGURE 4: Reflections off bare reflector board and bare tower.

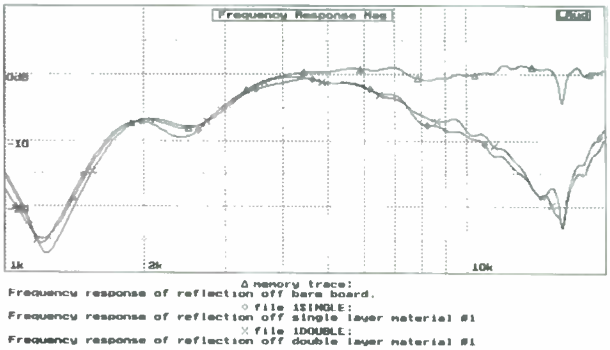

FIGURE 5: Reflections with material #1—felt pad.

als is a problem. Ideally, you would build a system, try all the materials on the front panel, and see what offered the best sound. This would be a major effort and would require enough damping material of each type to cover the entire front panel; also, the results would not yield any quantitative data. I decided to try to make a direct measurement of the sound reflected from a hard panel covered by each damping material. It is likely that the performance of each material could vary with the incident angle of the sound energy, but this was not investigated. All the testing was for sound incident at about 75° and an echo returned at about this same angle.

Figure 1 shows the test setup used. The reflector is a small piece of material set on a 36″ long 3″ OD tube (the tower) to distance it from floor reflections. The microphone hangs down over the reflector and is kept slightly above the source, which is an Audax TWO25V2 1″ dome tweeter suspended over the reflector.

Using Liberty Audiosuite, an MLS signal pulse is fed to the tweeter through a 15µF capacitor to limit the low-frequency content to the tweeter. There is a direct signal path from the tweeter to the microphone, and also a path from the tweeter to the reflector and back to the microphone. The "windowing" capability of Audiosuite can be used to separate these two signals so that the frequency response of the echo off the reflector can be established. If all positions and system gains are kept constant, then damping materials can be placed on the reflector; the change in level at each frequency in the reflected echo's measured response tells just how effective the material is in preventing echoes from the reflector.

If the damping material itself reflects a lot of sound energy (failing requirement 1), there will be little or no improvement. If the damping material is not good at attenuating sound passing through it (failing requirement 2), then again there will be little improvement—we will see the hard reflector

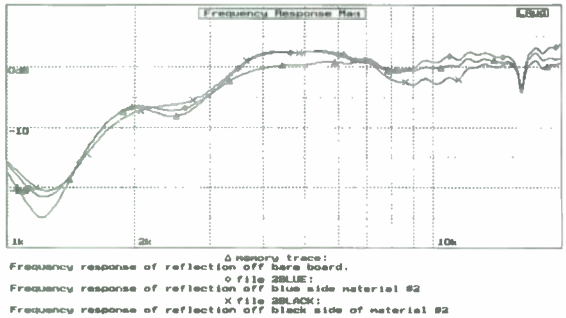

FIGURE 6: Material #2—mouse pad.

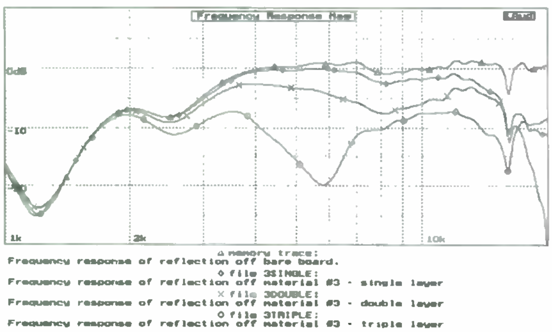

FIGURE 7: Material #3—latex-backed carpet.

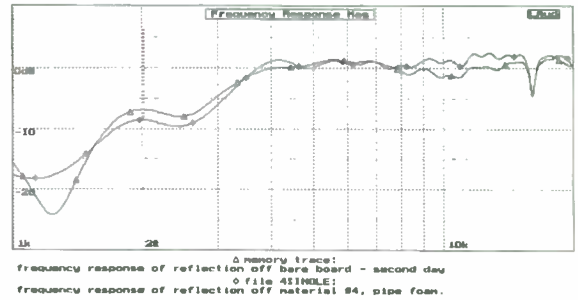

FIGURE 8: Material #4—pipe foam.

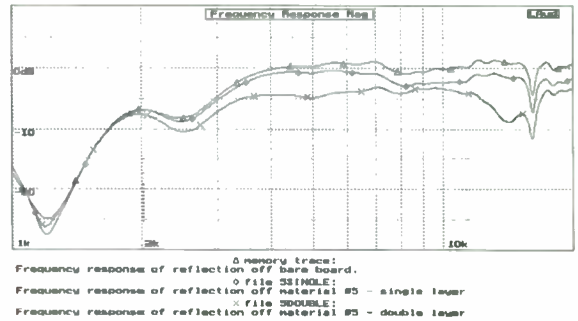

FIGURE 9: Material #5—polyolefin fiber carpet.

right through the damping material. Thus, for a material to show a major reduction in reflected signal in this severe test, it must be good in both of our requirements.

Figure 2 shows the time waveform displayed by Liberty Audiosuite for this test setup. The bottom trace is the electrical impulse fed to the tweeter through the crossover capacitor. The top trace shows the microphone output. The direct tweeter-to-microphone impulse is shown just past the electrical impulse; some time after this

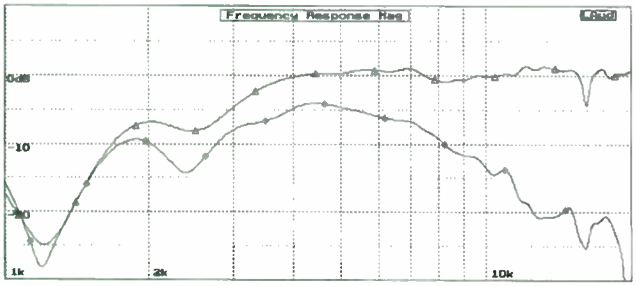

FIGURE 10: Material #6—thick acoustic foam.

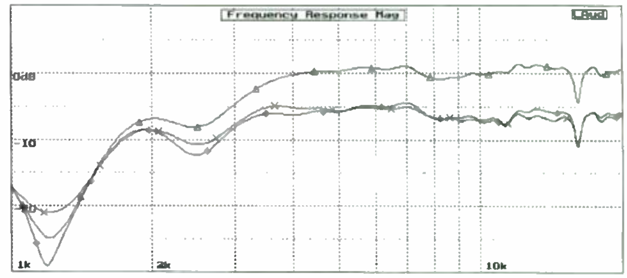

FIGURE 11: Material #7—½″ thick fiberglass.

the larger reflected impulse occurs.

The two time markers (small triangles labeled 1 and 2) show the time interval that was FFTed to give the frequency response. This time was selected to avoid the direct signal, but start early enough to pass echoes from the damping material that will start ahead of the hard surface reflector due to the shorter path length. Not all this testing was done in a single session, and the test setup could not be duplicated exactly. Thus all plots will show the frequency response of the hard reflector as a reference. It is then the difference between these two curves that will represent the improvement provided by that damping material.

PLYWOOD VS PARTICLEBOARD

One question that I had always wondered about was whether plywood had the same high-frequency reflection performance as particleboard. *Figure 3* shows the frequency response of the reflected signal for roughly the same-sized plywood (6.75″ × 6.25″) and particleboard (7.125″ × 6.25″) reflectors, both ¾″ thick. It is clear that throughout the useful frequency range of a tweeter they are about equivalent. The reflected signal is also a very good copy of the on-axis response of this 4Ω dome tweeter driven through a 15µF capacitor. Even the dip around 16kHz, typical of many 1″ domes, is fully preserved in this reflected signal.

The indicated rise in response down near 1kHz is the result of noise in the test room mainly from the computer fan, even though I set it to a slower speed than normal during a test. While the particleboard looks to be a slightly better reflector in the range near 20kHz, it is clear that both materials are very good reflectors from at least 2kHz upward.

For the testing of various damping materials, the hard reflector was always a ¾″-thick piece of particleboard (8½″ × 5¼″). The various damping materials tested are shown in *Table 1*. Most samples were cut to a size of 9″ × 6″ so they would fully cover the hard

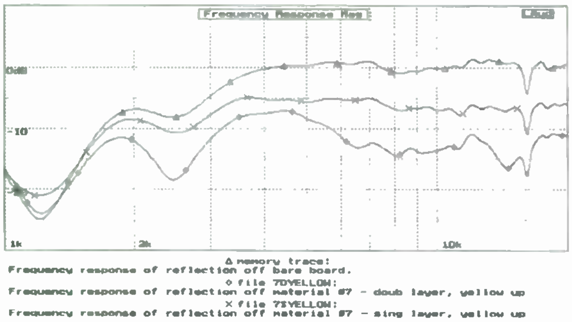

FIGURE 12: Material #7—single and double layers.

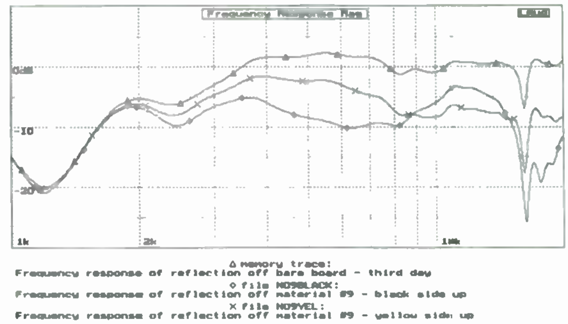

FIGURE 14: Material #9—duct-damping material.

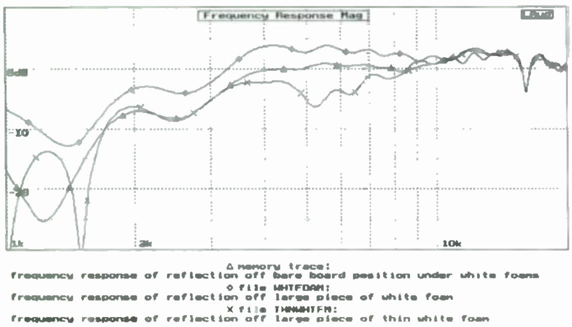

FIGURE 13: Material #8—white foam.

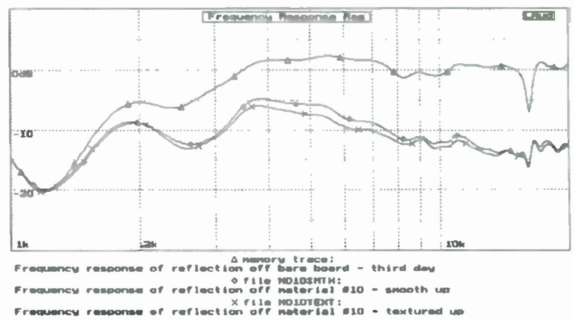

FIGURE 15: Material #10—stiff dense fiberglass.

reflector with a small overlap on each side. I will indicate occasions when the damping material sample was a different size.

One of the first questions to be answered is: "Does the test setup work as expected; i.e., is the signal reflected from the hard reflector or from various other objects in the region of the test?" *Figure 4* supplies the answer, comparing the results for the hard reflector versus those for the tower standing bare with the reflector board removed. It is clear that from 3kHz upward

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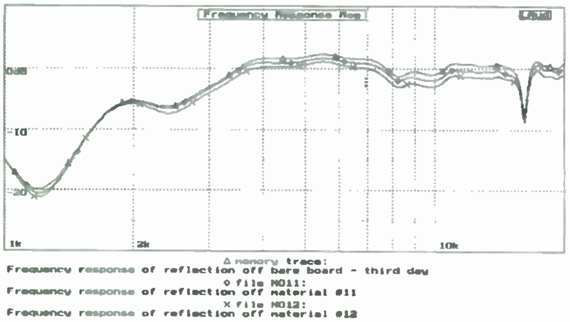

FIGURE 16: Materials #11 and #12—polyester batting.

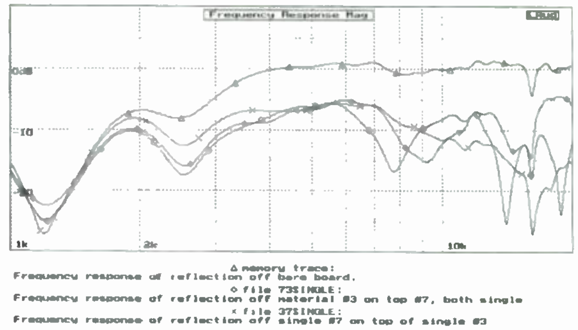

FIGURE 18: Combinations of materials #3 and #7.

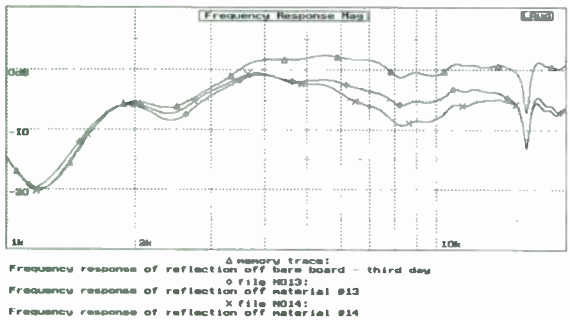

FIGURE 17: Materials #13 and #14—foam rubber mat.

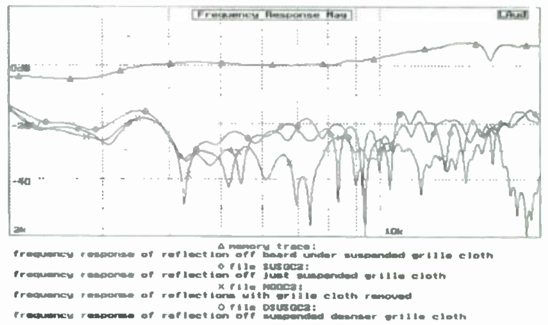

FIGURE 19: Reflections off grille cloth.

the major energy comes from the hard reflector board, since the background level is down at least 20dB.

MATERIAL RESULTS

Figure 5 shows the results for the felt pads, material #1. I had great hopes for this material. It is quite effective at high frequencies and may work well around a super-tweeter. The fact that two layers behave the same as one is not surprising because the material is composed of three layers. The top half appears to be soft felt, below which is an equally thick layer that looks like more densely packed felt. Finally, at the back, is a stick-on adhesive layer. Certainly, for use around the midrange driver or the bottom end of a tweeter's response, this material would not be effective.

One inexpensive material I thought might work is a mouse pad. The one tested (material #2) is rubber with a thin fabric layer (blue) on the top. As *Fig. 6* shows, it is about as good a reflector as particleboard and thus useless as a front-panel damping material.

Material #3, the latex-backed carpet, is rather thin (1/8″) and was tested in single-, double-, and triple-thick layers (*Fig. 7*). This material is relatively effective in double- or triple-thickness and would look quite good on the front panel. It is available in black (tested) or charcoal, easy to work with, and relatively inexpensive.

Material #4 is a foam pipe insulation that was a pain to test. It comes in long tube shapes with a cut along its entire length, so that you can slip it over pipes. I cut off two sections, pressed them nearly flat, and attached them to the hard reflector board with double-sided tape. I had initially thought the material would be ideal to cut into quarters of a circle and use to mask the junction of the grille frame with the front panel. Unfortunately, as *Fig. 8* shows, the material reflects about as well as the particleboard and is thus useless for a front-panel damping application.

Material #5 is an inexpensive carpet-like material that comes in a variety of colors. The catalog says it is composed of 100% polyolefin fiber. It measures about ¼″ thick so use would be limited to single, or possibly double, layers. *Figure 9* demonstrates that even in double thickness the material is good for less than 5dB and is not very effective.

Acoustic foam, material #6, is not practical for front-panel damping since it has a maximum thickness of 1.5″. I wanted to test its effectiveness anyhow. It is a foam material sculptured like the walls of an anechoic chamber. It is likely to perform well because it will have a scattering effect on the incident sound energy. It proves very effective at high frequencies, but only provides about 5dB at 5kHz and below (*Fig. 10*). This performance is not impressive for its thickness and surface shaping.

Material #7 is the ½″-thick fiberglass that I have used for many years. The material looks different when viewed from each side and I have often wondered whether it was more effective with the white or yellow side facing out. *Figure 11* concludes that the performance of a single layer is the same facing either way and is good for 5dB from about 4kHz upwards.

Figure 12 shows that a double layer is quite effective, providing about 12dB above 6kHz and about 7dB down to 3kHz. I used this thickness for the *SB* 8/96 article, and it showed good suppression of echoes in most tests. The 5dB result may not sound impressive, but this is a very severe test with the tweeter directly facing the material. Past experience has shown that a ½″ (single) layer of this material is sufficient to do the job.

I do not know exactly what material #8 is. It is a white, pliable foam material that is used in packaging applications. It absorbs impact very well—you can punch a ½″-thick layer of this material held against a hard surface without hurting your hand. Would it also be effective against sonic energy? *Figure 13* shows the answer is "no." The ½″-thick test sample measures 15″ × 9½″, and responds more poorly than the smaller particleboard reflector. The thin (1/8″) material was cut to normal sample size. Further testing was unnecessary since it did not work well for tweeter frequencies.

Material #9 is the fiberglass duct damping material composed of a thin, but very dense, black side bonded to a thicker medium-density yellow side. *Figure 14* demonstrates that the black side is very effective from 4kHz–8kHz (about 10dB for ½″ thickness), but unfortunately drops to 3dB around 11kHz. The yellow side shows less attenuation at low frequencies, but is still quite effective over the frequency range of a tweeter. This material is a candidate that should be considered.

Material #10 is a 5/8″-thick dense fiberglass-looking material. *Figure 15* reveals that it is the best material tested of the group. It provides about 5dB from 2.5kHz and up, with at least 10dB above 6kHz. It is stiff and may be tough to work with, but surely warrants a try on an actual system.

Materials #11 and #12 are polyester batting, a white fibrous material recommended by a friend. I thought they had good potential, but as *Fig. 16* shows they offer no reduction in reflected sound energy throughout the test frequency range, although they are ¾″- and 1¼″-thick, respectively. I did not test to see whether they were transparent to sonic energy or whether they simply reflected the energy themselves.

Materials #13 and #14 are foam rubber mats. *Figure 17* shows the ½″-thick high-density mat (#13) provides about 4dB from 5kHz and above. The ½″-thick low-density mat (#14) is about the same at the ends of the frequency range, but is better from 6–15kHz. These materials are inexpensive and easy to work with, but probably won't look much better behind a dark grille cloth than the yellow fiberglass. The low-density mat warrants a try.

Of the 14 materials tested in this work, the fiberglass materials seem to be the best. The best single material tested was the hard, dense fiberglass material #10. This material is 5/8″-thick and unfortunately still looks like fiberglass! The 1/8″-thick carpet material (#3) showed some merit and is nicer-looking. I wondered if it could be combined with the ½″ fiberglass (#7) for improved performance with acceptable thickness.

Figure 18 depicts various combinations. They all respond about the same as the bare fiberglass at low frequency, but improve above 7kHz and surely would look better if the thickness is acceptable. The other materials that showed some promise were the ½″ foam rubber (#14) and, for higher frequencies, the felt (#1). Foam material #6 was very effective at high frequencies, but much too thick for front-panel damping. I'm sure thin foam materials that would be effective are available, however we did not identify any in these test sessions.

GRILLE CLOTH EFFECTS

One other question I had was: "How well does grille cloth itself reflect sound energy?" I decided to test this with the same setup. I tried a cloth-covered grille frame, but the major reflection appeared to come off the frame rather than the cloth. I then stretched a piece of thin black grille cloth (Radio Shack # 40-1935) over the tower and hard reflector using four clip leads to nearby objects. I then tested this setup with the reflector present to provide a reference, and with the reflector and tower removed leaving just the stretched cloth.

Finally, I repeated the test to get a background level (*Fig. 19*) with a denser grille cloth and then with the grille cloth removed. The two grille cloth samples do reflect energy above the background level at frequencies above 5kHz. Note the scale on this figure has been changed to 10dB per major division. The denser grille cloth reflects more than the Radio Shack cloth, but both are down at least 20dB from the reference of having the reflector behind the cloth.

This tends to indicate that the major adverse effect of having a grille comes from the frame, not reflections from the cloth. This supports the work in the *SB* 8/96 article, but does not always agree with what I have heard. In some cases the sound was

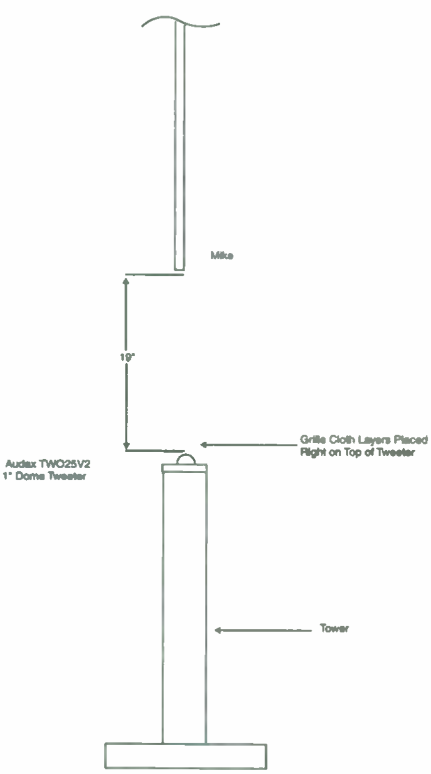

FIGURE 20: Test setup for grille cloth loss measurement.

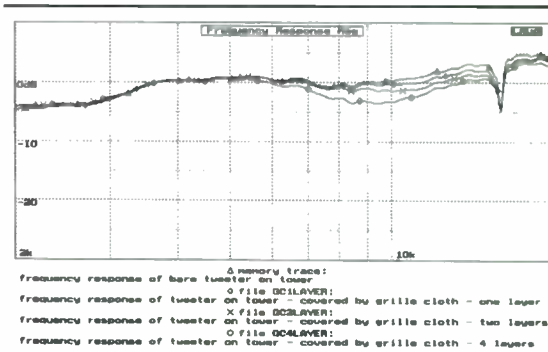

FIGURE 21: Grille cloth loss with dome tweeter.

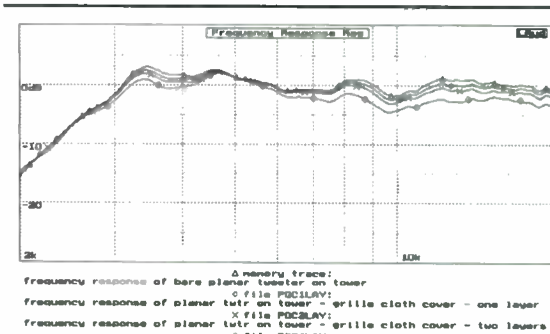

FIGURE 22: Grille cloth loss with planar tweeter.

acceptable with the frame installed, but deteriorated when the cloth was added. I believe there is more to this problem than the above tests measure.

Previous testing had always indicated that the addition of grille cloth in front of a tweeter caused little attenuation, which would agree with the fact the grille cloth reflects very little energy as indicated earlier. I decided to try to directly measure this attenuation. I modified the test setup as shown in *Fig. 20*, using a different sample of the Audax dome tweeter set upon the tower and facing directly toward the microphone, thus measuring the attenuation through the grille cloth at 90° to the plane of the cloth.

Figure 21 shows the measured frequency response of this tweeter with no, one, two, and four layers of the Radio Shack grille cloth covering the tweeter. I placed this cloth directly on top of the tweeter which has a plastic protector over the dome. Below about 5kHz the loss is very low, but by 10kHz it approaches about 1dB per layer.

Figure 22 is a rerun of this same test using a planar (leaf) tweeter. The crossover capacitor was changed to 7.5µF since this is an 8Ω tweeter. The results are slightly different in the 3–5kHz range, possibly because of loading on the diaphragm due to the grille cloth. Above 6kHz the results confirm the *Fig. 21* results, showing about 1dB per layer at the higher frequencies. You must consider even this small amount of attenuation when balancing your system.

SUMMARY

In this article, I reported on the results of studying 14 materials for covering speaker system front panels to prevent delayed echoes of the tweeter output. Felt, foam rubber, carpet, and fiberglass materials seem to show promise if the proper form of each material can be found. The ½" fiberglass covering I have used for years seems to be as good as most materials tested, but not as good as the 5/8"-thick rigid fiberglass (#10). Many materials were about as bad as leaving the front panel bare. Also, the need for front-panel damping seems independent of whether the panel is plywood or particleboard.

This and earlier work seem to indicate that the real problem is caused by the grille frame and not the grille cloth itself. This is not always what I have heard, so some question is cast on the test methodology. The testing has been of anechoic performance on one axis, while listening, of course, involves all the radiated power. It thus appears that you should cover the front panel in the area of the tweeter, possibly the midrange, and certainly any items the tweeter output might reflect, objects such as other driver frames and, mainly, the grille frame.

The testing also indicates that thin grille cloth reflects less than 1% of the energy striking it at an angle of about 75° to the plane of the cloth. Also such grille cloth seems to have an attenuation of about 1dB per layer at high frequencies when placed right in front of the tweeter. While this amount of attenuation may still need to be accounted for when setting up your system, it seems clear that the major variation in system response you sometimes see in test reports where a grille is added is caused by other effects.

Many thanks to Kim Girardin for his comments and assistance with this work.

SOURCES

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Product Review

THE TACT RCS 2.2 DIGITAL ROOM CORRECTION SYSTEM

Reviewed by Bill Waslo

The TACT RCS 2.2 is a DSP-based room and loudspeaker equalization system intended for use in two-channel high-fidelity installations (*Photo 1*). In addition to its nominal function as a frequency-domain equalizer, the TACT 2.2 also is designed to provide:

- a digital electronic crossover for splitting signals between main speakers and stereo woofers or subwoofers (with separate equalization for each subwoofer). You can choose crossover orders up to 60dB/octave.
- alignment of all four speakers to compensate for differences in signal arrival times at the listening position that might be caused by unequal distances to the left and right speakers or to the subwoofers. An advantage of this feature is that you can then place the subwoofers for best room loading independent of concerns about time-delay differences with the main speakers.
- preamplifier function with remote volume and balance control (in half dB increments) and selection between one analog or three digital inputs (*Photo 2*). S/PDIF, AES/EBU, and optical digital inputs are accommodated. You can also select different EQ settings via the remote, for different seating positions or desired response curves.
- automated frequency-response measurement and configuration equalization curve, using a calibrated microphone provided. (See this and other accessories in *Photo 3*.) The impulse response of your system is measured directly (by a series of averaged impulse stimuli applied to the speaker) into the TACT 2.2, and the correction factors are calculated using a user-supplied PC-type computer that connects to the TACT via a serial port (*Photo 2*).

PRACTICAL EQ APPROACH

The concept of digital equalization seemed to me quite enticing. It should be possible with such an approach to remove the effects of both loudspeaker irregularities and room reflections, leaving a flat, echo-free, time-coherent response for much of the band.

PHOTO 1: TACT RCS 2.2 Digital Room Correction System. Front view.

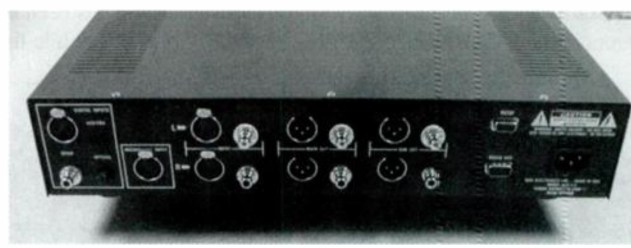


PHOTO 2: TACT RCS 2.2. Rear panel, input/output.

But there would seem to be many practical problems in doing so. For instance, anyone who has measured an in-room response of a loudspeaker with fine frequency resolution has found that the result is a strong function of just where in the room the measurement is made—move the microphone only a small distance, and the measured response at a single frequency point can change dramatically. A correction that is high in frequency resolution would be usable only if the listener's head is restricted to a quite small region of the room; outside that area the correction might easily become a contamination much worse than the original room response.

Response dips due to reflections tend to be much more severe in magnitude than response peaks. For instance, two equal amplitude tones of the same frequency summed in random phase can cause at most a 6dB increase (a doubling of acoustic pressure). But should the phase of the two tones be exactly opposite, the result would be a complete cancellation, effectively a near "infinity-dB" decrease in pressure. And attempted compensation of such a notch might easily turn into unnatural, very high response peaks should any factors (such as a listener's posture) change.

Furthermore, the concept of filling in deep response cancellation notches of 30dB or more with increased application of signal power makes sense only if the speaker and amplifier system has that much spare output capability to offer, which is not true for any system I have available. Still, I was excited by the idea of hearing how such an idealized response would sound. I have experimented with loudspeaker arrays aimed at reducing echo contamination and was interested to see whether the audible effect from a DSP-based high-resolution equalization would be similar.

But TACT has taken a more practical approach in the design of the RCS 2.2, resulting in some disappointment to me and my curiosity. For audiophiles or system designers seeking a practical equalization device and crossover, this is all for the better, however. Rather than aim for a conceptually pure (but functionally less usable) anechoic flat response for a single point in space, the TACT system performs more gentle response flattening, avoiding attempts to flatten narrowband, location-critical aberrations. It may not be desirable to remove room reflections or reverberation even if such a process should prove practical, for most recordings are likely monitored and mixed with the assumption that there will be some room reverberation during playback.

The result of equalization using TACT 2.2 is an in-room response that is flat (or matched to the selected target response) in a third- or sixth-octave sense. Of course, the benefit of performing this equalization via DSP rather than through contiguous bandbass filters (the analog "graphic equalizers" we all know but seldom use) is that the response flattening can be accomplished with great versatility and without introducing new group delay variations or bandpass ringing effects. Filters implemented with DSP are able to alter amplitude and phase characteristics relatively independently of each other.

TACT 2.2 is a quite ambitious design using four separate Motorola DSP chips and a PC-based software package. The unit is constructed in a chassis that is surprisingly heavy for a small-signal digital device, likely done with consideration for a target market that often looks favorably on features such as half-inch solid front panels. But there is no doubt that this is a solidly built device (*Photo 1*). (For a close-up look at the inside of this unit, see *Photos 4–8*.)

OPERABILITY

Though this review is meant mostly to be technical in nature, I have a few brief comments to offer about the operability and user friendliness of the system.

TACT has gone to much effort to provide

MANUFACTURER'S SPECS

Analog inputs	Stereo XLR & RCA
Input level	2.5V pk-pk
Input impedance	10k
Digital inputs	Coaxial (RCA) Balanced (XLR) Optical (TosLink)
Microphone input	XLR
Outputs	Full-range stereo RCA & XLR; stereo subwoofer RCA & XLR
Output level	2.5V pk-pk
System volume	+30dB
A/D converter	20 bit, 64x oversampling delta-sigma converter
D/A converter	20 bit, sign magnitude delta-sigma converter
THD+noise	0.003% closed loop 1kHz full-range input signal level
Dynamic range	104dB, 22kHz bandwidth, 1kHz –60dB below maximum input
Signal/noise ratio	104dB, 22kHz bandwidth, 1kHz at maximum output
Remote control	Yes

Manufacturer's suggested list price: $8,000.00

TACT Audio, Inc., 460 Sylvan Ave., Englewood Cliffs, NJ 07632, (201) 894-5905, FAX (201) 894-5736, http://www.tactaudio.com.

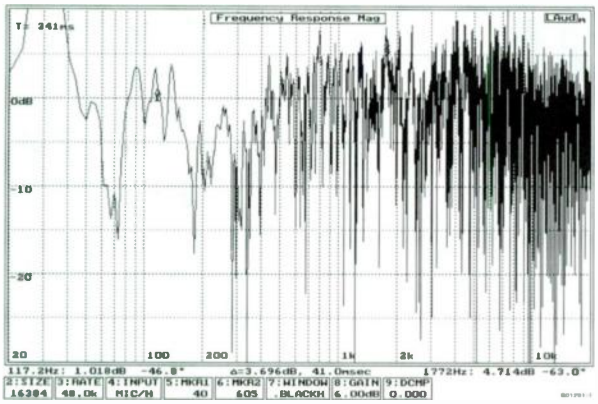


FIGURE 1: Loudspeaker, at listening position, before equalization.

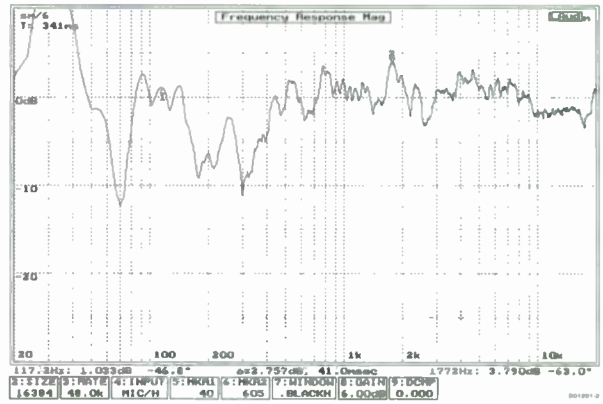


FIGURE 2: Same, 1/6 octave smoothed.

a simple setup of the system. I'm sure it was a real challenge to try to enable nontechnical audiophiles or dealers to configure computer hardware, measure loudspeaker and room characteristics, and correct the resulting response. In this effort, TACT has developed a well-written owner's manual (complete with an illustrated "click this, set that" Quick Installation Guide for those who are too impatient to read the rest of the manual) and a neatly organized Windows-based software package.

In general, the effort is successful. While I wouldn't suggest that someone who has never used a PC before could get the TACT 2.2 going, most users with access to perhaps a computer-wise friend or associate should have no major problems. And the computer is not required for subsequent use of the equalizer, only for initial setup (until you rearrange speakers or major furniture).

Not that there aren't some sticky points to the setup. I found that some of the display features on my portable computer's LCD screen were not completely compatible. For instance, one diagram which was intended to show energy flowing from speaker to mike, instead showed it flowing from a point between the speakers to nowhere in particular (on a normal picture-tube type of monitor, this animation displayed properly, however). And when I erroneously tried to change the name of an EQ setup in memory position 6 (when I had not loaded that position), the program "performed an illegal operation" and crashed; fortunately, I had saved to disk the measurement and EQ parameters that I had meticulously compiled before that point.

I was not particularly fond of the remote-control function. I had been looking forward to having a quick way to compare EQ-in versus EQ-out and to adjust volume and balance from my chair. But the unit provides almost no visual feedback. The small green display on the faceplate of the unit is not readable from any practical seating position more than about a meter away. So when you have been, for instance, punching the left/right buttons for balance and want to know where the setting is, you must keep count of button presses or get up from your chair and walk over to look at the display (which defeats the purpose of having a remote).

My biggest problem with the remote was its behavior when I selected the EQ memory settings. TACT lets you configure and store up to 12 different EQ settings, which is good. This lets you make different settings for different seats in the room or for different target EQ curves, even for different speakers. Memory setting 0 is "Bypass," good for before/after comparisons. But selecting these memory settings from the remote involves entering two numbers.

For instance, setting #1 is "01," a zero followed by a one. There seems to be a critical time period needed between the entry of the two digits. If you press "0," then "1" immediately, the system won't recognize the second digit, and then times out and discards the first digit. Same thing if you pause too long before pressing the second digit. I never quite got the hang of it and often had to walk over to the unit to read which setting had been accepted.

On the other hand, the volume-control function operated smoothly without clicks and always seemed clean at any setting that I used. The balance setting on the unit tracks with precision at any volume position, unlike the analog volume control on my preamp. I don't know how this volume control is implemented in the TACT unit, but I suspect that the adjustment is being made on the analog signal rather than by scaling the bits digitally (this latter method would reduce dynamic range at low volume settings).

The system performs room measurement using signal acquisitions from a microphone positioned at the listening chair with an impulsive stimulus driving the speaker. This technique can be sensitive to noise contamination, particularly in the lower frequencies, and TACT instructs you to average 25 or more of these acquisitions to minimize the effects of noise. Each acquisition is for 5000 time data points at a sample rate of 44.1kHz, with about a half second between clicks. I used 100 averaged acquisitions in my tests, and disabled all acoustic noise sources in the area (computer fans, VCR motor, furnace fan) for best results.

The average user isn't likely to go through the measurement process very often, so this time spent in measurement is no big deal, and the simple impulse method does work well if speed is neglected.

TACTFUL APPLICATION

I used TACT RCS 2.2 to correct the response on a set of internally triamped three-way speaker systems that I made a number of years ago. The plots given here are for the right speaker, which is positioned about 30″ from a wall at the right and about 20″ from a wall behind it (i.e., nearly in a corner). The other speaker was about equally ragged in measured response, but not as "interesting" a case for EQ as the right speaker. As you might expect, the after-EQ in-room responses of each speaker looked quite similar. Unfortunately, I was not able to test the crossover functions of the TACT (except electronically), as these speaker systems do not utilize separate subwoofer inputs or enclosures.

Figure 1 shows what RCS 2.2 is up against (all measurements shown were made using LAUD with MLS techniques). Clearly evident is a major resonant room peak at 30Hz, deep suckouts between 60–75Hz and between 180–400Hz (the floor-bounce or "Allison effect" range), and

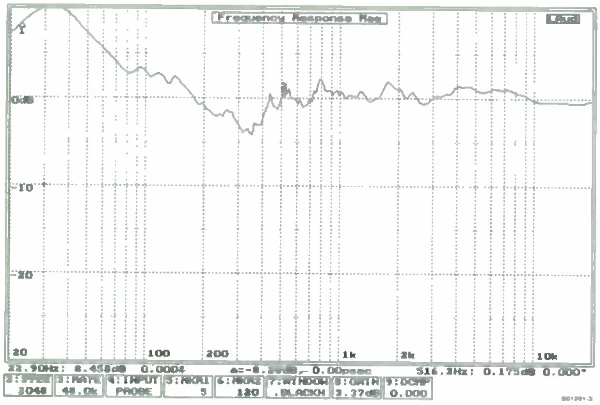


FIGURE 3: Central Spectrum of pre-EQ data.

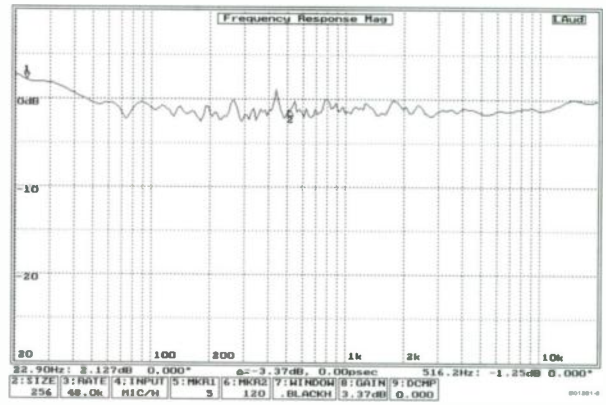


FIGURE 6: Central Spectrum of EQ'd response.

a multitude of sharp peaks and dips all the way into the higher frequencies of the spectrum. This picture highlights some of the problems involved in equalization of room-induced effects.

For instance, it would probably not be a good idea to try to add the more than 30dB of extra narrowband gain needed near 320Hz, where *Fig. 1* shows a narrow but deep cancellation notch—that would require the amplifier and speaker to deliver a thousand times more power at that frequency than before. And even if it were done, it would probably not help. The null is caused by cancellation, and will likely change dramatically with a small change in microphone (or listener) position. If equalized flat for the fixed microphone position, that point might show a 30+dB peak when the listener moved around. And of course, if cancellation were to be complete (zero output at that frequency), there is no amount of increased power application that could equalize it to flat.

And such total fine-resolution EQ is not really needed. While echoes from a speaker in a room are certainly audible, the raggedness of the response picture doesn't correspond to the perceived sound of these speakers (which, in fact, sound quite smooth). Human ears are apparently more forgiving than FFT analyzers of narrow frequency-response irregularities from room reflections. Perhaps more relevant is the curve in *Fig. 2*, which shows the same speaker response data with 1/6 octave smoothing applied. *Figure 3* shows this response processed per a mathematical model of human hearing (James Kates' "Central Spectrum," intended to indicate perceived coloration of wideband spec-

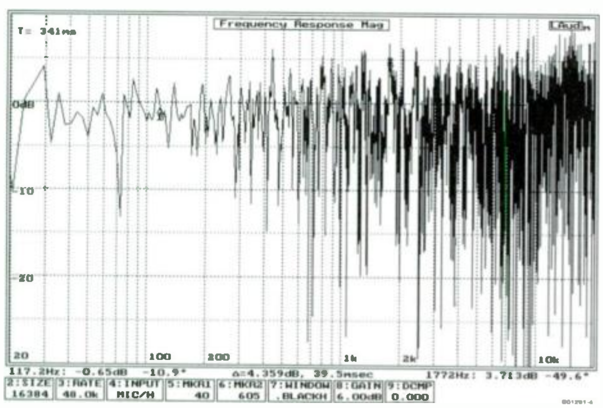


FIGURE 4: Unsmoothed in-room response after EQ.

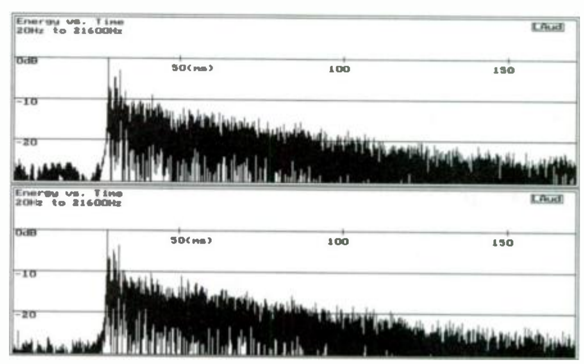


FIGURE 7: Energy-time curves without TACT EQ (above) and with TACT EQ (below).

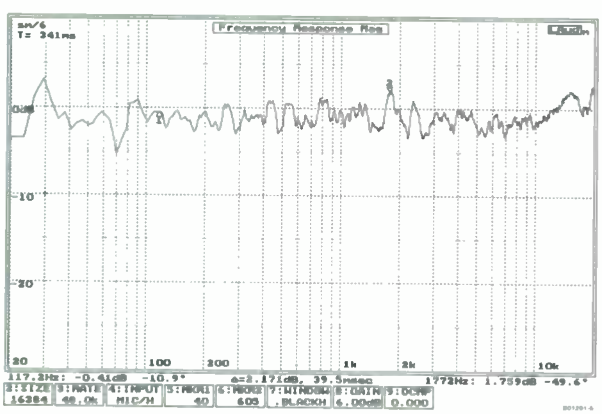


FIGURE 5: Sixth-octave smoothed response after EQ.

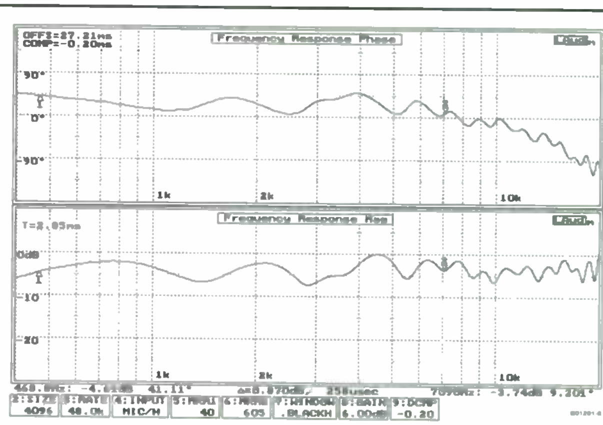


FIGURE 8: Quasi-anechoic response and phase, before EQ.

tra reproduced by a speaker).

I made the measurement from which these plots were derived by tying my Girardin measurement mike side-by-side on the same stand with the equalization mike supplied with the RCS 2.2 (a compact remote-powered electret unit made by LinearX). I positioned the mikes at ear height at my listening couch and then measured the room and computed the correction with the TACT unit and a portable computer (this same portable housed my Liberty Audiosuite measurement system, operating concurrently in Windows 95). The target curve for the TACT equalizer in this case was for a "flat" response curve (with a low-end limit of 30Hz).

TACT recommends setting the target curve for a response which is boosted in the

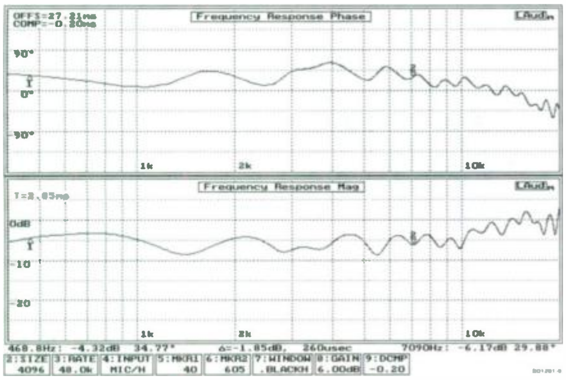


FIGURE 9: Quasi-anechoic response and phase, after EQ.

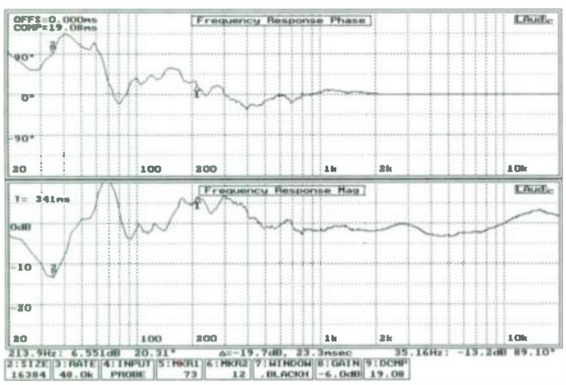


FIGURE 10: Electrical response of TACT EQ (as configured for these tests).

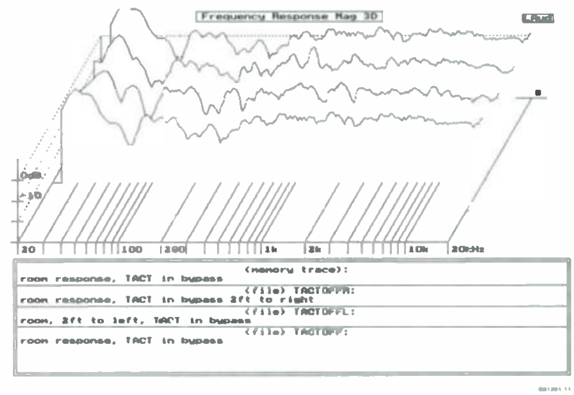


FIGURE 11: Smoothed response at three positions across 4′ lateral span, EQ off.

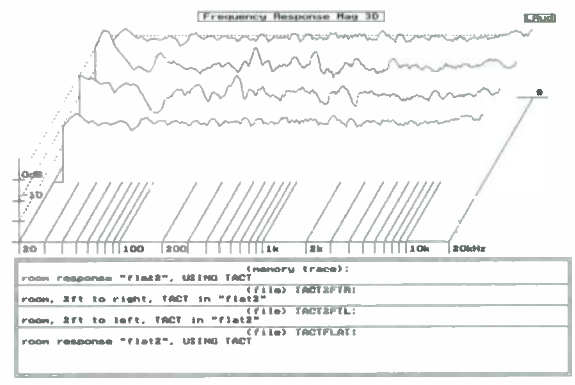


FIGURE 12: Smoothed response at three positions across 4′ lateral span, EQ on.

PHOTO 3: TACT RCS 2.2 accessories. Top row: power cable, RS-232 (9-pin) data cable, mike cable, remote control; bottom row: software CD, calibrated microphone with calibration file disk.

low frequencies and sloped gradually downward at the higher frequencies, because the literature states that this is how an anechoically flat speaker will behave in a typical room (and, presumably, most recordings are engineered for this flat speaker when used in a typical room). But for these demonstration plots, I used the "flat" setting for easier interpretation of the result. In listening, I also usually preferred the "flat" result with most recordings, with a few exceptions. That may be indicative of my listening tastes and choices in music.

Figures 4, 5, and *6* show the

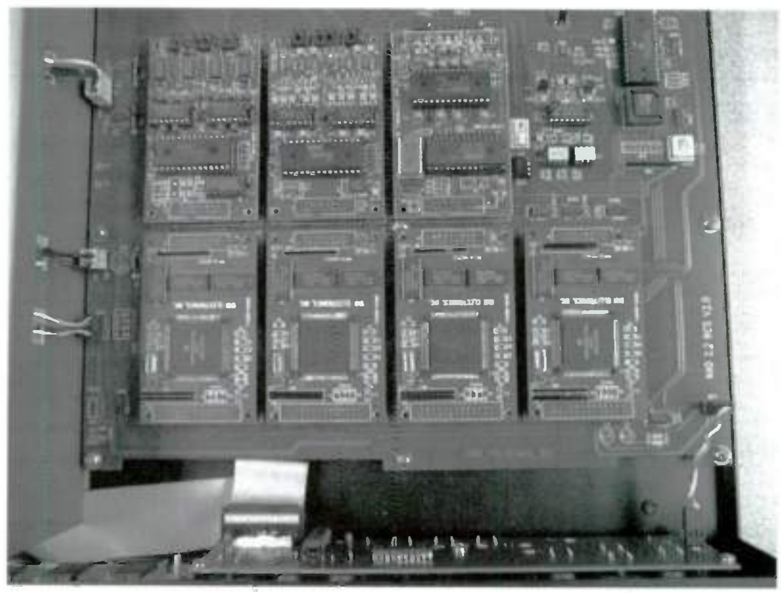


PHOTO 4: Main board of TACT RCS 2.2.

PHOTO 5: Front panel components board, rear view.

raw response, the sixth octave smoothed response, and the central spectrum curve for this speaker after equalization to a flat response curve. As you can see in *Fig. 4*, the TACT unit does not remove fine-grain echo effects from the frequency response itself, but apparently works to make the response flat in a sixth- or third-octave sense (*Fig. 5*). The central spectrum curve also predicts a very neutral perceived sound (*Fig. 6*).

The equalization result, performed simply using the "Quick Guide" in the TACT manual, clearly shows the desired flattening of the frequency-domain characteristics of the system. Yet to be seen are the time-domain effects of the equalization process.

Figure 7 shows the energy-time curves and reverberation for the system without (top) and with (bottom) TACT equalization. The plots do not reveal any major differences in the fullband energy-time characteristics due to application of EQ. The RCS 2.2 obviously does not attempt to change or correct room reverberation, only to adjust the response trends of the system.

Figures 8 and *9* show the "quasi-anechoic" complex frequency response for this relatively linear-phase speaker system before and after EQ. This is the response with the echoes removed, showing only the characteristics of the direct sound in the 2.85ms before reflections in the room arrive at the microphone. Notice how TACT, which considers the full-room response in its correction calculations, actually lifts the anechoic response from flat above 10kHz. This is because, as mentioned in the manual, a full-room loudspeaker response drops at high frequencies, and the TACT unit has been instructed in this case to "make it flat."

But this flattening, viewed "full room," brightens the response anechoically. More significant is what has happened to the phase response (top curves in *Figs. 8* and *9*), or rather, what has not happened: The phase curve has changed very little, perhaps even leveling a bit. Digital equalizers are capable of doing their equalization with completely linear phase, as is evident in the measurement of the electronic response of the TACT EQ (*Fig. 10*).

In this figure, about 19ms of fixed delay has been mathematically removed to illustrate the linear phase of the equalization being applied above 1kHz in the TACT system (upper curve). The magnitude response (lower curve) is quite similar to the inverse of the speaker's in-room measured response curve, as was shown in *Fig. 2*.

Next, I investigated the robustness of the TACT equalization algorithm as the listener changes position. TACT has provisions in software for making a spatially averaged measurement, rather than a single-point measurement, when performing the room correction. In this process, multiple measurements are made at different locations, which can then be weighted in a well-documented process to calculate an overall best correction curve. This presumably aims at striking the best compromise for a number of seating positions or over a defined area. For my test, however, I used only the single-point technique, assuming this would be a worst-case scenario, and would best reveal location sensitivity in the equalization process.

After equalizing the system to flat at the initial position, I moved the mike 24″ to the left, and then 24″ to the right, with measurements made with "EQ in" and "EQ out" at both positions. The results are given in *Figs. 11* and *12*. In each of these plots, the curve for the center (reference) mike position is repeated at the front and back of the plot, with the curves for the offset mike positions placed between them to best highlight differences. For the unequalized case, *Fig. 11*, the general slightly lumpy nature of the response curves is similar for all three positions.

In *Fig. 12* (equalization applied), you can see that the response for the center reference position for which the correction was made is much flatter than for the offset positions. But significantly, even in these offset seating positions the EQ'd curve is flatter and smoother than for the original unequalized case. This result was also borne out in listening: there was no increased sense of a supercritical sweet spot when the TACT equalizer was used. Audible improvement to the tonal character of the speakers was evident across this seating range, and in the rest of the room the sound was judged to be at least as good as without the equalization. Application of correction appears to be wisely (one might even say "tactfully") applied.

LISTENING TEST

The most easily identified audible change with the RCS 2.2 equalization applied was better fullness in the bass and the midbass. Acoustic guitars seemed to have more "body." Vocal sounds lost the "telephone" effect that is common when the lowest frequencies are not reproduced well—this effect was noticeable mostly on male vocals, but also with female vocals. This is likely due to the TACT unit cleaning up a deficiency of the response in the 100–400Hz range, which is deficient in many audiophile setups because of floor or wall bounce effects. I expect that the same improvement would be found in a large number of systems if TACT type equalization were applied. By switching from EQ-in to EQ-out, I became more aware of how this midbass deficiency can cause an artificial character to the sound—crisp sounding, but not quite human. Orchestral

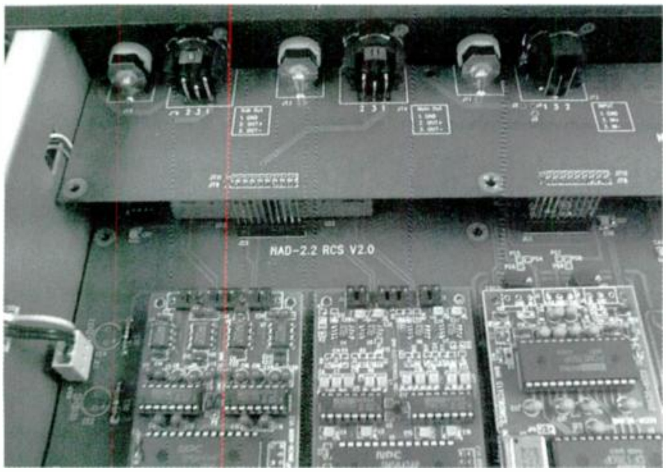


PHOTO 6: Rear panel circuit card, upper level.

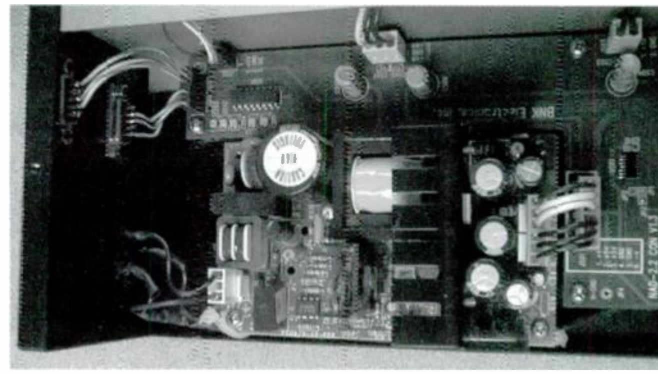


PHOTO 7: TACT RCS 2.2 main power supply.

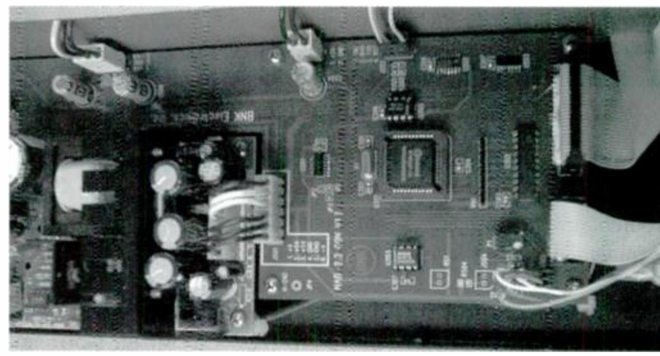


PHOTO 8: Secondary power supply section.

music seemed more balanced and less speaker-like.

Previously, when I played recordings in which footsteps are heard (or foot-tapping on a wooden stage), my system had a kind of "thump" overemphasis, perhaps impressive for home-theater effects but not very accurate sounding. This gave the effect with some recordings of having been recorded in a large barn! TACT 2.2 completely eliminated this character from the sound, removing the thump but still leaving the bass, and without the sense of something missing which had occurred with some previous attempts I had made to remove room peaks using analog parametric notch filters.

I had expected to hear a difference in the perceived image depth with the TACT system, but can't say I noticed much of this. This speaker system was already quite good in that regard, so this may not have been the best test for that. I was less distracted by the sound of the speakers when the TACT equalization was applied. Ensembles sounded more like ensembles rather than like two speakers playing ensemble music, but the difference wasn't so much in terms of image or depth but in "completeness" of the sound.

All in all, I very much liked the TACT RCS 2.2 equalizer system. Though I wish it had the option of doing a flat and anechoic single-point correction (only for my intellectual curiosity), the application of the TACT in my system made a definite improvement to the sound with no negative side effects. I am definitely going to miss having this (rather expensive and beyond my budget) device in my system and will regret returning it.

Trade Secrets

THE STORY BEHIND FERROFLUID

By Mike Klasco and Steve Tatarunis

In this series, Mike Klasco takes a close look at speaker components, providing an insider's perspective on how they work and how they can work better. Steve Tatarunis of Ferrofluidics Corporation contributes to this installment, which discusses what ferrofluids are; why they are used in woofers, midranges, and tweeters; and new developments in this technology.

Undoubtedly, you have seen the terms "ferrofluid-cooled," "liquid-cooled," or "liquid magnetic suspension" in speaker ads and read the cryptic descriptions of the benefits of ferrofluids in speaker manufacturers' literature. But after 25 years of use in the audio industry, with over 500 million speakers treated, it's time for a comprehensive look at the implications of magnetic fluids on audio quality in high-definition loudspeakers.

BACKGROUND

Ferrofluids are composed of sub-micron-sized particles of magnetite (Fe_3O_4) suspended in a liquid carrier, first produced in research for NASA. Ferrofluidics Corporation, founded in 1968, was licensed by NASA to research the technical and market development of magnetic-fluid technology. Magnetic-fluid applications include high-performance bearings and seals, such as in computer hard disk drives, optical scanners, and rotating semiconductor process equipment. High-performance robotic systems use ferrofluids in stepper motors to improve settling time. More recently, the latest generation of DVD players and high-speed CD-ROM and DVD-ROM drives use ferrofluids to damp the laser servo actuators for more stable tracking and focus, and to simplify manufacturing processes.

Speaker manufacturers use ferrofluid for its thermal conductivity, which reduces voice-coil burn-out, and its magnetostatic force (a uniform radial centering force) within the magnetic gap, which suppresses voice-coil rocking. Ferrofluids, being magnetically responsive, are attracted to the voice-coil gap's flux field, and the fluid acts like a spring when the coil moves off center in the gap, acting as a restoring force to maintain concentricity, thereby preventing rubbing and buzzing. This is only partially due to the levitation effect of ferrofluid's magnetostatic force.

Another factor in reducing voice-coil scrapes against the speaker's top plate is the lower voice-coil operating temperature, which limits voice-coil expansion and reduces gap clearance. Even the simple lubrication effect of the fluid reduces the abrasiveness of coil/top plate collisions. Ferrofluid also is a deterrent to dirt or particles entering the gap and acts as an inhibitor of corrosion in the coil and gap. Speakers used for life-safety and voice-warning applications almost always use ferrofluids in order to pass stringent UL qualification tests.

Many speaker manufacturers have had positive results with ferrofluids in tweeters and midranges, but required a more appropriate ferrofluid for woofers—one with low-viscosity and high-saturation magnetization (magnetization strength) in order for the fluid to stay in the gap even at large excursions without requiring extensive changes to existing woofer designs. About seven years ago, woofer-grade ferrofluids were commercialized. The first speaker companies to use ferrofluids in subwoofers were pro-sound manufacturers, followed by autosound manufacturers. Quite a few high-end speaker companies first began evaluation of ferrofluid woofers about five years

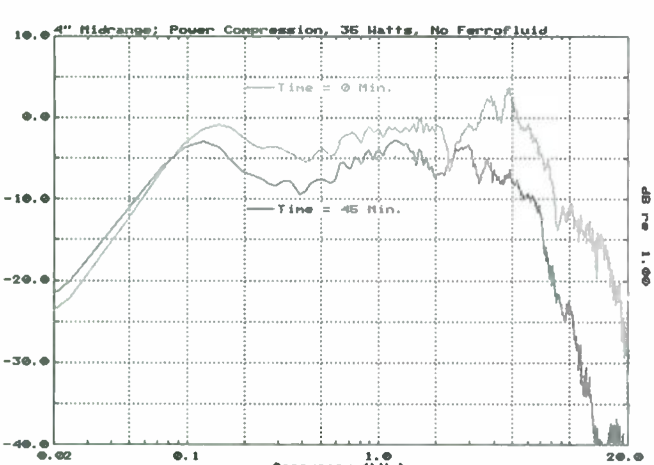


FIGURE 1: Power compression without ferrofluid, 35W.

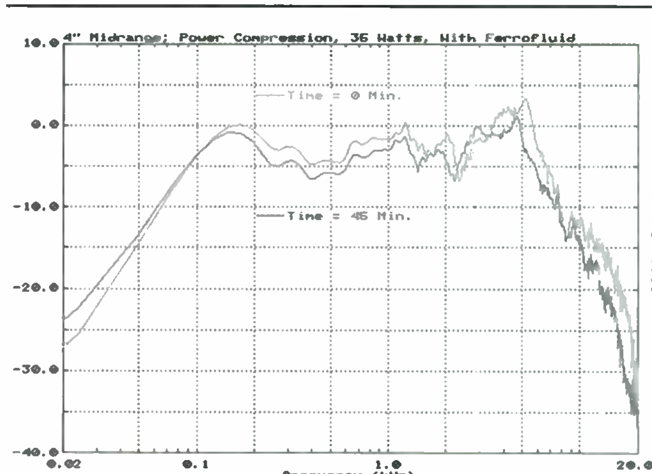


FIGURE 2: Power compression with ferrofluid, 35W.

ago, and today there are many commercial examples available.

WOOFER SOUND QUALITY

Ferrofluids usually improve transient-response settling time (the ability of the speaker to stop when the signal stops). The damping is effective since it is applied directly in the motor, rather than on the cone or suspension, as with "after the fact" damping treatments. If a woofer has a top-end response peak, ferrofluids tend to bring this under control with fewer side effects than a passive crossover-network solution. Ferrofluids reduce certain mechanical noises that speakers make in and around the voice coil when the cone moves.

Perhaps the biggest difference is not what you hear, but what you don't hear. Speakers change their sound quality as the voice coil heats up—normally while playing music for an extended period of time at realistic (or beyond realistic) sound levels. Since ferrofluids inhibit much of the power-compression effects that would otherwise result, the sound characteristic is more stable over time with ferrofluids.

HEAT BUILDUP AND POWER COMPRESSION

The heatsinks in power amplifiers are a familiar sight, since most amplifiers are less than 75% efficient, with the wasted 25+% energy resulting in heat. An amplifier that consumes 100W will output about 75W of audio power and 25W of heat. But when the 75W of audio signal is connected to a speaker, almost all of the power is dissipated as heat within the driver, with only a tiny amount of signal actually converted to acoustical output. Only the most sensitive speakers achieve efficiencies of 5%, with most well-damped dome tweeters and medium-density paper-cone or poly-cone woofers performing closer to 1–2%.

Let's say a speaker is connected to an amplifier and tested at a 1W level for impedance and frequency response. Next the power is cranked up. After the CD plays for an hour, the speaker is retested and the impedance found to be significantly higher. For those into T/S parameters, the rise in voice-coil temperature results in an upward shift of the DC resistance of the speaker's voice coil—thereby shifting the Q_{ES}. This results in a loss of both the low- and top-end response of the driver. The crossover network turnover points will also drift significantly due to the shift in the speaker's impedance.

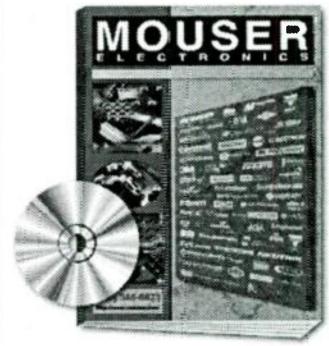
The two response plots (*Figs. 1* and *2*) show a 4″ midrange speaker driven at 35W. The first graph is the speaker without ferrofluid, one plot taken immediately, the second after 45 minutes. There is a loss of 3–8dB throughout most of the response range, with a strong loss of output at the top-end. The second graph is the 4″ midrange treated with ferrofluid, under the identical conditions. Note that the loss of output is only 1–3dB, with only a slight shift in top-end response. The ferrofluid has helped to preserve the linearity of the speaker's response, eliminating some of the aural cues that are a common byproduct of power compression.

SPECTRAL CONTAMINATION

Aside from frequency response and other basic measurements typically used to judge the quality of a speaker's reproduction, complex multitone techniques have also been devised. Using spectral-contamination testing, the "self-noise" of the speaker can be revealed. Spectral contamination is a measure of intermodulation distortion, but IM typically consists of only two test tones, while music consists of many more. All speakers

ABOUT THE AUTHORS

Mike Klasco heads Menlo Scientific, Ltd., which serves as a consultant to a number of loudspeaker firms. Steve Tatarunis is the Audio Products Manager at Ferrofluidics Corporation.



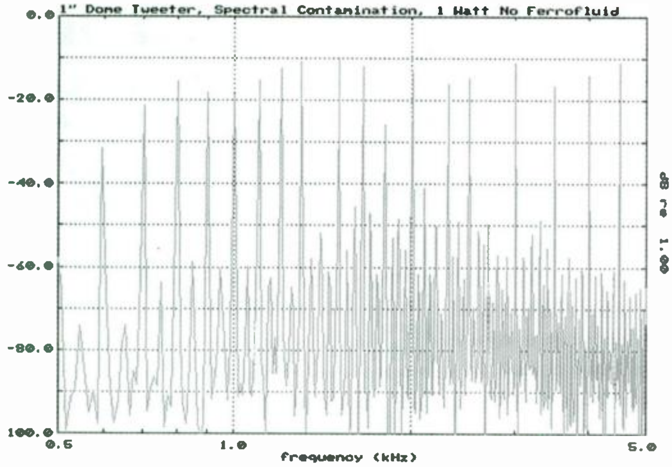


FIGURE 3: Spectral contamination without ferrofluid (1″ dome tweeter at 1W).

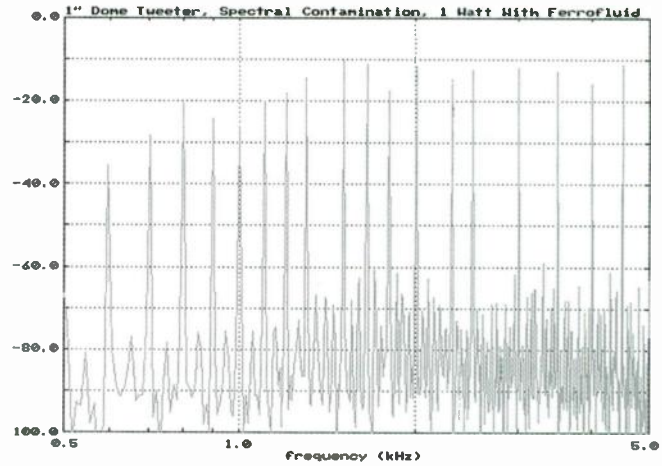


FIGURE 4: Spectral contamination with ferrofluid (1″ dome tweeter at 1W).

have subtle (or not so subtle) buzzes, rattles, noise modulations, and other anomalies.

Spectral contamination uses multitest tones (as many as 50 or more) that are generated simultaneously and sent to the speaker. Each tone is about 1Hz wide and spaced some number of hertz apart. This test signal approximates the complexity of music.

Deane Jensen devised one variation of this technique and wrote an AES paper (preprint #2725; 85th Convention, Los Angeles, 1988) describing the procedure. The Bell Labs SYSid acoustic analysis test system offers the spectral contamination test, while the Audio Precision One uses a similar procedure called FASTEST. Interestingly, Audio Precision refers to FASTEST as a QC procedure for automatic testing of buzz and rubs of speakers coming off a production line. If high enough dynamic range is available (which is achieved with the SYSid using a technique known as synchronous averaging), then different speaker designs that are operating correctly can be evaluated for their relative freedom from spurious resonances.

In this test, the intrinsic self-noise of the speaker tends to fill the space between the tones, albeit quite a bit down from the tones. For example, a shallow and relatively undamped metal-dome tweeter will usually be noisy. When the tweeter is excited by the multitone test signal, all the resonances are excited and contribute to the output. Perhaps the "acoustical dirt" will be down only 30 or 35dB on a fairly large midrange metal-dome tweeter. A very high-quality treated fabric soft-dome tweeter may contribute negligible noise and mostly pass just the discrete test tones, with the self-noise of the speaker being down 40dB to 50dB or more.

Why is this important, and what does this have to do with ferrofluids? Actually, spectral contamination may just be the most important single test of a speaker's quality. If a speaker has low spectral contamination, then ambience and the inner voices of the music will not be masked by the speaker's muck. A poor speaker will fill in the space between the tones with junk.

RESONANCES

While ferrofluids cannot help damp cone-diaphragm breakup directly, the voice-coil bobbin's torsional resonances are dramatically reduced (*Figs. 3* and *4*). In fact, ferrofluids are primarily used in seals and precision spindle motors for computer disk drivers to damp these torsional resonances.

A bobbin, the former that the speaker's voice-coil wire is wound on, is a critical element in sound reproduction, since the vibration from the voice coil must travel through the bobbin to reach the cone or dome diaphragm. Any resonances within the bobbin will contaminate the sound quality before it reaches the diaphragm, and therefore this spurious energy will be radiated into the room. With ferrofluid in the gap, the bobbin is damped and its "noisiness" is attenuated.

Are these resonances a real problem, and do ferrofluids really help? One way to evaluate this is to build up a speaker without the cone, with only the voice coil, spider, and dust cap. The self-noise of the bobbin will be clearly audible without ferrofluid, while the fluid-damped version will be significantly quieter. Pioneer measured similar phenomena almost 20 years ago when they first began using ferrofluids in dome tweeters.

In a dome tweeter, the diaphragm is very tightly coupled to the bobbin, and the bobbin's distortion (and the distortion-reducing effect of ferrofluid) is most dramatic. But in the case of speakers with large cone areas, what occurs in the bobbin has less of a direct effect on the flapping of the cone far away, and performance will depend more on the speaker engineer choosing a quality cone rather than ferrofluid damping.

VENTING TECHNIQUES

Still another aspect of ferrofluids in woofers is proper cavity venting. The techniques used to prevent ferrofluid splashing due to pressure buildup behind the voice coil (from air trapped within the magnetic structure) also dramatically reduce the speaker's modulation noise. Reducing the modulation noise of the speaker is critical to maintaining clarity and definition. *Figure 5* shows various woofer-venting schemes.

Behind the speaker dust cap is the pole piece. During the backward stroke, the air trapped behind the dust cap will cause it to puff outward, and on the outward stroke, the dust cap will buckle inward, in both cases making unpleasant noises. The dust cap may even eventually be blown off (very impressive on the *1812 Overture*!).

The pole pieces on many woofers are vented both to relieve the under-the-dust-cap cavity pressure and to aid in cooling the voice coil. It is not always possible to vent the pole piece, either due to cost considerations or when venting would reduce the magnetic return path efficiency (such as with a small-diameter pole piece). Prevention of pressure build-up can then be accomplished by a breathing dust cap or a vented voice-coil bobbin, or venting the cone body underneath the dust cap.

A chamber is created by the internal diameter of the magnet, outer diameter of the pole, and the space between the top and bottom plates. On the downward stroke the voice coil displaces this volume, increasing the pressure in this cavity. Without venting, the trapped air is forced through the voice-coil gap at high velocity, and the ferrofluid may splash if the excursions of the coil are large. This pressure can be vented by holes in the back plate, or, if the pole piece is vented, a crosswise vent can link the back-plate cavity to the vented pole piece. Vented top plates have also been successfully used to relieve this pressure.

The chamber between the spider and the basket should also be vented to relieve the air pressure in this cavity. The spider is the tan-colored woven fabric that you can see through the speaker-basket windows (when looking at the rear of the woofer). Although the spider appears to be an open fabric, the air resistance of the treated fabric is very high.

With proper cavity venting techniques, the air velocity and turbulence noise within the voice-coil gap is greatly reduced, but, without using ferrofluid, there will be an increasing chance of voice-coil rubs and buzzes. Without venting, the high-velocity air streaming through the gap (from the unvented air cavities) creates an air-bearing effect in the gap, accompanied by much whistling noise. But when the cavities are properly vented and ferrofluid applied, gap modulation noise stops completely, both because of the sealed gap and the fluid damping of bobbin torsional resonances.

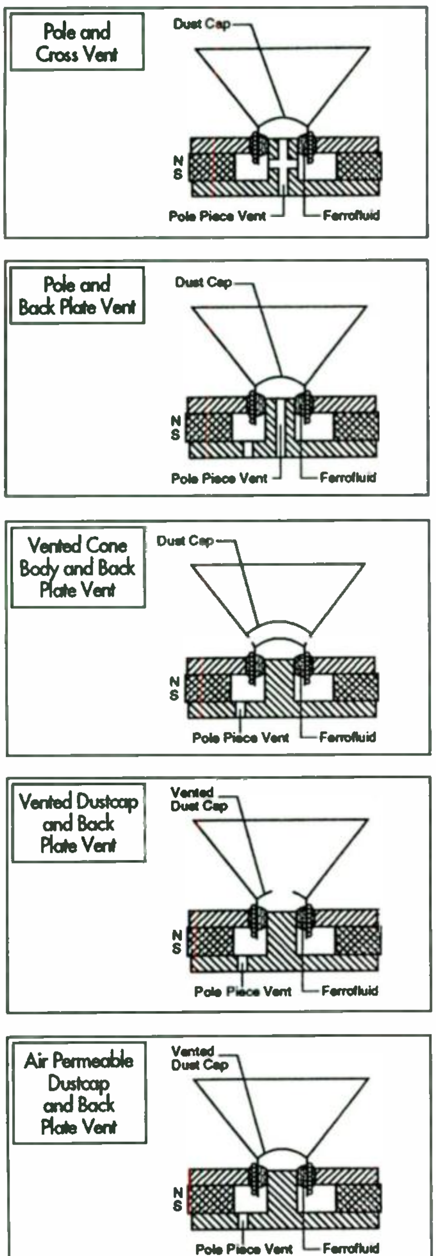


FIGURE 5: Venting schemes for woofers.

AIR MODULATION NOISE

A cone speaker has a number of internal chambers: behind the spider, behind the dust cap, and behind the voice coil (the space created inside the magnet ring and the back plate). As the driver moves back and forth, the cone/spider/dust cap is either compressing or creating a vacuum in these chambers. This can cause air-sucking noise ("air modulation noise"), or even worse, the buckling of the driver diaphragm elements or spurious noise.

It is important that the optimum viscosity of ferrofluid is selected with attention to the viscosity-versus-temperature curve and the effect on the efficiency and operating bandwidth of the driver. When normal operating temperatures are typical, then the damping effects of high-viscosity ferrofluid can be used as an integral design factor. When high-temperature operation is common, you should select the viscosity so that the passband response of the driver is not affected by ferrofluid at normal operating temperature. The decreasing viscosity of ferrofluid counteracts the effect of power compression during high-temperature operation so that the frequency response will not be appreciably altered.

Early design efforts to use ferrofluids occasionally went overboard in the use of excessively high damping characteristics. Ferrofluids were used not only to control top-end resonance problems, but also to limit low-frequency excursion of tweeters. In the 1970s, one Japanese studio monitor (which also made it into the US and was sold in the audiophile market) eliminated the crossover network completely and used high-viscosity ferrofluid of a few thousand centipoise. Today, most APG (Audio Product Grade) ferrofluids are only a few hundred centipoise, or less.

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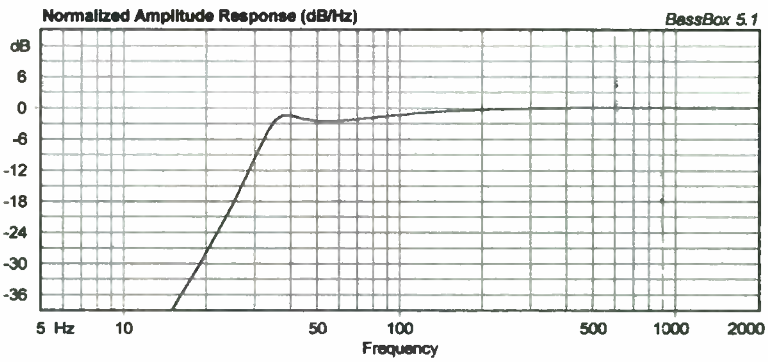
BOX DIMENSIONS

As I read "Test Drive: Dual 5″ Vifa MTM," by Dick Carlson in *SB* 1/98, I realized what I believe to be Mr. Carlson's error in building this loudspeaker.

In column 2 of Fig. 1, the number of drivers listed is "1." Therefore, all BassBox 5.1 calculations were based on a system using one driver, while the Vifa kit design utilized two. This explains why Mr. Carlson's design model specified a 10″ port tube, while Parts Express called for a 3″ tube.

Using the same T/S parameters that were shown in the article, I ran my own models on BassBox 5.1. When I specified a single driver and a 10″ long vent, I came up with the same results that are shown in the article's Fig. 1, including a group delay of 26ms (!) at F_3. However, when I changed the number of drivers to "2," low-frequency rolloff changed to 62Hz. This was approximately where Mr. Carlson heard his system rolling off during his overall evaluation. When I changed the vent length to the prescribed 3″, the box tuning rose from 40.6Hz to 60.4Hz, and F_3 changed to a very respectable 53Hz with a group delay of 12ms.

Ken Ketler
Shrewsbury, MA



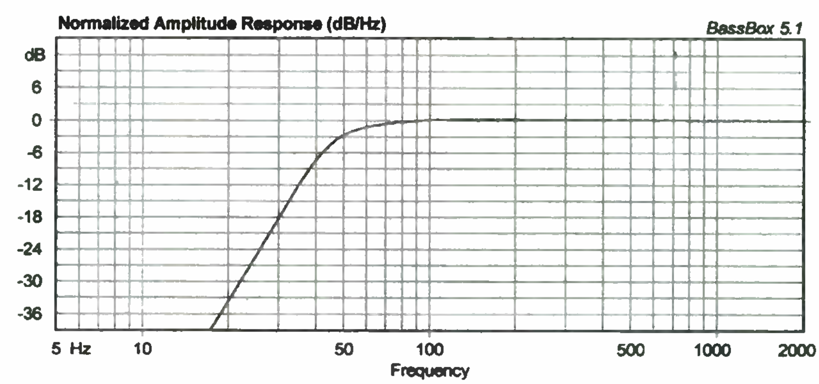
Loudspeaker Parameters
--General Information--
Company: Vifa
Model: P13WG-00-08
Note/SN: 5-inch woofer
--Mechanical Parameters--
Fs = 60.0 hertz
Qms = 1.700
Vas = 0.300 cu.ft
Cms = 0.167 in/lb
Mms = 0.265 oz
Rms = 4.452 lbs/sec
Xmax = 0.130 in
Sd = 13.3 sq.in
Dia = 4.1 in
--Electrical Parameters--
Qes = 0.680
Re = 5.5 ohms
Le = 0.7 mH
Z = 8.0 ohms
BL = 4.8 N/A
Pe = 40.0 watts

--Combination Parameters--
Qts = 0.480
ηo = 0.380%
Sens = 88.00 dB (1 watt)
--Multiple Drivers--
Number = 2
Standard
Wiring: parallel
NetZ = 4.0 ohms
NetRe = 2.8 ohms
NetSens = 94.0 dB

Box Parameters
Custom Vented Box
Vb = 1.800 cu.ft
Fb = 37.8 hertz
F3 = 34.7 hertz
QL = 7.0
Fill = heavy
Ports = 1 (round)
Dv = 2.75 in
Lv = 2.97 in

SL-carls-1

FIGURE 1: Dick Carlson's revised schematic reflecting box size in relation to manufacturer's vent calculations.



Loudspeaker Parameters
--General Information--
Company: Vifa
Model: P13WG-00-08
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--Mechanical Parameters--
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--Electrical Parameters--
Qes = 0.680
Re = 5.5 ohms
Le = 0.7 mH
Z = 8.0 ohms
BL = 4.8 N/A
Pe = 40.0 watts
--Combination Parameters--
Qts = 0.480
ηo = 0.380%
Sens = 88.00 dB (1 watt)

--Multiple Drivers--
Number = 2
Standard
Wiring: parallel
NetZ = 4.0 ohms
NetRe = 2.8 ohms
NetSens = 94.0 dB

Internal Box Dimensions
--box--
Vol = 0.700 cu.ft
Shape: Square Prism(s)
h = 21.50 in
w = 7.50 in
d = 7.50 in
no = 1

Box Parameters
Custom Vented Box
Vb = 0.700 cu.ft
Fb = 49.5 hertz
F3 = 49.5 hertz
QL = 7.0
Fill = heavy
Ports = 1 (round)
Dv = 2.75 in
Lv = 5.82 in

SL-carls-2

FIGURE 2: Revised schematic with correct number of drivers and vent calculations.

Dick Carlson responds:

Mr. Ketler was very observant, and his comments are appreciated. He is correct in assuming that if I had used a 10″-long vent during construction, the box may have sounded better, especially in the 40–60Hz regions. But, I must confess that is not what I did at all. Had the box been built to support a vent length of 3″, the result would have been a box made to an internal size of 1.8ft³ (Fig. 1).

However, since I did not implement any change to the box vent and used all the materials provided by the manufacturer, the computer prediction, including the actual performance results, for this loudspeaker resulted in a technical mismatch. That is, if I were to replace the 3″-long vents with 5.82″ vents, the low-frequency performance should improve significantly. Figure 2 *shows that using the 0.7ft³ box with a vent length of 5.82″ would result in an improved response. A comparison of both predictions shows that the response of* Fig. 2 *will occur with optimum results.*

Furthermore, in Mr. D'Appolito's test report, he mentioned that most of the acoustic output should have come from the port, which, in the case of this system, it did not. Rather, the system behaved more like a closed-box configuration. In addition to the incorrect vent length, this occurred mostly due to the overwhelming amount of damping material used in each system (roughly 70% of the total volume). In his words, "This much damping kills the box Q."

The speakers are currently set up in my living room and are connected to a (vintage?) model 9090DB Sansui receiver. They are augmented by a small subwoofer I built using a 10″, dual voice coil low-frequency driver from Madisound. They complement each other quite well. I plan to replace the vents in the loudspeakers with 5.82″-long vents, then conduct another series of low-frequency tests to confirm the aforementioned predictions.

I am also very pleased that Parts Express will be making the necessary changes to this potentially outstanding loudspeaker system, which will improve cabinet strength, aesthetics, and grille-frame construction and mounting.

DIFFRACTION DISSATISFACTION

I confess to a certain amount of irritability motivating this letter. The term "low diffraction" has passed before my tired eyes more times now than I can bear without speaking up.

Before I skewer myself on the spit of current audio-design thought, allow me to say that I have been involved with loudspeaker design for 30 years and have a degree in laser/electro-optics. I am currently employed in electronic design work involving ultrasonic sound generation and detection. Because of this background, and particularly because much of my loudspeaker work is with horns, I am painfully aware of diffraction-related effects.

The current obsession with tiny frontal area loudspeakers as a means to divorce yourself from the effects of diffraction is a mistake and a conceptual design error. Diffraction *is*, and all you can do about it is either find a means to exploit it or curse it. For example, we exploit it by using a small-diameter dome tweeter to provide a broad dispersion pattern. The next consequence of this decision is discovering that the broad dispersion pattern has delivered a lot of acoustical energy to the edges of your speaker front baffle. When it again diffracts you get a "ring of fire" of secondary wavelets different in time (at the observer's vantage point) and displaced spatially as well.

Now, you get to work minimizing this negative effect of diffraction. If you reduce the baffle diameter of the tweeter to such a size that the inevitable second diffraction event occurs very close to the first, the time and physical displacements will be so small that the negative effects will be reduced to insignificance. This has worked so well that the technique is extended to the speaker which carries the wavelengths longer than the tweeter. The test setup reports that by judicious sizing and cunning physical arrangement the frequency-response altering interaction between the primary sound source and the secondary edge wavelets is reduced to a piddling minimum. This constitutes success, yes? Don't be so sure.

1. The term "minimum diffraction" is either ill-defined or misapplied. A speaker's diffraction is the same whether it is mounted on a large or small baffle. We are really discussing the interaction of the secondary wavelets on the amplitude response or the directivity-changing effect of the speaker baffle. Superficially, "minimum diffraction" implies that a small mounting baffle somehow reduces diffraction. This is simply not true.

The reduced baffle size only pushes upward in frequency the point where the secondary wavelets begin to cause changes in the response curve. If this point lies in the cutoff region of the loudspeaker (or above the range of human hearing with tweeters), you win what appears to be an important technical battle. The point here is that you did nothing to reduce diffraction, so the idea that diffraction has been minimized is misleading and thus ill-defined.

Conversely, the small baffle results in little increase in the natural directivity of the loudspeaker. The result of this is that diffraction is allowed to broaden the distribution of the acoustic radiation to the maximum possible extent. The small-baffle loudspeaker would be more correctly labeled "maximum diffraction" type if it were the sound dispersion to which we refer. This is at best a misapplication of the term.

2. The second point has to do with technology and the typical application of speakers in real listening environments.

It has occurred to me that the preoccupation with the so-called "minimum diffraction" speaker system coincided with the widespread use of pulse analysis by FFT. Please don't get me wrong, I think FFTs are the cat's meow; it's just that I see a way it can lead to investigating minutia while ignoring the enormous. The way in which the technique provides the very useful anechoic data precludes us from viewing the true nature of the reproduction system, as it will be used.

For clarity's sake, this measurement technique involves a very brief staccato impulse in which the whole of the audio range may be contained. This is fed to the loudspeaker, and a microphone receiver is engaged immediately before the pulse arrives at the microphone and is turned off immediately thereafter, so that reflections from nearby walls or other objects are excluded from the analysis. This is a very useful technique and provides a means of comparing different loudspeakers or tuning alignments without external influences.

The problem I see is that those pesky baffle edge signals do lie inside the open microphone interval and make a mess of what might otherwise be a very pretty picture. Reacting as normal humans do, we find solutions to get the pretty picture back and ignore the fact that the reflections from the typical listening room's jumble of stuff is going to make an even bigger mess. This mess will happen at a lower frequency range and at longer time displacements than the stuff on the mounting plane of the speaker, but the effect is there.

At this point I should distinguish between the consonant slurring effect of front baffle dimensions and the second effect of confusing whatever time and spatial information was supposed to be on your favorite recording. The two effects have the same root but differ in that close object diffraction/reflection events produce secondary audio sources of a time delay which lies inside the note/sound coherence length, to borrow a concept from laser optics. The two signals interfere coherently, and rapid variations in amplitude and phase result.

This slurs consonants and makes percussion instruments such as cymbals hiss in an odd and unpleasant way. This is the primary effect designers are trying to avoid with the small baffle sizes. The region most likely affected is that covered by midranges and tweeters.

Objects near the speaker enclosures also produce coherent interactions, but at lower frequencies because of the increased distances involved. But these nearby objects also reflect audio images which may/will come from all sorts of distances, time delays, directions, and frequency content. However, please note that before this jumble of audio sources can happen, *the acoustic radiation must be transmitted to these remote objects by the loudspeaker.* This action probably does more to destroy the recording's spatial information in the midrange and tweeter bands than it does to produce coherent-type slurring effects.

If you build a "minimum diffraction" type loudspeaker, the dispersion is maximized and the ratio of direct to reflected sound is reduced. Thus the greater amount of sound you'll hear coming from all those unauthorized secondary sound sources. You have "fixed" one problem in exchange for another.

3. Observations, comparisons, and the root of all evil.

At the other end of the "does size matter" controversy are such technical approaches as dipole (or bipolar, if you prefer) baffles. These large flat alternatives to small boxes (or any box) are the very antitheses to the "minimum diffraction" enclosure. To summarize my experience with them, they can indeed have a sweetness and clarity to be desired. However, I cannot justify any baffle system that kills off fundamental tones and leaves the distortion components untouched. Worse yet, the low-frequency loss must be accommodated by increased woofer drive that produces even more distortion. Sigh, what might have been....

My dipole experience was not a loss because it provoked thought on what aspect of the design produces the good results. I've concluded that there are two features of the dipole loudspeaker that aid good sound reproduction. 1) The lack of radiation in the plane of the baffle (no sound to bounce off objects next to the speaker, and 2) low compression of the air, and hence no nonlinear effects to speak of.

I found it necessary to use semi-rigid fiberglass absorption materials (Armstrong #420 commercial ceiling panels with the cosmetic plastic surface treatment peeled off) on the panel face to reduce the edge effects in the midrange, because the path lengths for the front and rear of the mid-speaker were not uniform enough to ensure cancellation. Likewise the tweeter required this modification because it had no rearwave to cancel the edgewise radiation at all. It occurred to me that if this absorption scheme worked on a dipole that it must work on more ordinary enclosures also.

To try and capitalize on this new knowledge, I constructed an enclosure that featured a large (30″ × 39″) dipole-like face. I deviated from a dipole by enclosing the rear with a box of such size that pressure modulation was minimal. (The enclosure was in fact a tuned B2 alignment using a high Q_{ts} 8″ woofer with the tune frequency well below resonance.)

Once the acoustic treatment was in place on the front baffle, the same immediacy and clarity on the better dipoles was operating here also. I seemed to have found some of the dipole's charm without the low-frequency loss. Four symmetrically placed ports were located on the enclosure rear face to couple the output to the wall behind the speaker and to discriminate against the seemingly inevitable spike or two of higher frequency sound which makes its way out.

The bass output still sounded like a box, though resonant, and always colored in a signature way that I find less than perfect. The design still lacked a final touch, which turned out to be the use of a large cross-sectional area transmission line for rear wave

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processing. Using a 4:1 pipe cross-section to driver piston area design filled with ½ lb per ft^3 of spun polyester fill material produced the bass qualities I'd been looking for. This unusual combination of design elements (a large front baffle on a column-like transmission line enclosure) does for the direct to reflected sound ratio what a hood dipole does with a bass output, which is reinforced rather than diminished.

Many people have heard this device. To a man, the listeners concur that the speaker projects sound into the intended listening area with an authority unheard in small enclosures and that the transmission-line bass transparency was a match for the dipole. I'm only sorry I didn't get around to investigating the transmission-line enclosure sooner. Some people who build speakers for a living have a reaction to it that is very telling and brings up the next important point—money is the root of many evils.

4. Some contend that the "minimum diffraction" speakers produce an excellent stereo image when properly placed—that is, four feet or so out from the wall and with as much clearance laterally as you can muster. (I have taken this as a mumbling admission of the truth of which I speak). And they are right, "minimum diffraction" enclosures do produce some excellent results when given such leeway. Apparently the loss of efficiency inherent in a small front baffle is not given any value either. In fact, the subject of the smooth impulse test results keep coming up, reinforcing my idea that the probable motivations for small boxes are visual and commercial.

In what Paul Klipsch once referred to as "chargeable space," the "minimum diffraction" speaker is the largest you are likely to bring into your home. Chargeable space is the volume of space reserved for the use of an implement, regardless of its actual physical dimensions. If you are required to reserve a space of at least four feet or so around a speaker, then the chargeable space is larger than for any other type of home loudspeaker. The large baffle loudspeaker described earlier is rather insensitive to objects to the right or left and works well when moved back to the wall behind it. In this sense, then, it is actually smaller.

The following is my list of reasons why this large baffle concept is not/will not be found in commercial loudspeaker offerings (you can probably think up more): a) the sales of large dipole speakers is poor (wives and interior decorators are frequently causes), b) small speakers are cheaper to build, c) they are cheaper to warehouse, d) they are cheaper to ship, e) "minimum diffraction" response curves are pretty good advertising copy, f) when you manufacture loudspeakers for a living, innovation is always a means to increase sales, seldom a means of advancing an art form, and g) small speakers mean bigger profit margins.

Dan Gustafson
Grand Prairie, TX

MORE ON THE AR-3*a*

While I note that your editorial policy must be driven by the John Stuart Mill quote in your masthead, I wonder whether you apply it uniformly. I ask that as a new reader after seeing the way you published an apparently unedited reply from C. Victor Campos to the article by T.D. Yeago (*SB* 2/98, p. 54).

I must give Yeago credit for the grace with which he replied to the unabashed vengeance from his attacker. I was disappointed that a former manager with a reputable company thought he had to unleash so much venom on his prey. I question your editorial policy that didn't edit the caustic remarks in the first paragraph. Just because much of the national media is trying to "let it all hang out" in the name of freedom doesn't mean it is necessary or helpful to our society or your professional journal. My concerns are *not* contrary to Mill's statement in your masthead. I am opposed only to the unnecessarily caustic remarks that did nothing for a professional journal.

There are two ways that publishing the unedited reply from Campos could actually damage your journal. I have already spoken to the unnecessary venom from Campos some will find offensive. The other matter can limit the success of your journal. Some potential writers who might have something worthwhile to contribute may not write if they know they may be attacked so viciously. I wonder how much time Yeago spent having to reply to his attacker when he could have been doing something more creative to help us all.

Finally, writers will be discouraged from writing if you do not uniformly give them the opportunity to reply in the same issue in which their critics appear. I know. I stopped writing for a journal in another field because of these reasons. All this causes me to still lean toward canceling my subscription.

James W. Blevins
Salemburg, NC

I would like to thank Tom Yeago for an excellent series on rebuilding the AR-3*a*. Let me also thank Victor Campos for his very in-

TABLE 2

RELEVANT DATA AND PARAMETERS: WOOFER

	M_{ms}	C_{ms}	f_s	V_{as}	Bl	R_{dc}	Q_e
Stock	82g	1.3mm/N	15.5Hz	340 ltr	7.6T-m	2.5Ω	0.34
Modified	125g	1.3mm/N	12.5Hz	340 ltr	16.2T-m	4.5Ω	0.17

	R_{ms} (est.)	Q_m	Q_{ts}	Q_{cb}	f_{cb}
Stock	1.5	5.0	0.32	0.9	43Hz
Modified	3.0	3.3	0.16	0.46	35Hz

TABLE 3

RELEVANT DATA AND PARAMETERS: MIDRANGE DOMES

	M_{ms}	C_{ms}	f_s	V_{as}	Bl (est.)	R_{dc}	Q_e*
Stock	1.5g	0.4mm/N	205Hz	$155cm^3$	3.25T-m	2.7Ω	0.5
Modified	1.7g	0.4mm/N	190Hz	$155cm^3$	5.8T-m	4.0Ω	0.24

	R_{ms} (est.)	Q_m	Q_{ts}*	Q_{cb}*	f_{cb}
Stock	1	1.9	0.4	1.0	500Hz**
Modified	6	0.34	0.14	0.28	400Hz

TABLE 4

RELEVANT DATA AND PARAMETERS: TREBLE DOMES

	M_{ms}	C_{ms}	f_s	V_{as}	Bl (est.)	R_{dc}	Q_e*
Stock	0.24g	0.07mm/N	1.2kHz	$1cm^3$	1.4T-m	3.5Ω	3.1
Modified	0.28g	0.15mm/N	775Hz	$4.4cm^3$	2.8T-m	5.1Ω	0.88

	S_d	R_{ms} (est.)	Q_m	Q_{ts}*	Q_{cb}*	f_{cb}
Stock	$3.1cm^2$	0.5	3.6	1.6	2.25	1.7kHz
Modified	$4.5cm^2$	1.7	0.8	0.42	0.61	1.15kHz

*level-setting pots, resistors, and crossover components in series with the motor coil will affect this directly. These are free air figures up to the "cb" numbers.
**A guess here, because most of the enclosed air is coupled through the resistive restriction of the motor former/pole piece crevice. The rough impedance curve I ran on a stock midrange also indicated 500Hz.

formative letter about AR. I would concur with Mr. Yeago that he was not looking "with disdain" at the AR-3*a* but with reverence for a truly remarkable engineering accomplishment. But please, Mr. Campos, tell us more! I for one would like to know more about the early days at AR. I still regret selling my AR2axs.

Richard Jenkins
76346.1220@comupuserve.com

As a follow-up to my AR-3*a* series, in Part 4 (*SB* 1/98) on p. 26, the end of the paragraph at the top of the third column should read:

I'm guessing I have a Q_m in the 3 range, instead of the 6 or more that's typical. And what liquid-cooling I have is a huge improvement over a motor surrounded by air. The motor can stand a lot of heat anyway. It's a pretty big chunk of wire.

And readers might wonder why there's more to the "Data and Parameters" for the dome units (*Tables 3* and *4*) than for the woofer (*Table 2*). The answer is, a goof was made and the second line of the data was omitted. So here are all three tables. In the treble dome table (*Table 4*) a mistake was printed (Part 6, *SB* 3/98). C_{ms} for the modified unit should read 0.15mm/N as was discussed in the text.

Tom Yeago
Staunton, VA

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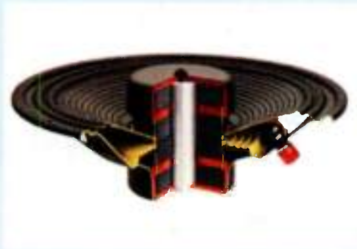
Through the course of the years Morel brought many unique and innovative products to the speaker industry. The introduction of the 3″ voice coil in a 5″ basket, using hexagonal shaped aluminum wire, utilizing a double magnet system and ducted design woofers and mid-basses are a few examples of the company's breakthroughs. Also introduced were the Integra concept (single motor system for both the tweeter and woofer) and the Push-Pull 8″ and 10″ subwoofers (dual motor system, dual voice coils with a single cone).

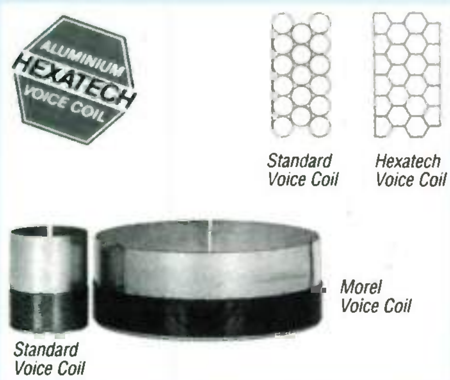
Integra



Push-Pull

Double Magnet

Morel Acoustics USA, Inc. has come a long way since 1978. Currently, the company has a diverse line of exciting products which includes over 40 models of tweeters, midranges, mid-basses, woofers and subwoofers. Being a leader in the field of speaker design, for our 20th year anniversary we are scheduled to launch several new products that are sure to attract attention.



Typical double magnet ducted woofer.

For further information please contact:

Double magnet tweeter

morel®

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