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Speaker Builder®

THE LOUDSPEAKER JOURNAL

UPGRADING THE WOOFER FOR A BIGGER LISTENING ROOM
Philip Abbate

TEST DRIVE
ARIA 5/RAVEN R-1
CRAIG STARK & JOE D'APPOLITO

BUILD A CURVED ARRAY WITH NINE ELECTROSTAT PANELS EACH
Bill Waslo

A Monster Dipole Takes Shape & Is Measured by Tim Sandrik

Anthrax Wrecks your Ears Faster Than Mahler

D'APPOLITO Does A Book On Loudspeaker Testing
G. L. Augspurger

Good News

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Reader Service #134

HOME-THEATER ADDITION

Niles' video-shielded HD-CTR[BX] free-standing center-channel speaker employs a 5¼″ mica-filled polypropylene woofer and ElastoDynamic™ tweeter. Housed in a ported, black-ash woodgrain finish enclosure, it features a high-frequency switch, dialogue intelligibility switch, gold-plated five-way binding posts, and a frequency response of 45Hz–21kHz, ±3dB. Recommended amplifier power is 10–15W, and sensitivity is 90dB for 2.83V of pink noise. Niles Audio Corp. Inc., 12331 130 St., Miami, FL 33186, (305) 238-4373, FAX (305) 238-0185, Website www.nilesaudio.com.

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A CENTER-CHANNEL FIRST

Westlake Audio has introduced the Lc255.1, its first center-channel speaker for use in either home-theater or high-quality stereo systems. Features include a three-way design that offers wide response, and you can position it either vertically or horizontally. It consists of ported dual 6″ woofers and a 5″ midrange, combined with a coaxially-mounted 1″ dome tweeter that contains a frequency response of 48Hz–18kHz ±3dB. The cabinet is available in a satin walnut hardwood veneer finish, and is magnetically shielded, which allows it to be placed on or near video monitors. Westlake Audio, 2696 Lavery Ct. Unit 18, Newbury Park, CA 91320-1591, (805) 499-3686, FAX (805) 498-2571, Website www.westlakeaudio.com

Reader Service #135

About This Issue

When you—or your spouse—remodel a room, you'll probably determine you must modify your sound system as well. **Philip Abbate** found himself in that situation and relates his experiences in downsizing his subwoofer setup (to a 12″ driver) without suffering any performance loss ("Afterthoughts on Aftershock," p. 10).

Since audio and video are familiar bedfellows, it's no wonder how greatly the video experience is enhanced by good audio. In "Audio Video Revisited" (p. 12), **Patrice Pelletier** constructs and tests a tweeter/woofer combination for use with his VCR.

Along with riding a motorcycle without a helmet and running with scissors, listening to your favorite tunes can also be hazardous to your health. And it doesn't necessarily need to be heavy metal music, either. **Jesse Knight** takes an interesting look at the factors that may damage hearing ("Dangerous Music," p. 16).

Getting the sound to the listener in a focused array setup requires creativity, patience, and also some carpentry skills. **Bill Waslo** shows you how to build the curved frame to support the electrostatic arrangement ("Focused Array Electrostatic, Part 2," p. 22).

The "Boys of the Dipole Design" are back. This time they disclose the measurements and test results for the enclosure, which takes advantage of the dipole features that maximize the best bass sound ("Designing the Dipole Monster," Part 2, p. 34).

Craig Stark (son of tape and computer guru of the same name) teams with **Joe D'Appolito** to assemble and test the Aria 5/Raven loudspeaker kit ("Test Drive," p. 44).

Also in this issue we're pleased to present **G.L. Augspurger**'s review of what promises to be the definitive work on testing loudspeakers by our own Joe D'Appolito ("Book Report," p. 54).

Speaker Builder®

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.
JOHN STUART MILL

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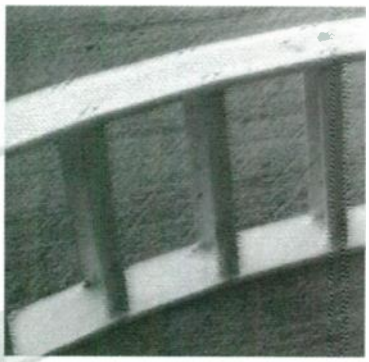
Contact Advertising Department, Speaker Builder, PO Box 876, Peterborough, NH 03458, 603-924-7292, FAX 603-924-9467, E-mail advertising@audioXpress.com.

Speaker Builder (US USSN 0199-7920) is published every six weeks (eight times a year), at $32 per year, $58 for two years; Canada add $8 per year; overseas rates $52 one year, $93 two years; by Audio Amateur Corporation, Edward T. Dell, Jr., President, at 305 Union Street, PO Box 876, Peterborough, NH 03458-0876. Periodicals postage paid at Peterborough, NH and an additional mailing office.

POSTMASTER:
Send address change to:
Speaker Builder, 305 Union St.
Peterborough, NH 03458-0876

# Speaker Builder
THE LOUDSPEAKER JOURNAL
VOLUME 19 NUMBER 6 OCTOBER 1998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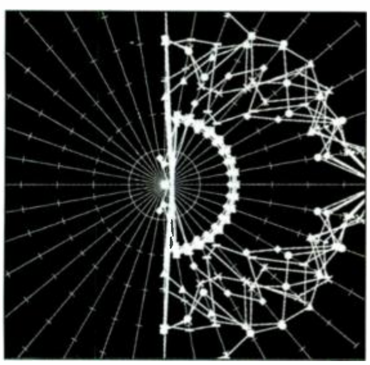
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Guest Editorial

VIRTUALITY

By Perry Sink

It was an Indian-summer afternoon in September 1995. I was driving south on Cicero Avenue near Chicago's Midway Airport when I saw what was, for me, the first public proclamation of a new era in technology. More importantly, it was a sign that the way we communicate was about to change forever.

It was a Southwest Airlines billboard. One thing about it was different than any ad I'd seen before: at the bottom, it read *http://www.southwest.com.* I recognized it as an Internet address and thought, "The Internet is now officially for everyone, not just grad students and computer hackers!"

And in only three years' time, that observation has proven correct. Now *every* ad includes a web address! The World Wide Web is arguably the most significant *social* development this decade. I ask friends, "Do you have an E-mail address?" and if not, their response is almost invariably "But I'm going to get one soon." Isn't it ironic that computer literacy, once a domain exclusively for nerds, is now the "in thing," and those who don't have a computer are considered nerds? Isn't it interesting that Bill Gates, computer nerd extraordinaire, is (for better or worse) the most famous captain of industry today?

At a dinner several weeks ago with three other couples, one person mentioned his plans to purchase his first PC. The discussion turned to megahertz, CD-ROMs, CRTs, gigabytes, kilobaud, and memory without missing a beat. What's remarkable is that all eight people at the table more or less understood all of those terms. High technology has become so widespread and inexpensive as to carry along even the average "man on the street" in its feverish pace.

Electronic communication has created a vast microcosm which simply did not exist a decade ago. I dare say that few people ten years ago anticipated the level of *connectivity* that has now become possible in such a short time. Who would have guessed that something as vast as the World Wide Web would be readily available to nearly anyone with a $1,000 computer and $20 a month for unlimited usage? Who would have guessed that in 1998 a student with a PC and a modem might instantly have, at his or her fingertips, more research information than is available at an entire college library?

A PEEK AT THE FUTURE

What does this have to do with building speakers? Anyone who's been in the audio industry long enough can attest to the fact that commercial speaker systems improved *drastically* between 1980 and 1990. *Speaker Builder* magazine itself had a part in this.

SB published practical implementations of Thiele and Small theory in a systematic, step-by-step fashion. It established a demand for complete specifications and parameters from manufacturers so that much of the guesswork was eliminated. Hobbyists oand manufacturers were no longer willing to plunk down $40 for a woofer just because the McGee catalog said it had "butt-kicking bass"; they now demanded proof and documentation.

Computer-design programs emerged, taking advantage of the education which builders now had and further increasing the designers' speed and accuracy. Tools such as FFT and MLSSA became a reality, and today, with a few hundred dollars of hardware and software, the hobbyist has more sophisticated tools than the largest manufacturers of only a few years ago. *Technology levels every playing field for those who participate.*

What's the next step? It won't be long before you can visit a manufacturer's website and download actual measurement data on any driver in its catalog. Imagine remotely simulating and optimizing an entire design, including room acoustics, plotting frequency, impulse and phase-response curves, and polar patterns, prior to purchase. You print out the bill of materials, which shows the cost, and the display asks whether you'd prefer to pay via Visa or Mastercard. Sounds crazy, but someone will do it. (Maybe they already have!)

The PC will also become the centerpiece of the home entertainment system. Beyond its obvious ability to play CDs, it also has tremendous digital signal processing potential. Why spend $1,000 on a dedicated D/A converter if a $50 piece of software running on your PC could do the exact same thing? Why would you buy or build a "traditional" active crossover when you could accomplish the same thing with a soundcard, and digitally compensate for every nonlinearity?

Then there is the music itself. Right now, a 56k modem provides about one-fourth of the data capacity necessary to transmit compressed hi-fi music in real time. Digital telephone lines surpass that somewhat, but new methods of transmission within telephone networks will be an order of magnitude better. Cable companies, in some locations, now offer Internet service via cable TV lines. The speed is more than ten times what is necessary for music. We're also witnessing the emergence of Internet service via satellite. We're not far from listening to real-time music via the Internet!

CROWDED AIRWAVES

I bet every reader knows at least one person who has tried (very hard) to "make it" in the music business, and failed. I can list a half-dozen. And it's not that they lacked skill. In most cases they were just as talented as the artists on the Top 40, but the fact of the matter is *there just wasn't room for them!*

Consider how often you hear the same old songs on the radio, over and over and over again. Sometimes you hear the same ten- or twenty-year-old song several times a day! I believe I read somewhere that Paul McCartney is worth $900 million. I'm not criticizing him for that, but doesn't it seem as though there should be room for more musicians in such a diverse world?

After considerable thought and numerous conversations, I believe I know why. It's simply because operating a radio station is very expensive—usually thousands of dollars per day—and there's bandwidth for only about 20 FM stations in any given area. Station managers can't afford to take risks, so they play only the music they know people like, and introduce new music slowly and cautiously. So there simply isn't room for very many people with musical gifts to express themselves and be heard.

Within five years, much of this will have changed. The entire music industry—and broadcasting industry—will experience a massive transformation because of the Internet. The revolution will begin as soon as quality real-time audio is transmitted and received via the web, probably one to two years from now. It is already possible to download music from the Internet; it's just terribly slow. But when we can all connect at 560k instead of 56k, watch out!

"Virtual radio stations" will pop up everywhere. "WebRadio" operators, not wanting to pay royalties to big record companies and organizations such as ASCAP, will play music with royalty-free agreements—and, of course, supply listeners with information on how to purchase the "unknown" artist's entire album, downloadable right from their website. A disc jockey (webmaster) will simply set up a schedule a week in advance, which will then automatically link to music sites throughout the week and provide 24-hour commercial-free programming. Got a special request? Coming up instantly! (Of course, it's only a matter of time before live video is included in this, too!)

This will help live music, because the bar or jazz club, whether halfway round the world or down the street, will broadcast that night's band over the Internet, inviting you to come on down and see and hear it live. If you're busy, you might simply record the show, complete with video, on your hard drive for later viewing.

The easy accessibility of recorded music will not make music less valuable. Rather, it will, like the radio which preceded it, serve to make the real, live performance all the *more* valuable. And in the same way that technology puts the hobbyist on equal footing with the large manufacturer, it will give the local artist potential access to the same large audience as the Rolling Stones.

I'm sure music-industry executives, awakened and alarmed by this grass-roots movement, will appear before Congress. They'll lament their fate and urgently press the government to do something about it. There will always be opposition to progress. Department stores and franchises were once illegal in some places. Galileo was nearly burned at the stake. The record companies tried to outlaw DAT and add a "duplication tax" to cassette tapes.

Many times, wonderful new inventions and technologies have been purchased and "buried" by companies whose existing investment was threatened. There will always be those digi-phobes holding on to the status quo, but they'll eventually go the way of the LP record. The old technology may be quaint, nostalgic, charming, and even surprisingly good, but a thing of the past, nevertheless. There's nothing more powerful than an idea whose time has come.

OF NOTE IN *Audio Electronics*

Issue 4, 1998

- New Approach to Lineamps
- Optimizing the S/PDIF
- A Passive Preamp
- The Zen Cousins
- Product Review: Marantz PMD320

VIRTUAL AUDIO

The biggest trend in the audio industry during the next five years will be the integration of the PC and the Internet with music and audio reproduction. You might start thinking of your CPU as a *receiver*. The soundcard will be your *preamp*. DSP software will serve as a *crossover* and time/frequency-domain *equalizer*. Your CD-ROM, of course, will be your *CD player*, and the zip drive will replace your *tape deck*. Think of websites as radio stations, and join a chatroom for a round-the-clock virtual audio hobbyist club.

Having worked as a professional in both the audio industry and the computer industry, I must say that the audio industry moves much too slowly in developing and using new technology. In audio, many so-called "new ideas" are really just recycled old ideas. But as audio merges with virtuality, watch out! Remember the impact the CD had when it came out 15 years ago? The next wave of technology will bring even greater changes as the PC permanently alters our definition of "hi-fi."

The audio hobbyist will be empowered in ways never dreamed of 20 years ago. The Heathkit builder has given way to the programmer, the hardware specialist, and the web surfer. The power of the hobbyist's tools has expanded exponentially. As we strive to embrace change, we're in for an exciting 21st century, with genuinely new innovation in communications and digital signal processing beyond our present imagination.

Meanwhile, those of you who are still a bit intimidated by computers need to remember that *action conquers fear*. Undoubtedly, we all have a friend or two who would be more than happy to take us by the hand and show us the wonders of modern technology, if only we'd ask. And those of us who are in the know need to remember what it was like *before* we knew—everything seemed hopelessly complicated until someone patiently taught us.

Coming up next century: The guy who said we could never have the true sound of a concert hall right in our homes is in for a surprise. *Viva Virtuality!*

ABOUT THE AUTHOR

Perry Sink spent three years designing speakers at Jensen and is now sales manager at Synergetic, a company specializing in factory automation networking. He remains active with the Illinois audio club, PSACS (Prairie State Audio Construction Society). He and his wife Laura are the proud parents of a baby boy, Cuyler Marshall, born on Aug. 25, 1998.

He can be reached at perry_laura@interaccess.com.

AFTERTHOUGHTS ON AFTERSHOCK

By Philip E. Abbate

One Saturday I was awakened by the sound of loud music and hammering. I wondered whether my wife Monica was trying to show me what it is like when I wake her with the same sounds. To my amazement, I walked into my listening room and there she was, taking out the wall of the adjoining room. Most wives would complain if your speakers are blocking their way; mine just made the room larger to accommodate me.

What a difference enlarging a 14′ × 15′ room into 14′ × 27′ makes. The system deserved the new room after all I have done to it in the last few months. Some of the major changes were the elimination of the triamp electronic crossover and also of Aftershock (*SB* 6/96). The little room was OK with a pair of JBL LE14 bass drivers in 50 ltr boxes running on a passive crossover, but in the big room I began to miss the low, low bass Aftershock once provided.

PHOTO 1: The Alusion 12″ driver in the Aftershock subwoofer enclosure.

BUNGEE EFFECT

I turned Aftershock off a few months ago because I could no longer handle the "bungee distortion" the Altec 421 8H LF drivers were generating. I call it bungee distortion because the very short X_{max} region of its underhung voice coil propels the cone into an incredibly long and uncontrolled stroke as though it were bungee jumping. I was looking for another 15-incher with which to replace the Isobarik Altecs when I came across the AlumaPro drivers in the Parts Express catalog. They looked massive and expensive.

I called AlumaPro's Matthew Honnert, who brought attention to the X_{max} limitations of the Altec ("Aftershock Objections," *SB* 1/97, p. 43), to see whether AlumaPro had a 15″ driver that could take the place of the Isobarik Altecs. AlumaPro does not make 15″ woofers, but Matt said his mid-tier Alusion 12″ woofer would probably do the job, so I tried one.

This driver is very substantial, using a cast frame made by Eminence for its professional-series sound-reinforcement speakers, and an aluminum cone with a treated foam surround. The voice coil appears to be 2″. This driver is a piston (*most are, I believe. —Ed.*), with an F_s of 28Hz, a Q_{ts} of 0.61, a V_{as} of 3.4 ft^3, and an X_{max} of 10mm. I modeled the driver with Boxmodel[1] in Aftershock's 111-ltr ported box tuned to 25Hz. It looked workable even without the sixth-order alignment (*Fig. 1*).

To install the 12″ driver in the Altec's 15″ hole, I fabricated an adapter ring from ¾″ MDF. My first listen was with the Woofer Wizard equalizer/crossover (also available from Old Colony) set to Aftershock's sixth-order alignment (about +9dB of gain at 20Hz in the circuit). The Alusion is a 4Ω driver, and the Isobarik Altecs are 8Ω each. I used one of the amp's channels to drive the Alusion, expecting the impedance difference to compensate for the volume change, but it needed to be turned up about 3dB louder than the Altecs.

I imagine that the Altecs at a low volume are more efficient than the Alusion. The difference would be that the Alusion does not compress at loud volume, and the Altecs do. The Altecs sound great as bass-guitar-amp speakers for music production (which is where they are headed), but as a hi-fi subwoofer, they just don't have the X_{max} for low distortion at low frequencies.

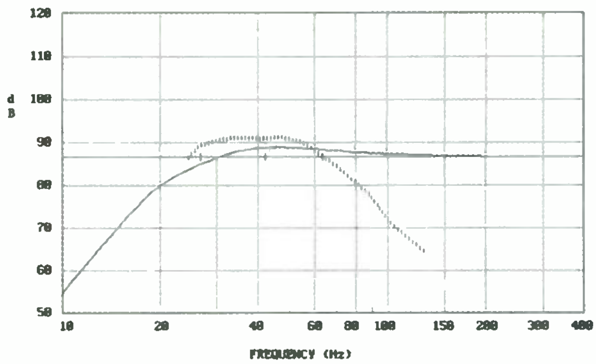

FIGURE 1: Real and predicted SPL seem to agree. Room gain and side-wall reinforcement not accounted for in the prediction. Measured @ 1W/4Ω/1m. Dash = measured response; solid = Boxmodel prediction.

THE ALUSION ADVANTAGE

One advantage of the Alusion over the Altecs is that it has less midrange leak-through. I think this is due to its lack of the rising response the Altecs display because I was using the Woofer Wizard to force them to operate in a region where they were not intended to go. I thought I needed a higher-order crossover on the top side of Aftershock with the Altecs, but with the Alusion, the Woofer Wizard's pseudo 18dB low-pass filters sounded adequate.

I added a little more gain to the Woofer Wizard and adjusted its equalizer for as little peaking as possible on the 20Hz high-pass section, which turned out to be about 2dB at 25Hz. Then when I listened, the Alusion provided a pleasing concussive bass that I could cross over as high as 165Hz (the limit of the Woofer Wizard; however, it was modified at the time of this trial) without hearing the harshness the Altecs had when I brought them above 70Hz. This single woofer in a 4 ft^3 ported enclosure had more deeper, cleaner bass than the two 15″ Altecs.

After a couple of weeks of getting used to this new driver, I measured its room response near the floor and one of the side walls with the Audio Control third-octave real-time analyzer. I was rather impressed with the smooth rolloff at both the top and bottom of the spectrum (*Fig. 1*, again).

Photo 1 shows the adaptor ring and the 12″ woofer mounted in Aftershock. I could get it only to 105dB with pink noise before the speaker started making some minor sounds of distress. This noise would be audible only if you were hammering the speaker unit with band-limited pink noise and had nothing over 165Hz playing through the system, since actual music would mask this noise.

What about Aftershock? Well, I will use it with the Woofer Wizard, although I think it will work fine without an electronic crossover as the low-frequency-effects speaker in my Dolby Digital® system... when I get it.

REFERENCE

1. Boxmodel and the Woofer Wizard are available from Old Colony Sound Laboratory, PO Box 876, Peterborough, NH 03458, 603-924-6371, FAX 603-924-9467, E-mail custserv@audioXpress.com.

AUDIO VIDEO REVISITED

By Patrice Pelletier

I read with great interest Mark Zachmann's article, "Simple High-Quality Computer Speakers" (*SB* 8/97) and would like to share my own experience. Playing movies on a hi-fi VCR can be surprisingly demanding for a TV mini monitor; I blew mine during a plane crash. So I decided to build a new and more rugged one.

There are not many choices for shielded speakers, so I too chose the excellent Vifa M13SG-09-08. But for a complement, I used the Seas 25TAF/DTV, a shielded, ferrofluid-equipped metal-dome tweeter. *Table 1* summarizes the measured Thiele/Small parameters of the woofer after a 48-hour break-in.

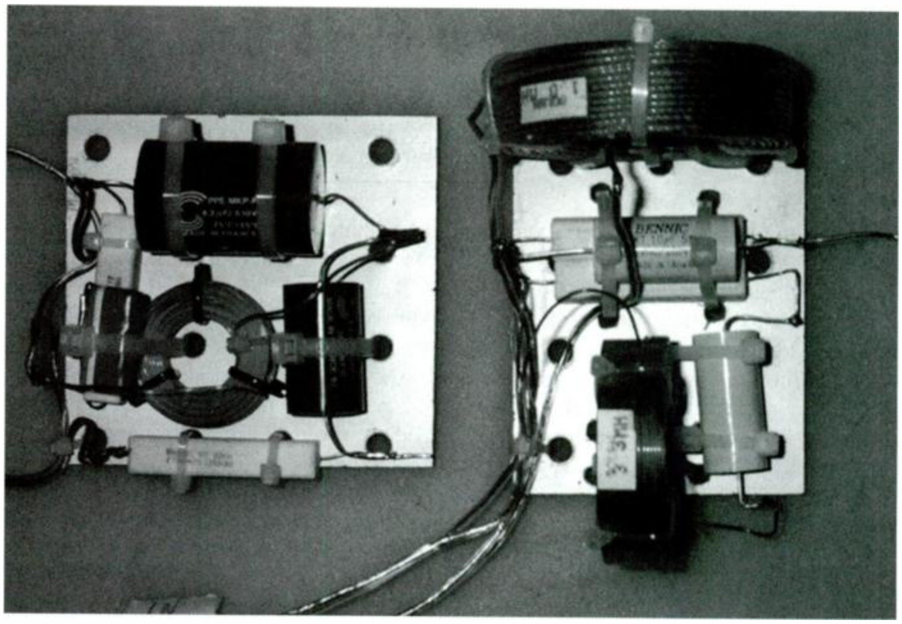

PHOTO 1: The crossover: the high-pass section with a single coil (left), and the low-pass section with two coils (right).

SIZE AND CONSTRUCTION

Adding 0.3Ω series resistance and using the new Q_{ts} resulted in a closed box of 4.4 ltr with a Q_{tc} of 0.7 and an f_3 of 96Hz, or a 9 ltr vented box with an f_3 of 56Hz. Since I wished to reuse the original TV brackets, the box needed to be compact and lightweight, so I opted for the closed box.

I built it with regular 11/16″ particleboard covered with black Formica® to match the TV set. Height and width are essentially the speakers' dimensions, plus space for a possible grille. I chose the depth to complete the volume requirements, allowing space for the crossover and Dacron® stuffing effect. I mounted the tweeter flush, but not the woofer. Once finished, the overall dimensions were 6 1/8″ W × 11″ H × 7 7/8″ D.

ABOUT THE AUTHOR

Patrice Pelletier, now 39, has worked for Bell Canada in Montreal as a computer communications technician since 1981. Since the age of 15, he has been fascinated by speakers. Building loudspeakers for friends and electronic components for himself is his preferred hobby.

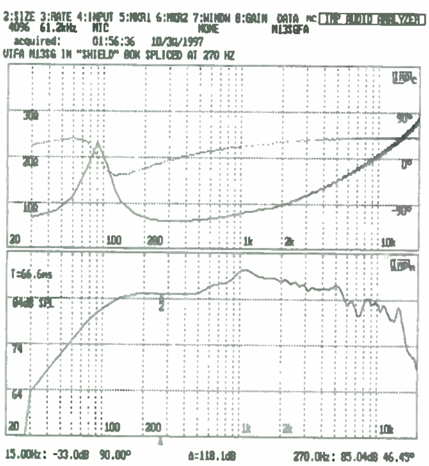

FIGURE 1: Frequency response and impedance of the boxed M13SG-09-08.

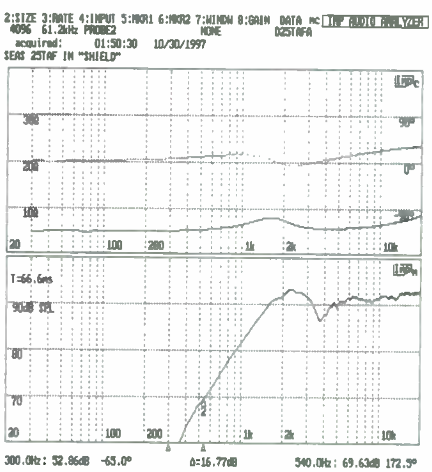

FIGURE 2: Frequency response and impedance of the boxed SEAS 25TAF/DTV.

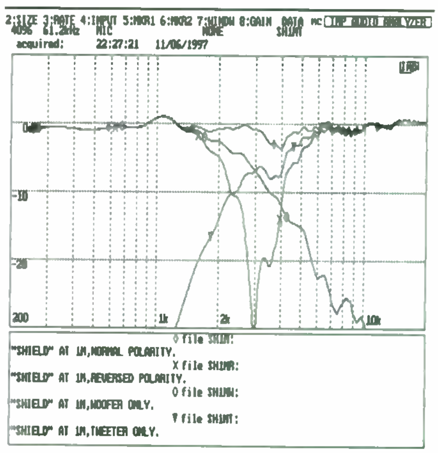

FIGURE 3: How the speakers combine at 1m.

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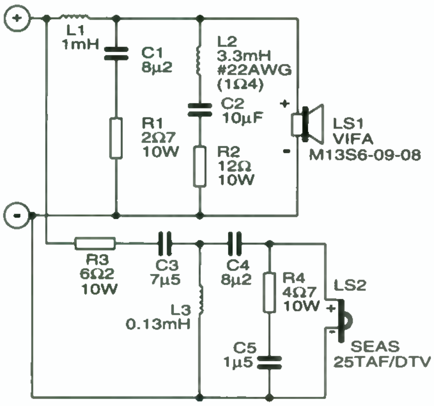

FIGURE 4: Crossover filter.

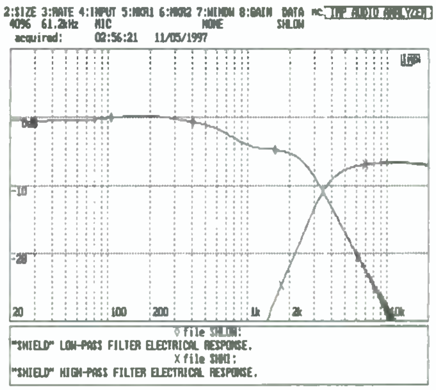

FIGURE 5: Electrical response.

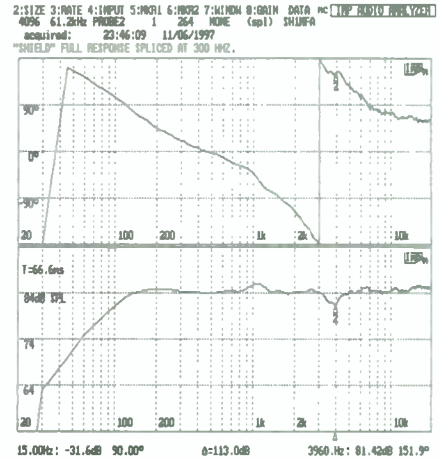

FIGURE 6: The complete frequency response.

GRAPHICS

Figure 1 is the basic 2.83V/1m SPL frequency response (FR) of the boxed M13SG-09-08. The close-miked response below 270Hz was spliced to a gated IMP MLS response. Note the 5.5dB step from 400Hz to 1kHz, shown partially equalized in Fig. 1, *SB* 8/97, p. 12. *Figure 2* is the basic 2.83V/1m SPL FR of the boxed Seas 25TAF/DTV. There is a bump at 2.5kHz and a suck-out at 4kHz that are typical of a high-frequency diffraction artifact (mounting the tweeter off-axis in mirror-image might have yielded a smoother FR).

Figure 3 shows how the speakers combine at 1m using the 3kHz crossover filter of *Fig. 4*, and *Fig. 5* is the crossover's electrical response necessary to equalize the step of the woofer and to pad and protect the tweeter. *Figure 6* is the complete FR at 2.83V/1m spliced at 300Hz. The average SPL is 84dB, essentially flat from 150Hz–20kHz, with two minor incidents at 1kHz and 4kHz.

In *Fig. 7*, you can see that the hole at 4kHz is partially filled "off axis," confirming a diffraction effect. *Figure 8* is the FR at 2m (who listens to music at 1m?). Compared to *Fig. 3*, the response is smoother, and the bump at 1kHz is not there anymore. This may be due to the earlier truncation of the pulse response removing some small echoes. Note that the slopes combine at –6dB and their rate is around 18–24dB/octave. Reversing polarity produces a sharp null as in a fourth-order Linkwitz-Riley crossover.

TABLE 1

MEASURED T/S WOOFER PARAMETERS

M13SG-09-08	#1	#2	MANUFACTURER'S
F_s	53.8	55.8	54.0
Q_{es}	0.452	0.455	0.46
Q_{ms}	1.71	1.75	1.50
Q_{ts}	0.358	0.361	0.35
R_e (Ω)	5.45	5.45	5.60
V_{as} (L)	10.6	9.73	12.0
L_e (mH)	0.66	0.67	0.70
SPL (dB)	88.8	88.9	88.0

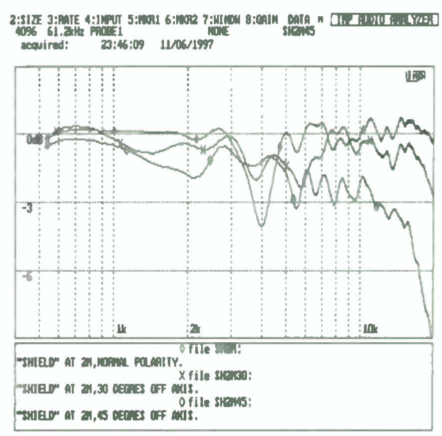

FIGURE 7: Measurement confirming a diffraction effect at 4kHz.

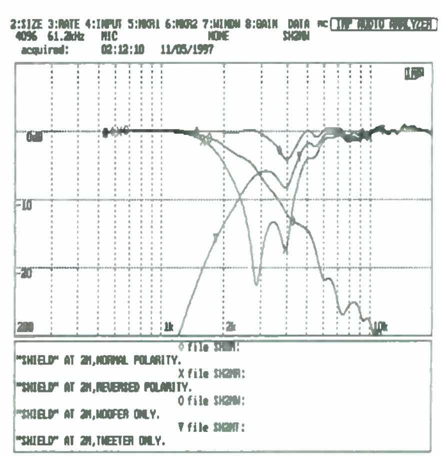

FIGURE 8: Frequency response at 2m.

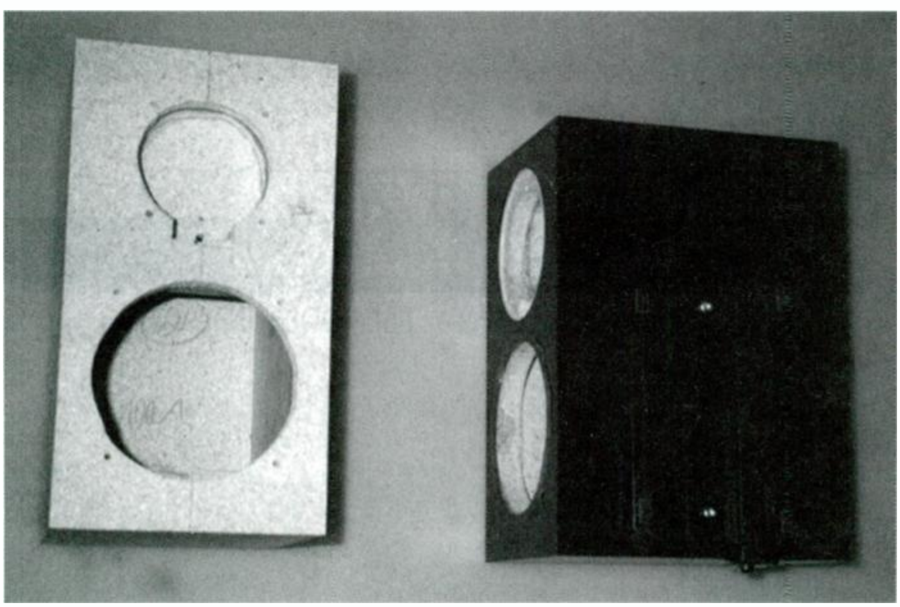

PHOTO 2: Unfinished box (left) and finished box (right).

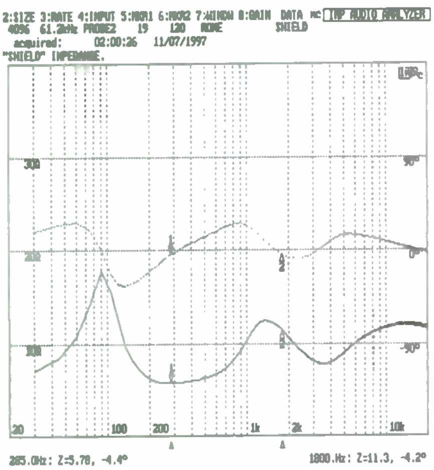

FIGURE 9: System impedance.

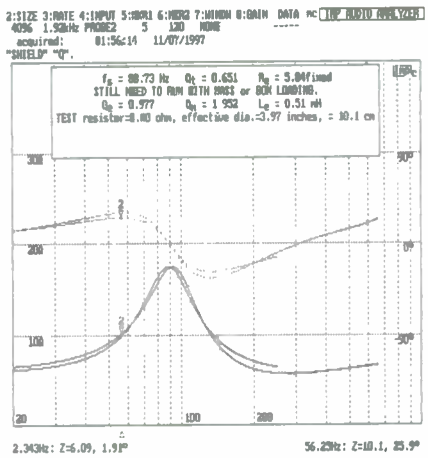

FIGURE 10: The Q_{TS} for the complete system.

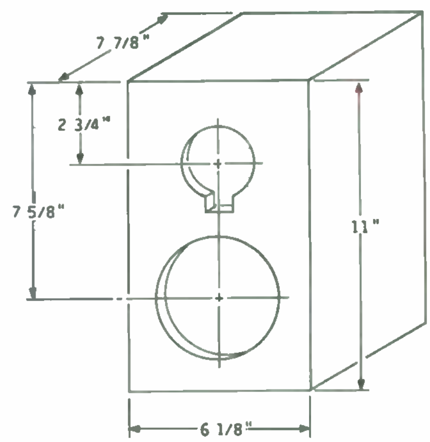

FIGURE 11: Enclosure dimensions.

Figure 9 is the system impedance, the minimum being 5.7Ω at 285Hz, with smooth amplitude and phase variations—an easy load for any power amp. *Figure 10* shows a Q_{ts} of 0.65 for the complete system, a little overdamped due to the slightly oversized depth, and *Fig. 11* shows the enclosure dimensions.

LISTENING

In my regular system, the presentation is smooth and detailed, with a good and stable stereo imaging. Voices sound very familiar. Compared to my Sapphire II (*SB* 8/90, p. 56)/Sub-1 reference system, very low deep bass is missing, but this is not at all annoying, since the smooth-rolloff, low-Q_{TS} closed-box system integrates well with the room and compensates for it. The power handling is excellent, and the sound remains clean at high levels.

Used as TV monitors, they easily reveal differences in quality of channels or programs. Watching video with "full sound" is very entertaining, but for me the best part is using them with my hi-fi VCR. I enjoy watching films at realistic levels and not needing to turn down the volume during the special effects. Finally, if you wish to add a single passive subwoofer, the Madisound dual-coil 1252DVC would seem to be an excellent candidate.

DANGEROUS MUSIC

By Jesse W. Knight

Quite frankly, I wish that what I am about to express were not necessary. We hear of new dangers to life and limb every day, and to add music is disturbing at best. Fortunately, here you can measure the factors that produce the danger.

By returning to the old concept of high fidelity, which is to re-create live music with the same spectral balance (tone), dynamic range (PPPP-FFFF in musical notation), peak level (volume), and freedom from distortion that you hear in the concert hall, you can safely enjoy acoustical music. At lower levels, processed music is dangerous, requiring a more technical approach.

DEAFENING NOISE

CNN recently broadcast a story on deafness. Baby boomers (President Clinton included) are becoming deaf in record numbers, and loud music and a noisier world were cited as the causes. All of us are at risk, and those who are speaker builders can be an important part of the solution, since we have the knowledge and the tools to understand one cause of this epidemic.

A good objective is to limit exposure to sound-pressure levels (SPL) of over 90dB (*Table 1*). This means holding peak levels to 90dB for hard rock and heavy metal, or 100–105dB for classical listening. Compressed music contains more SPL/hours than uncompressed music, and therefore is dangerous at lower peak levels.

Classical music with wide dynamic range produces fewer SPL/hours and therefore is safe at higher peak levels. During soft passages in the music, your ears recover a bit, as well. Spectral balance is the other important factor in hearing loss, and here the treble is much more dangerous than bass.

Remaining considerations are moral and ethical in nature, and while very important, have little to do with hearing loss. Criticizing music on moral grounds stamps it as "banned in Boston," often increasing sales. The examples below will give you a hint of the slippery slopes involved in music criticism.

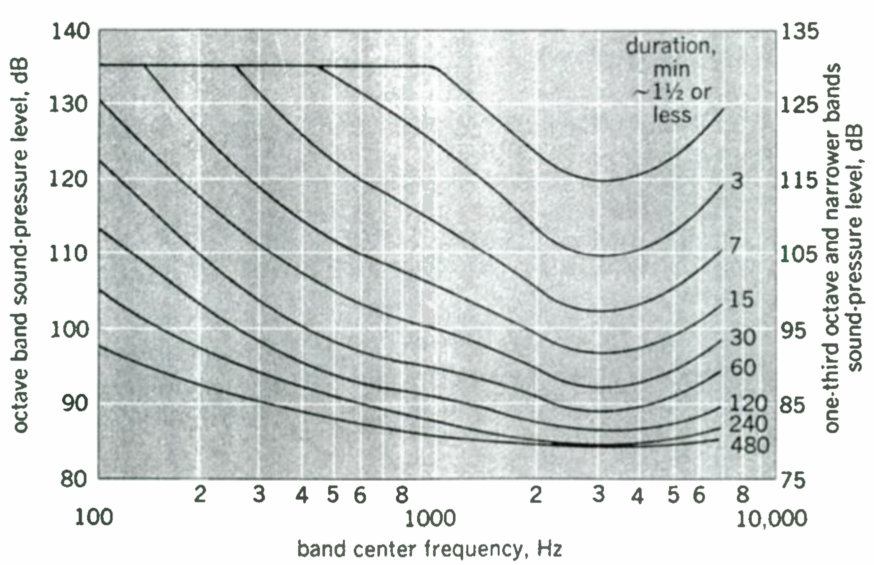

FIGURE 1: Damage-risk contours for one exposure per day to one-octave (left-hand ordinate) and one-third-octave or narrower (right-hand ordinate) bands of noise. This graph can be applied to individual band levels that are present in broadband noise. (Used with permission from *Encyclopedia of Science and Technology:* "Acoustic Noise," McGraw-Hill, Vol. 1, p. 80, 1997.)

THE GOOD AND THE BAD

Music snobs have for years tried to condemn rock music by saying it is the work of the devil, that it lacks structure, or that it is too repetitive. If you say that old music is better than new, you have neglected the "filter of time" that quietly lets inferior music be forgotten. Try listening to all 200 or so Bach cantatas if you doubt me on this. I pick the cantatas because they have all been recorded—good, bad, and indifferent—presenting old music unfiltered by time.

Being a church musician has exposed me to all manner of arguments, ad nauseam. These arguments don't hold up when carefully examined, and most of the ethical problems are not new. Mozart wrote some songs that, sung in English, would be right at home on *South Park* or *Beavis and Butt-head*, two recent TV cartoon shows that offer something to offend everyone.

Rock artists have written sacred music and many songs of exceptional sensitivity, while opera provides some extremely bad examples, to say the least. Schubert's song "Der Erlkonig" was as terrifying to 19th century German kids as today's worst horror movies are to ours. There is music from all periods that is unsuitable for children; however, today it is much more available to them, since they have economic means in excess of their ability to choose wisely. I will leave this question to theologians, psychologists, and child-care professionals.

There is a theory that classical music raises the IQ of young children, and music therapy has growing support in medicine. Research will, I believe, find much music good, and give solid guidance that is sorely needed.

LOUDNESS LEVELS

Ironically, the hazard to the ears posed by much modern music has been largely overlooked or underestimated. Most rock recordings and performances have only one dynamic level—very loud. Classical performances may reach high levels for a limited time, but then the sound levels drop off. You can prove this by graphing a CD's recording level as displayed on your cassette recorder's VU meter. The CD time-code

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ELECTROSTATIC SUBWOOFER

Let's face it, some of you are hard-core bass addicts. You're not happy unless the walls are shaking and the neighbors are complaining!

I don't care how much you paid for your subwoofer, it will never come close to the electrostatic bass produced by SHOCKWAVE. It makes ALL subwoofers sound like boomy mush. No matter what type of system you own --- conventional box speakers, horns, electrostatics (ESLs), planar magnetics, or dipoles of any type, SHOCKWAVE is the ultimate solution to attaining extremely deep and detailed bass.

Everyone recognizes electrostatic speakers for their unparalleled clarity and accuracy and SHOCKWAVE extends these attributes to the INFRASONIC (10. Hz).

This milestone product mates with, and vastly improves, ALL types of loudspeakers and you only need one unit, even if you desire two channel bass.

SHOCKWAVE uses the latest technology to produce articulate but EAR-SHATTERING bass. Phase cancellation (anti-noise) principles plus compound or parallel diaphragms, give SHOCKWAVE the equivalent electrostatic surface area of over 108 square feet!

Truly EXPLOSIVE acoustical power that exceeds 120. db at your listening position. And it only takes up 1.5 square feet of floor space in a "V" shape. The height is just 36 inches and it may be placed in ANY corner, behind one speaker, or at the center of your two speakers.

But the best part is that it connects into your system without the need of conventional active or passive crossovers (set your crossover point with a simple resistor from Radio Shack). AND YOU DON'T NEED A SECOND AMP TO DRIVE IT! Use your current amp for stereo or mono bass. Frequency response is 10. Hz to 500. Hz. Nominal impedance is 8 ohms and it requires only 30. watts to drive it (300. watts max.).

Disclaimer:
We cannot be held responsible for any vibrational damage done to your home or contents.

SHOCKWAVE is fast and easy to build and most of the parts can be obtained from your local hardware store. Depending on your system, materials on-hand, exterior trim and grill cloth, SHOCKWAVE will cost between $100. to $500. to build. But most of you will end up spending about $250. And please don't think that its low cost translates to low performance, since this design would sell for at least $5,000.00 to $8,000.00 at your local Hi-End audio shop!

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readout provides the time base of the graph.

I measured the level versus time for the first movement of Bruckner's 9th Symphony (see sidebar), since many regard this as a particularly bombastic work. By doing so, I think I was testing symphonic music that is a good deal more of a hearing risk than the average symphony. I found that the average level is about 10dB below the peak levels. For approximately 15% of the time the level was −28dB or lower. It is interesting that late Romantic composers tended to write softer passages than earlier composers.

I suspect that at least during a live performance, the risk to hearing is restricted to the musicians and maybe the first few rows of the audience. If you further assume that a concert goer is exposed only to one or two concerts per week, the risk is negligible. Hearing damage is cumulative just like damage from X-rays (*Fig. 1*).

TABLE 1

AUDIBILITY
10dB—a whisper
20dB—quiet conversation
30dB—normal conversation
40dB—light traffic
50dB—a typewriter
60dB—a noisy office
70dB—normal traffic; a quiet train
80dB—raucous music; the New York subway
90dB—heavy traffic; thunder
100dB—pneumatic drill
110dB—rock band on stage
120dB—jet takoff (at 100m)
130dB—jet engine (at 25m)

What's more, consider that during a typical concert season, many works will not reach the sound levels typical of the late romantic works such as the Bruckner. Operas are longer, but the limitations of the unamplified voices keep levels much lower than symphonic works.

LISTENING AT HOME

When you listen at home to classical recordings, the danger is increased because listening time typically is not limited to three to six hours a week. Worse, homes are in general not as quiet as a concert hall. In order to hear the softest passages of the music, you turn up the volume, thus making the loudest passages more dangerous.

To me, the absolute level of music is not what makes it dramatic; rather, it is the dynamic range and artistic content. Many sources give a maximum dynamic range of 60dB for music. If you build a quiet listening room where you can hear 35dB SPL, then the loudest music need be no higher than 95dB SPL.

Going back to the Bruckner symphony, it becomes evident that you will be exposed to more than 90dB only about 25% of the time. In a less quiet room, turning up the volume 10dB to mask background noise both increases the peak level to 105dB SPL and lengthens the time you are exposed to more than 90dB to around 50%. Hearing damage is now a possibility if you listen for long periods (*Fig. 1*).

You can also limit exposure to excess sound levels by playing music of limited dynamics at a lower level. It is easy to forget that to make a good recording, the engineer will adjust the record level during the loudest passage of the music to the highest level the recording media will handle without distortion. Not doing this reduces the number of bits on a CD, thus increasing noise and distortion. The listener then must use the volume control to re-create a realistic level for each recording. For example, a Mozart symphony played by a 30-piece orchestra should not be as loud as a late Romantic symphony using 120 musicians.

EXCESS TREBLE

Excess treble levels from stereo systems can also damage hearing. This can happen in stages, so it is vital to recognize the early warning signs. Never assume that a speaker system has a flat response based on T/S parameters, driver or room measurements, and so on. Tweeters and midranges often need to be padded down several dB.

I think the best way to monitor this is frequently to attend live unamplified concerts. Recordings that utilize close-up microphones may not incorporate the treble losses incurred by the sound as it travels from the stage to your seat. Assuming you sit 33′ from the instruments, losses at 10kHz will vary from about 2.8 to less than 1dB, depending on both temperature and humidity in the hall.

This confirms an old observation that hot, humid conditions give a brighter sound. Hot and dry is the most lossy for highs. Any concept of a correct absolute balance is blown away.

In balancing recordings, conductors tend to favor the brighter sound at the podium over that which the audience hears. Telarc, at least in the few samples I have tested, has resisted this practice. If you start sensing that the live music is too soft, turn your home stereo volume down. If the highs sound weak in the concert hall, turn down the treble control when you get home.

This is far better than waiting for the next indication of deafness, tinnitus. An equalizer allows for a better match, but requires much patience to use correctly. When using good drivers, I find tweaking the crossover a more effective way to equalize a system.

LISTENING TECHNIQUE

I am going to indulge now in a hypothesis that is far from complete and may be somewhat inaccurate, but I am at a loss to explain my listening technique in more concrete terms. It is based on the well-known principle that your brain goes to great lengths to make a consistent and sensible picture of your environment. Your ears (and brain) are conditioned by whatever you hear most; in other words, your brain tries to normalize what you are currently hearing so that it agrees with what you have heard in the past.

Distortion or response errors in a stereo system produce sounds that are not heard under normal circumstances. This results in stress known as listener fatigue, because the brain is working so hard to fit what is heard to a known pattern of sound. Electronic distortion correction is not a part of the brain's extensive algorithms that effortlessly remove the effects of sound reflecting from the walls near someone who is speaking to you.

Your hearing, a product in part of 100,000 years of natural selection, does a good job of saber-tooth-tiger detection as a result. Unfortunately, speaker builders have, by necessity, an adaptive process to compensate for ear wax and aging that seems to turn up the treble in the brain to maintain a constant sense of the sounds around you.

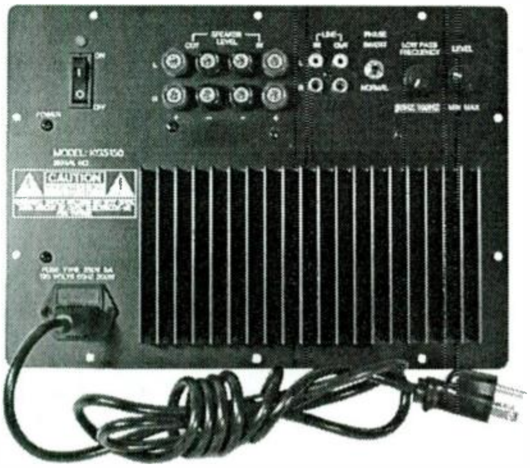
For example, after removing a large amount of ear wax, you will hear too much treble until your brain resets. I can only play around with an equalizer for a few minutes before this process starts. Wrong then starts to sound right.

I have learned to sense this happening after years of contemplating the problem. Unfortunately there is no "bypass switch" I know of to turn the ear into a stable measuring device. Conditioning to excess treble is fully reversible if caught early, before actual damage is done. Treble addicts betray themselves by suggesting that live music is not bright enough, that something was wrong with the performance.

TASTE FOR NOISE

Rock music is often compressed to the point where the only drop from peak level is the fade-out at the end of the song—and the all too short break before the next song. Many compression algorithms are available, and the ones that sound the loudest become the favorites in many rock studios (teenage taste for noise "rules").

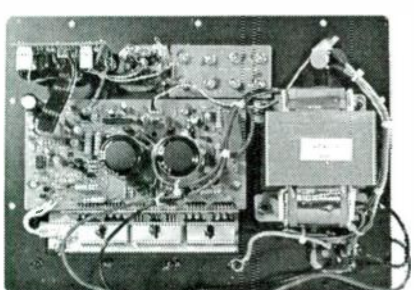
Sometimes clipping is used to make a recording even louder than the effects of compression alone. The devices that do this are called "toys," which raise the RMS level relative to peak as well as adding greatly to the high-frequency content. To add insult to



Subwoofer Amplifier 150/200 Watts

Madisound is pleased to offer the **KG-5150 Subwoofer Amplifier with Electronic Crossover** for powered subwoofer systems. We have worked hard to find an amplifier that has an adjustable electronic crossover and has continuos power of 150 watts into 8 ohms or 200 watts into 4 ohms. We are confident that we now have a product that can meet the demands of both home theater or high end audio systems. Powered subwoofers are a versatile addition to your audio system. With the variable crossover and volume control, you can match a subwoofer to any existing system. The **KG-5150** can be used with single or multiple woofer systems (a total impedance of 4 to 8 ohms is recommended). **Madisound** can help you choose an appropriate woofer that will meet your audio and architectural needs.

- **Discrete high current drivers and output devices**
- **LED power indicator**
- **Woofer volume control**
- **50Hz to 150Hz continuously adjustable low pass 12dB electronic crossover**
- **Low and High level input, summed to mono signal**
- **Low and High level output**
- **Phase inverter switch**
- **Master power switch**
- **Auto power on/off activated by input signal**
- **Thermal protection auto shutoff**



Specifications for KG-5150 Subwoofer Amplifier	
Power output @ 8 ohms / 4 ohms	150W / 200W
THD	0.025%
S/N ratio @ rated power	95dB
Input sensitivity @ 100Hz - low level	150mV
Input sensitivity @ 100Hz - high level	2V
Input impedance	22k ohms
Auto on sensitivity	5.8 mV
Time for Auto Off	8 minutes
Weight	10.5 lbs
Dimensions W x H inches	$11\ ^{7/16} \times 8\ ^{1/4}$
Cutout hole W x H inches	$10\ ^{7/16} \times 7\ ^{1/4}$
Price Each	**$169.00**

Woofer Suggestions

Item	Size	Ft³	S/V	F_3	W
Madisound 1252DVC	12"	3.5	S	30	75
Madisound Swan 305	12"	4.25	V	28	200
Eclipse W1238R	12"	3	V	28	200
Peerless 831727	10"	2.25	V	30	220
Peerless 850146	10"	3.1	V	27	220
Peerless 831857	12"	6	V	24	220
Scan-speak 21W/8555	8"	1.8	V	31	100
Scan-speak 25W/8565	10"	3	S	34	100
Scan-speak 25W/8565-01	10"	3.5	S	28	100
NHT1259	12"	3.5	S	29	300
Vifa M26WR09-08	10"	2.2	S	35	130
Dynaudio 30W100	12"	5.5	S	30	130
Dynaudio 30W100XL	12"	4.5	S	37	130

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injury, the live reference for rock is almost always too loud.

Like prohibition, legislating sound-pressure levels in discos has not been a success. Drinking alcohol and smoking increases the damage done by loud music, partly through vitamin-B depletion and reduced awareness that the music is too loud. At high levels, sound becomes drug-like in its action on the body and mind. You become conditioned to a given level, and then want more to get the same "high."

After a period of time, the collective listener's high-frequency hearing is reduced, and the musicians respond by cranking up the treble more and more. It is a perverse reality that treble is cheap and bass expensive in the world of speaker manufacture. A point of no return is eventually reached where the listener cannot return to a concert hall and enjoy acoustic music without a hearing aid.

I am aware of no other art form that destroys your ability to enjoy earlier forms of the art. Certainly there are art forms that attempt to discredit earlier art, and deconstructionist philosophies (Nietzsche's *Hammer of Questions* comes to mind) that ask us to throw away old dogmas so we can freely test new thoughts. No matter how much Nietzsche you read, you will not go blind any faster than reading any other book. You might repent your sins, leave the church, or decide to accept more than one view of life. Hearing loss is often permanent, and a high-fidelity hearing aid is an oxymoron.

Anyone listening to or playing loud rock should obtain a sound level meter and limit levels to 90dB or lower. Bear in mind that live levels for rock music are often in the vicinity of 110dB, a level that is only safe for a few minutes a day.

COMPARING TWO WORKS

To assess the risk of listening at a 105dB peak level, it is necessary to plot linear SPL rather than dB, which is logarithmic. I compared two musical pieces: Bruckner's *9th Symphony* and Anthrax's *Spreading the Disease* (*Figs. 2* and *3*). (For the sake of clarity, in *Fig. 3* I plotted the music out of sequence—by ascending level.)

Risk is often defined as being the product of linear level (Y-axis) and exposure time (X-axis) for broadband noise. Therefore, you can regard the risk as the area under the plot of the music. The recommended daily limits are eight hours at 90dB, 24 minutes at 103dB, and 16 minutes at 105dB.

The first movement of the Bruckner piece—23.25 minutes—uses up only one-third of your daily sound allowance, indicating that you are probably at no risk from listening for an hour at this level twice a week. Calorimeter measurements—which measure true energy levels over time, regardless of frequency or waveform—indicate that the Bruckner is less dense, suggesting that only a quarter of your allowance has been consumed.

Anthrax, with a very high energy level, consumes your entire daily allowance in 23.25 minutes. After an hour of listening at this level, you are likely to increase the level by 3dB or so to compensate for the temporary deafness that results, thereby getting the maximum safe dose every 13 minutes.

Hearing damage is a given. There is nothing exceptional about the Anthrax recording; any highly compressed music is dangerous at this level, and it need not be heavy metal.

—JWK

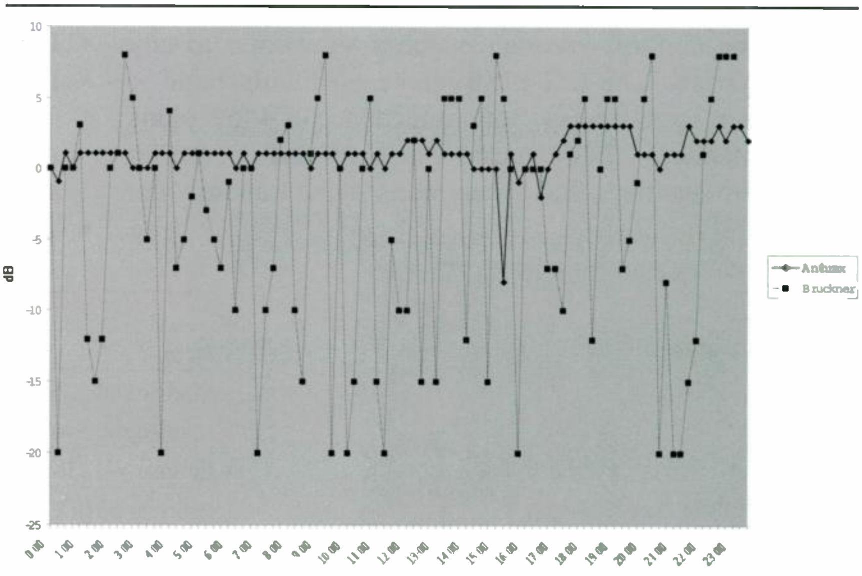

FIGURE 2: Peak sound-level graph, as displayed by VU-meter.

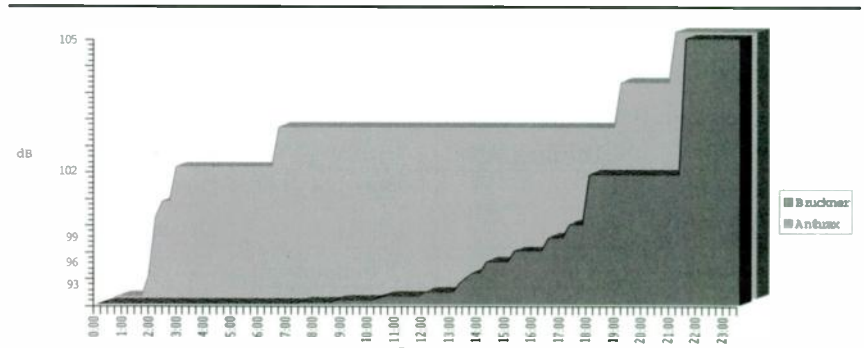

FIGURE 3: Sound-level comparison of Bruckner and Anthrax pieces; the graph is linear for SPL power.

FOCUSED ARRAY ELECTROSTATIC, PART 2

By Bill Waslo

In Part 1 I presented the concept and goals of my experimental Focused Array Electrostatic (FAE) design. I follow up in this article with technical information about measurements and construction—I encourage readers to use or discard the techniques presented here to improve upon the design.

CONSTRUCTION

I will not discuss the construction of the bass cabinets, about which there is nothing remarkable. Were I to start over, however, I would make each tower narrower and as a single unit with inner dividing walls, rather than as a stack of separate modules. Designing a box such as this should not pose any particular problems. The only other advice I would offer is to be sure to brace very securely any large inside surfaces between walls.

Constructing the frames that hold the ESL panels in the curved arrangement was easier than planning how to do it. The frame was made from pine, and the design was based on the dimensions of inexpensive wood pieces available at the local building supply house. All the pieces were cut using a continuously adjustable hand miter saw that could accommodate boards of 5.5″ width.

I designed my FAE for a focus distance of 5′ (every panel is designed to be 60″ from the listener's ear). In practice when listening, the speakers seem much closer than I had envisioned. It may be advisable to design for a longer focus distance, although this would result in less effective echo minimization.

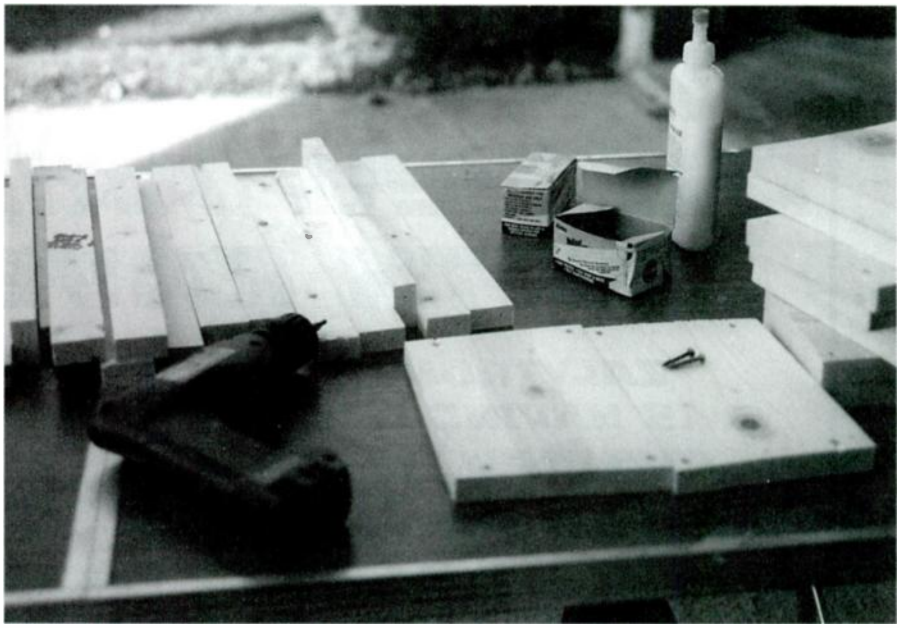

PHOTO 2: Small wood frame pieces and tools awaiting assembly (Photos by Darek Ball).

The sides are made from lumber of about 5.5″ × 0.75″ × 48″ (sold as 6″ × 1″ × 4′). To make the arcs for each side of the frame, I cut nine pieces (*Fig. 8*), and used the miter to make the 94° angle cuts at the ends, which causes the entire assembly to assume the arc shape when put together.

The side pieces connect to each other, left side to right side, using horizontal pieces of about 1.5″ × 0.75″, cut to a length of 15.25″. I used #8 × 1½″ square recess construction screws—along with the screws' mating bit in a cordless electric drill/driver for quick construction—and wood glue. The side pieces also connect end-to-end and are glued and screwed to the ends of the horizontal pieces. This is sketched in *Fig. 9*, in which only the ends of the horizontal pieces are shown (the lengths extend perpendicular from the page).

I predrilled each of the side pieces so that the horizontal pieces—connected near the

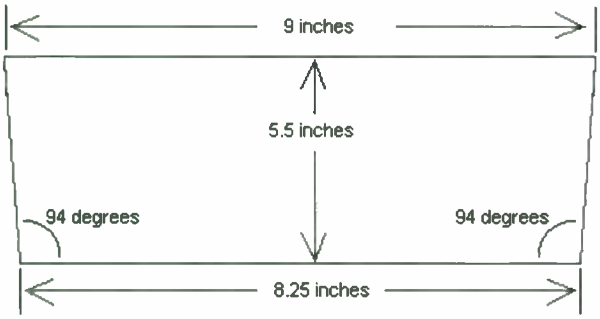

FIGURE 8: Dimensions for frame side pieces (18 required per frame).

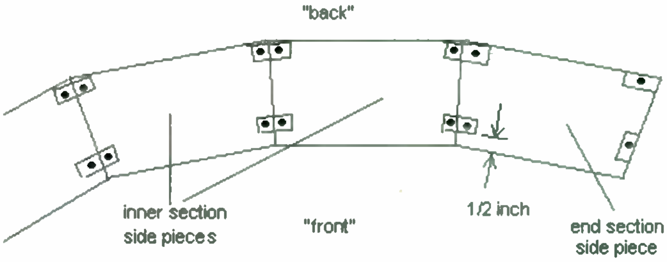

FIGURE 9: Interconnection between side pieces.

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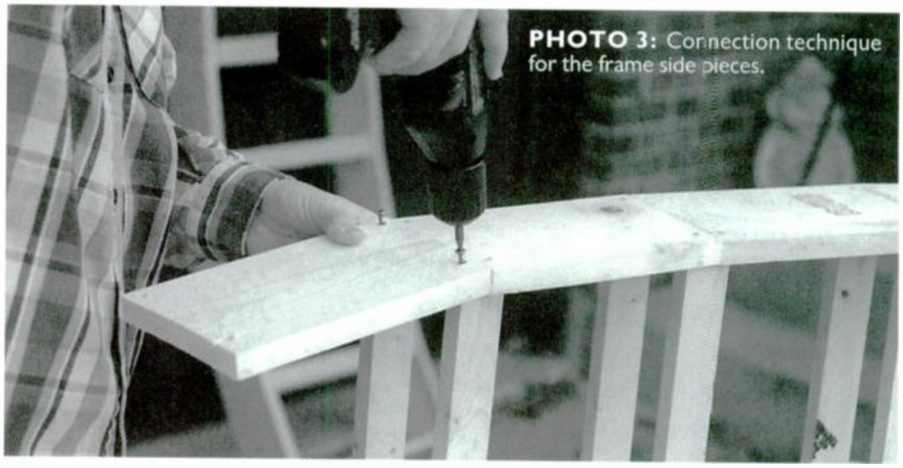

PHOTO 3: Connection technique for the frame side pieces.

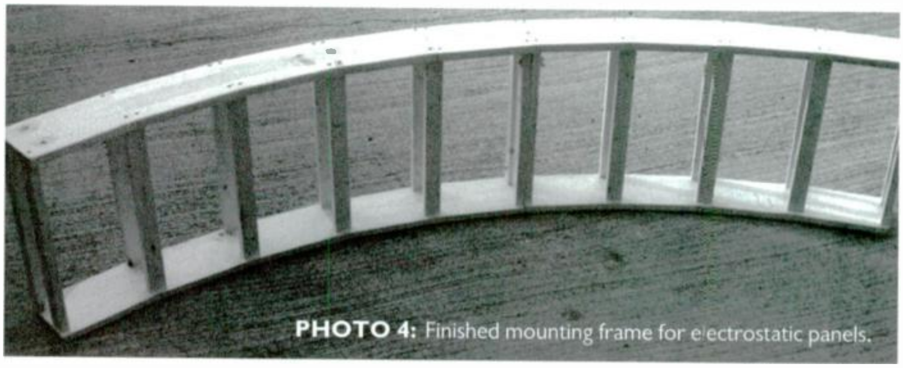

PHOTO 4: Finished mounting frame for electrostatic panels.

corners of the "front" (8.25″) edge of the side pieces using the screws—provide a mounting surface recessed ½″ from the front, for mounting the electrostatic panels. I also predrilled near the corners of the back (9″) edge so that the horizontal pieces lie nearly flush with the back surface. *Photo 2* shows some prepared wood pieces and tools, ready for assembly into the frame.

FRAME ASSEMBLY

When connecting a new side piece to an adjoining side piece already connected to the horizontal piece, I tilted the new piece slightly down from the intended seam. Then I drove the screw at a slight angle (but perpendicular to the new side piece), so that the new piece pulled tightly against the already connected piece (*Photo 3*). Some care was required to minimize any "twist" that developed in the frame during assembly. I tried to match the left and right side pieces at each section to lessen accumulated imbalances in the two sides. The completed frame resembles a curved ladder (*Photo 4*).

After assembling the frame, I used an electric sander with coarse paper to round off all of the exposed corners. This is a practical operation with a soft wood such as pine. I then used fine sandpaper to smooth the outside of the frame and applied several coats of polyurethane. The overall appearance is quite nice and, with the addition of grille material, should be quite presentable.

PHOTO 5: The completed FAE speaker system (back view).

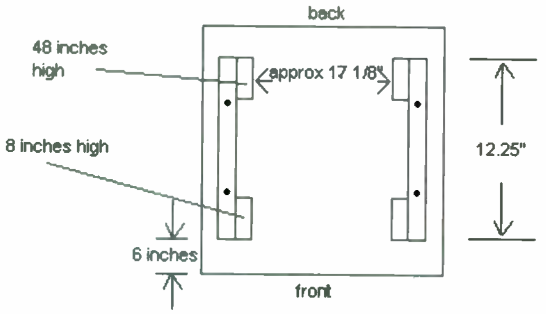

FIGURE 10: Base assembly for ESL frame.

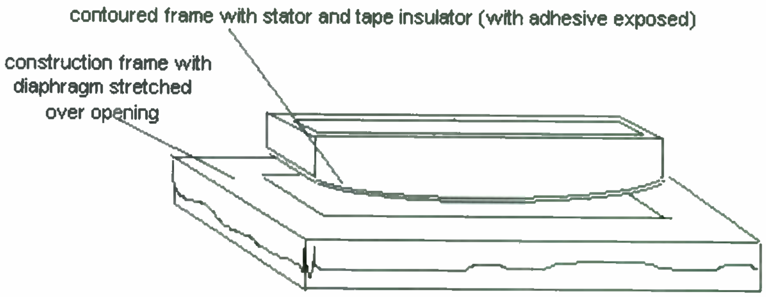

FIGURE 11: Installation of the diaphragm to the curved tweeter frame.

To mount each frame, I made a simple stand using a 2ft^2 piece of white laminated particleboard as a base and lengths of 3.5″ × 0.75″ (4″ × 1″) pine (*Photo 5*). At each side of the frame I used an 8″ vertical length to connect (using screws and wing nuts) to the front bottom of the frame and a 48″ vertical length to connect to the back (at the fourth cell from the top). These two lengths connect at the bottom to a 12.25″ length that I had mounted on its 1″ wide surface, running front-to-back on the particleboard base (*Fig. 10*) and affixed with construction screws.

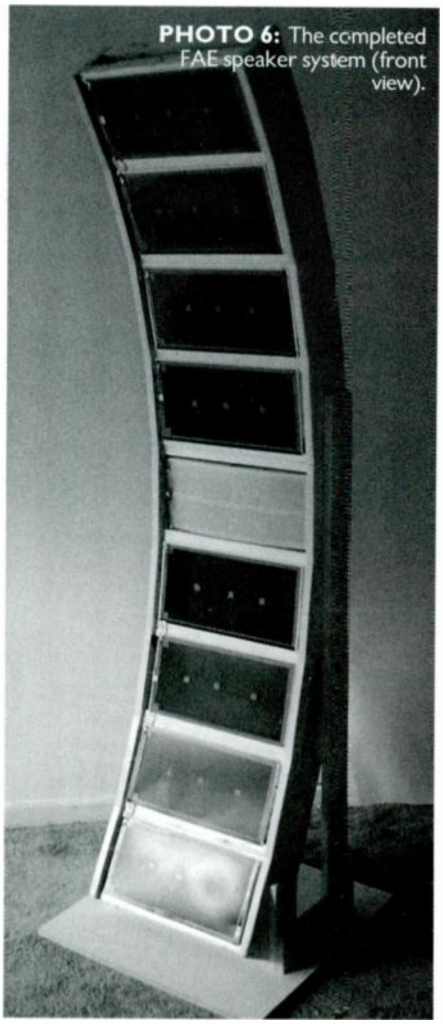

PHOTO 6: The completed FAE speaker system (front view).

FILLING THE CURVE

I used materials from David Lucas to assemble my ESL panels and followed the techniques (with some variations of my own) given in the directions supplied (though I was aware of most of these techniques from previous reading and discussions). The insulating spacer material, as mentioned earlier, was adhesive foam, which indeed makes for fast assembly. I won't describe in detail the construction method, because David Lucas considers the techniques proprietary.

I had some problems with the small panels twisting and bowing, which limited the amount of high-voltage bias I was able to use to under 1.5kV. This affects the speaker's overall sensitivity. I think that with larger panels that are normally made with the foam dielectric, this is not a problem, since the tensions would probably balance out over the larger area. But for these small panels, the more rigid construction of solid dielectric may work better. If I were to do the project over, I would probably try Roger Sander's technique using solid plastic dielectric instead.

PHOTO 7: Panels and wiring in place.

For the midrange stators, I used powder-coated perforated aluminum, precut at my request to 14.25″ × 7.125″ by the Lucas company. The Lucas method of making electrical contact to the graphite-treated diaphragm served well, but to connect to the stators, I simply used #4 nylon screws (with the heads mounted between the stators) to hold grounding lugs to the outside of the stator, providing a solid, easily connected contact for the stator connections. I put these screws at the corners and isolated that section of the cell with the foam insulating tape so that the diaphragm was not installed over the small areas where the screws were used.

The panels go together in a relatively short time (*Photo 6*), and none of the steps involved in Lucas's process are difficult. But you must keep your mind on what you are doing. If you miss a step, there is usually no going back and fixing it—you need to disassemble the cell (a messy affair involving solvents and sometimes gently applied heat to salvage the perforated aluminum) and start over.

More times than I care to admit, I experienced slightly late revelations ("Oops, forgot to pull the backing off an insulator," or "didn't put the contact on the diaphragm," or "forgot to put the nylon screw in the stator") and had to disassemble and start again. Making a few large cells would be a quick process, but making the 18 smaller ones required several days.

I used the supplied foam from Lucas for the insulators in the cells, as recommended. But I mounted the finished panels to the frame using horizontal strips of the less-expensive double-sided foam tape available from most office-supply stores (you can order rolls of this for best cost efficiency and because a moderate amount is used). This tape is also useful for tacking down the high-voltage wiring used to connect the panels to each other or to the crossover and bias supply. I used normal light-gauge stranded hookup wire here (there is very little current conducted in an ESL), and used the tape to space the wires from each other and to keep them from lying directly on the frame (*Photo 7*).

NO MOVING PARTS

TRASH ALL OF YOUR WOOFERS, MIDRANGES, AND TWEETERS - AND I'LL EVEN INCLUDE ALL DIPOLES AND PLANAR MAGNETICS. ANY SPEAKER, WITH EVEN A SINGLE MOVING PART, IS FROM THE DARK AGES. THE ENERTIAL MASS OF THESE DINOSAURS GENERATE SPURIOUS WAVE FORMS, DISTORTIONS, AND UNPREDICTABLE OPERATION.

CRAWL OUT OF THOSE PRIMITIVE SPEAKER BOXES YOU'VE BEEN FUSSING WITH AND TRY TO ENVISION THE IDEAL LOUDSPEAKER - A MASSLESS SYSTEM THAT WOULD IMPART ACOUSTICAL MOVEMENT TO THE AIR IN EXACT DUPLICATION OF ITS ELECTRICAL INPUT WITH NO MOVING PARTS.

WE INVITE YOU TO ESCAPE THE TWILIGHT ZONE OF PREHISTORIC SPEAKER TECHNOLOGY AND JOIN THE AGE OF ENLIGHTENMENT - IONIC AND PLASMA LOUDSPEAKERS! MOVE THE AIR WITH NO MOVING PARTS! HEAR MUSIC THE WAY IT SHOULD BE HEARD, NOT FROM AN OUTDATED MECHANICAL CONTRAPTION, BUT FROM THE PURE VIBRATION OF AIR. THE SAME VIBRATIONAL SOURCE EQUIVALENT TO MUSICAL INSTRUMENTS OR HUMAN VOCAL CORDS.

I KNOW IT SOUNDS IMPOSSIBLE AND I EXPECT A CROP OF "IFS" AND "BUTS" AS YOU ARE PROBABLY SAYING TO YOURSELF: "BUT HOW CAN SOUND OR MUSIC BE PRODUCED IF THERE ARE NO MECHANICAL PARTS VIBRATING THE AIR?"

A VALID QUESTION THAT IS ONLY COMPOUNDED BY THESE FACTS: 1. IONIC AND PLASMA SPEAKERS USE NO BOXES OR ENCLOSURES. 2. THEY USE VERY FEW PARTS. 3. THEY CAN BE BUILT ON YOUR KITCHEN TABLE. 4. THEY ARE VERY INEXPENSIVE TO BUILD. 5. MOST OF THE PARTS CAN BE OBTAINED FROM COMMON LOCAL STORES SUCH AS BUILDING SUPPLIES, RADIO SHACK AND/OR ELECTRONIC SURPLUS.

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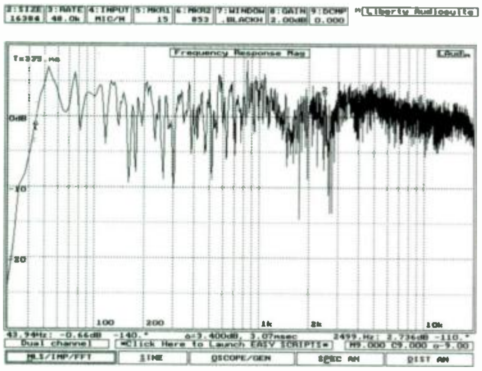

FIGURE 12: In-room frequency response over 335ms for FAE system.

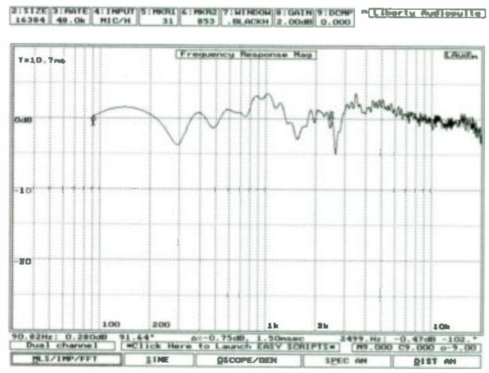

FIGURE 15: FAE frequency response, windowed to first 10ms.

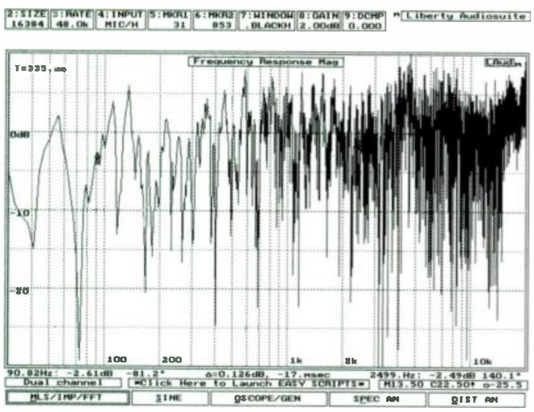

FIGURE 13: Same measurement on conventional three-way loudspeaker.

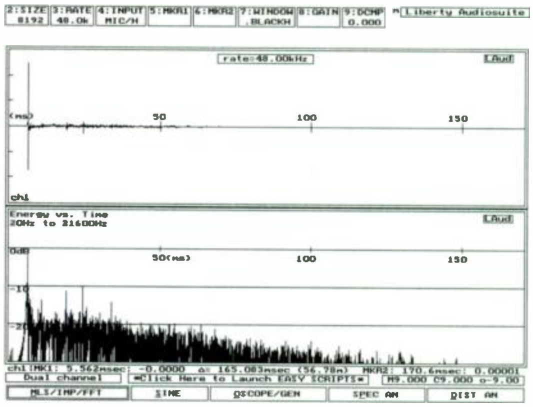

FIGURE 16: Time energy curve of FAE system from the listening position.

TWISTY TWEETER

I devised the method for making the curved tweeter panel myself. There may be a better or easier way to do it, but this technique worked rather well. I calculated the curvature I wanted and made a pattern on a piece of poster board, then traced it on the edges of four 15″ pieces of pine, two per tweeter. Then I used a belt sander to shape the pine to follow the traced contour.

I used these shaped sides to make a small frame to be mounted between the side panels in the center cell of the large ESL frames. The shaped pieces of pine formed a contour that would curve outwardly when mounted in the large frames, so the tweeter sound would spread laterally. I then mounted a piece of hardware-store light-gauge perforated aluminum onto the curved edges of the small frame, using office-supply foam, and applied the insulator foam tape to the outer surface of that mounted aluminum stator—and was careful to remember to add the contact screw and lug!

For the curved diaphragm, the lateral distance between insulator spacers should be reduced because the diaphragm film will tend to stretch toward a straight path in some regions between the spacers. That brings the film closer to the continuously curved stators and could limit high voltage or SPL handling in the tweeter. An extra continuous-strip spacer running left to right in the middle of the curved stator, along with smaller spacers distributed in the remaining space, seemed particularly helpful.

To install the diaphragm over the curved surface, I made another construction frame, this one with a flat surface (*Fig. 11*), to allow the tweeter frame to be inserted into it. Then, I treated the Mylar diaphragm with graphite and stretched it across the opening in the construction frame. Next, I exposed the adhesive on the insulator foam which was on the mounted stator, and pushed the curvature of the tweeter frame gently into the Mylar stretched over the construction frame.

When all of the foam tape had been pushed into diaphragm material, I cut around the periphery of the Mylar on the construction frame and applied the heat shrinking to the film now stretched across the insulator. The remaining part of the construction was similar to the process used for the midrange panels, except the operations were performed on the curved surface rather than a flat one.

When completed, the tweeter included the contoured frame on which it was made. I mounted this frame into the large frame using wood screws to the insides of the side pieces. This allowed the tweeter's position to be adjusted relative to that

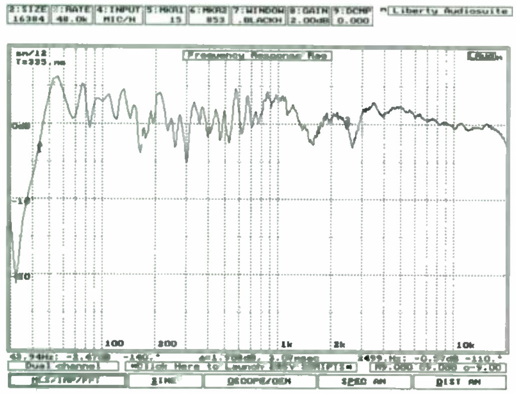

FIGURE 14: Same as *Fig. 12*, but with 12th-octave smoothing applied.

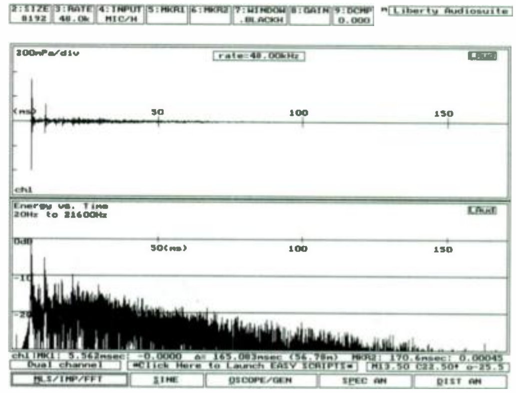

FIGURE 17: Time energy curve of conventional three-way system.

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.

Retail price: US$ 530.00 (delivered)

Delivery in US within 4-6 days.

Warranty 3 years, 30 days money back guarantee.

High sonic resolution Loudspeaker with room friendly performance

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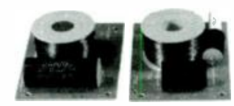
The step beyond the limits

RT1C Tweeter

F5 Bass-midrange



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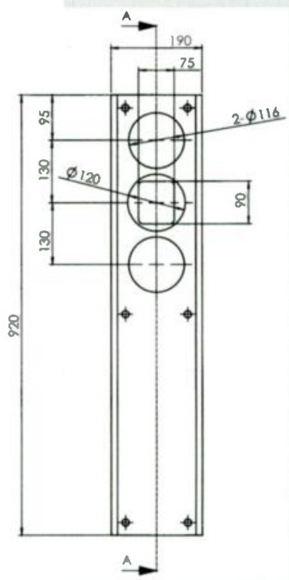

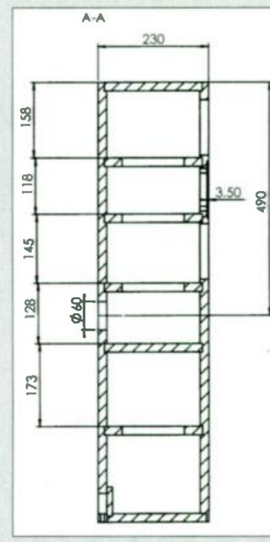

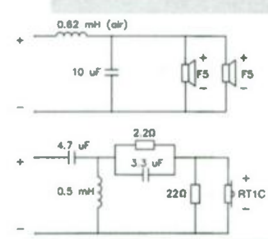

SPECIFICATIONS

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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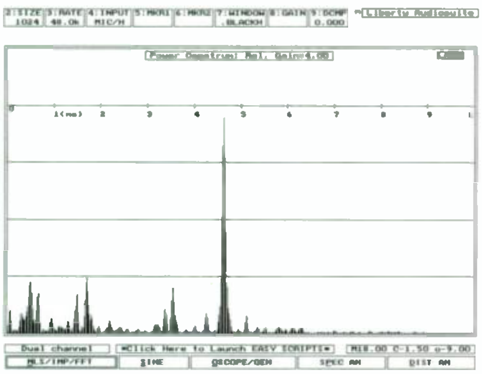

FIGURE 18: Power cepstrum to 10ms of FAE, from listening focus.

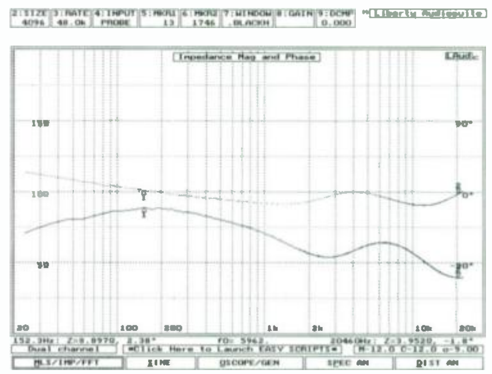

FIGURE 20: Impedance of combined tweeter and midrange sections.

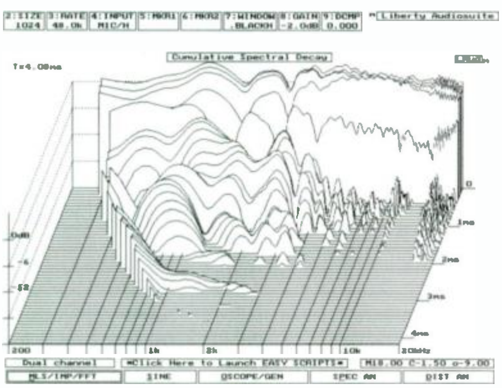

FIGURE 19: Waterfall plot of FAE for a window length of 4ms.

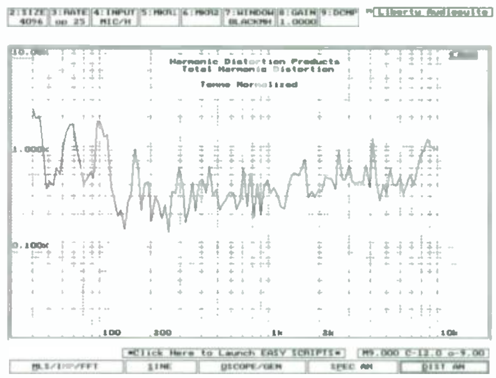

FIGURE 21: Harmonic distortion of FAE at 80dB SPL, with Temme normalization.

of the midrange panels during alignment, for best frequency and pulse shape response.

ALIGNMENT AND MEASUREMENTS

Aligning Focused Array arcs is a time-consuming process. First you must define the listening position and make sure that it allows for proper placement of the arcs and of the woofer units. A tape measure is recommended for this. Just connecting all the cables, connectors, amplifiers, and power supplies—all in proper polarity—can take over an hour.

Then, you must adjust the toe-in of the arrays, as well as the forward/backward tilt and the overall distance from the listening ear position for each speaker. You can make a pretty good approximation of the toe-in and tilt of the speakers by directing a flashlight, held next to your ear, toward the panels and looking for the reflection in the center of each of the panels. But this involves quite a bit of sitting down and getting up unless you have help moving the arrays. You can mark and adjust the focus distance with an appropriate length of dowel rod.

Some kind of measurement gear is recommended to align this kind of system, as you will likely need to adjust the relative levels of the drivers to make up for variances in panel sensitivities due to construction methods and design choices. A measurement system may also be required to configure correctly the relative polarity of all the drivers, and aid in fine-tuning the tilt and toe-in. These adjustments can be made by changing transformer taps in the ESL section or by adjusting the sections in the electronic crossover.

Figure 12 shows the measured in-room response, including about 300ms of acquisition time (far beyond the typical quasi-anechoic limit of around 5ms) of one of the FAE arcs and its woofer tower. I measured at the defined listening position 5′ away, with the microphone over the listening chair (sofa) that would be used, because I was interested in approaching the actual in-use response rather than in just making a "marketing" plot. As you can see, room reflections still manage to corrupt the response, even in the focused array system. But compare this with the same measurement made on a three-way cone system under the same conditions (*Fig. 13*).

Figure 14 shows the same data as given in *Fig. 12*, but with 12th-octave smoothing applied to better approximate the perceived response and to reveal the underlying data. *Figure 15* shows the same data again, but with the time window limited to only 10ms (and no smoothing).

The full-band time-energy curve of the FAE system, measured in-room and at the listening position, is shown in *Fig. 16*. The energy drops quickly by about 20dB, and, amid several distinct reflections, the reverberation of the room (and the rest of the house) becomes evident and then decays. For comparison, the time-energy curve of the conventional three-way is shown in *Fig. 17*.

The power cepstrum (reflection) plot of the FAE, over an analysis quefrency of 10ms and with gain of 4, is shown in *Fig. 18*. The only predominant feature is a single reflection at about 4.7ms; this same reflection can also be seen in the time energy curve of *Fig. 16* (the first distinct line after the main impulse). A similar reflection is also evident (though much stronger)

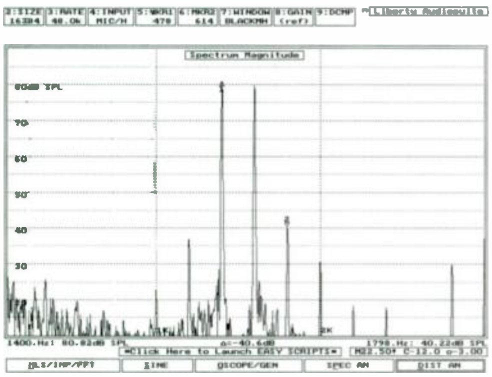

FIGURE 22: Two-tone intermodulation spectrum in midrange.

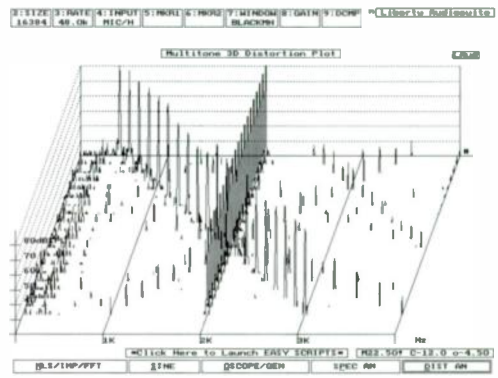

FIGURE 23: 3-D intermodulation stepped frequency plot in midrange.

for the simple three-way system's data (*Fig. 17*). The reflection interval of 4.7ms corresponds to a sound-wave travel difference of 5.3′ (i.e., the reflection path is 5.3′ longer than the direct wave's path). This appears to be caused by my computer's cabinet, which I have set on a worktable, level with and several feet to the left of the microphone position, for performing the measurements. Other than this peak, the cepstrum result is quite uncluttered.

GOOD BEHAVIOR

A cumulative spectral decay (waterfall) plot made from the first 4ms of the FAE's impulse response is shown in *Fig. 19*. The electrostats are well behaved and show little in the way of resonance problems, though the response irregularities and decay discontinuities around 3kHz suggest a need for further crossover work.

The impedance curve of the electrostatic tweeter/midrange section is shown in *Fig. 20*. This curve is for the circuit using separate transformers and external dividing components. The curve is quite mild and does not present a difficult amplifier load, in contrast to the severe impedance dip that results if one of the large stepup transformers is pushed to a high turns ratio and inflicts its load at the upper end of the audio band.

Total harmonic distortion (*Fig. 21*) was measured at a level of 80dB SPL (without gating) and normalized using Temme's method (by which each harmonic is inverse-weighted according to the response at its frequency, relative to the response for the fundamental frequency, to minimize the effect of frequency response on the distortion measurement). In a measurement of individual distortion components, second harmonic predominated below about 100Hz, but was approximately equal to third harmonic at higher frequencies.

Figure 22 shows an intermodulation spectrum for two tones at 80dB SPL and spaced 200Hz apart, centered at 1.5kHz (the range covered by the midrange panels). The third-order product levels are higher than I'd like, and may be a result of the limited bias voltage used for the ESL panels. *Figure 23* is a similar measurement in which one tone is held at 2kHz and the other is stepped from 500Hz to 3kHz, showing the intermodula-

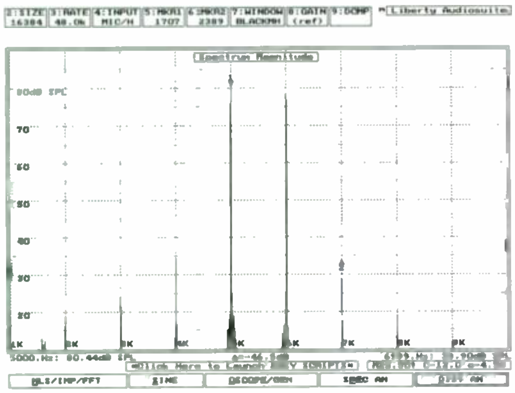

FIGURE 24: Two-tone intermodulation spectrum from tweeter.

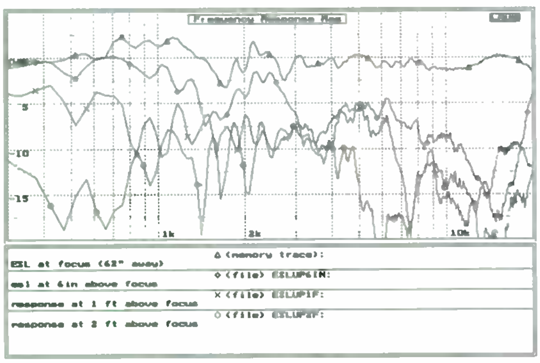

FIGURE 25: Response variation with vertical position.

tion product patterns for the top 60dB of output range. A plot similar to *Fig. 22*, but for the tweeter range, is shown in *Fig. 24*, showing good performance from the small 70.7V transformer. Coherent averaging was used in these three acoustic measurements to minimize effects of room noise.

The vertical directivity effect of the FAE curved midrange and tweeter section is shown in *Fig. 25*, in which the response curve for the FAE (single transformer crossover version) is shown for four microphone placements. *Figure 26* shows you the midrange response effect of moving the microphone toward the array over a two-foot range, demonstrating the unusual result that the level *drops* in this range as the microphone moves closer to the speakers (farther from the focus).

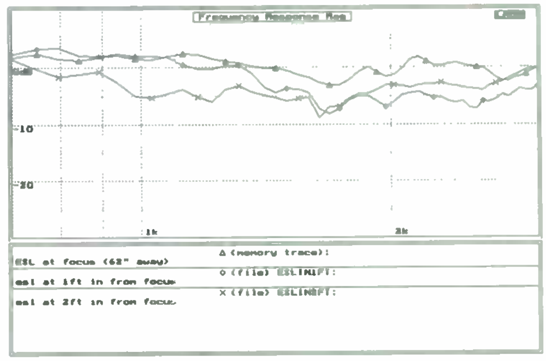

FIGURE 26: Response and level variation as array is approached from focus.

In the final installment of this series, I will discuss my listening tests and describe the sound of this focused array electrostatic system.

SOURCE

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DESIGNING THE DIPOLE MONSTER, PART 2

By Timothy E. Sandrik

Part 1 (SB 5/98) described the experiences of the author and his friends in designing, building, and exhibiting the Dipole Monster for the Western Michigan AES Loudspeaker Design Challenge in May 1997.

Like most designers, our instincts led us to begin by picturing the best box for our speaker. In the past, members of the team have built close to 100 sealed, ported, and bandpass enclosures seeking the "right" kind of bass. With each of these, the room proved to be one of the most influential factors affecting the low-frequency sound quality.

Room resonance, or standing waves, cause strong coloration and masking. In other words, if you compare the input signal to the speaker (delayed to compensate for the speed of sound, of course) with the signal at the listener's ear, they are likely to differ in amplitude, frequency, and phase. You can see this in the waterfall, or cumulative spectral-decay plot, of a loudspeaker measured at the listening position in a typical room.

This brings us to recording and playback theory, on which we're far from experts, but our instincts tell us that for many recordings, the electrical signal at the input terminals of the speaker should be translated into an acoustic signal at the ear. We considered reflections and resonances as noise.

SYSTEM ACCURACY

You can judge the accuracy of systems designed to be linear, which include amplifiers, preamplifiers, and loudspeakers, by these criteria: distortion, bandwidth, frequency response, phase response, and signal-to-noise ratio.

Amplifiers, for example, are expected to achieve their performance specifications under typical operating conditions. On the other hand, a loudspeaker is often considered adequate if it measures well in an anechoic chamber, which is not a typical operating environment. So, what if a loudspeaker, in its typical environment, could deliver a signal to your ears with less noise from reflections and resonances?

Siegfried Linkwitz of Audio Artistry believes that increasing the ratio of direct (signal) to reflected (noise) sound with a carefully designed hybrid dipole results in a more accurate signal for the listener. Shannon Dickson of *Stereophile* seems to agree in describing his listening impression of Audio Artistry's *Dvorak* (Vol. 19, No. 4, April 1996). We also agreed, and, given the freedoms and sponsorship for this project, we were able to pursue a hybrid dipole design with few compromises.

ROOM INTERACTION

Many designers assume that a dipole interacts strongly with a typical room. While it is true that the rear radiation matches the front radiation in amplitude, the total energy the speaker delivers is about two-thirds less than that of a similarly proportioned boxed speaker. It is only when the rear radiation reaches the listener as a strong early reflection that a dipole is troublesome.

Keep in mind that most speakers are increasingly omnidirectional as the room

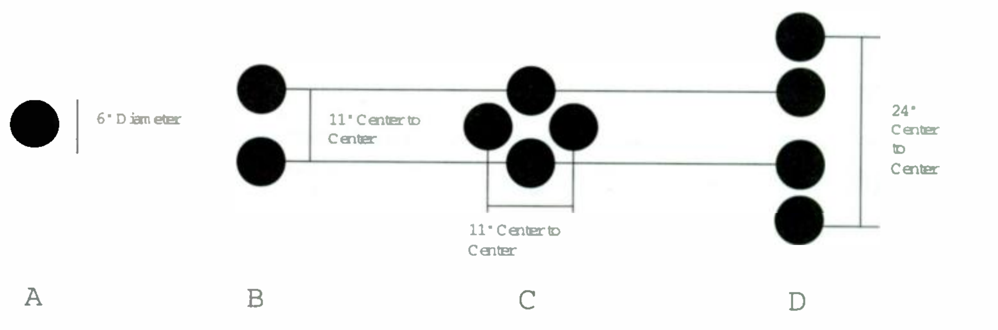


FIGURE 2: 6" driver configurations.

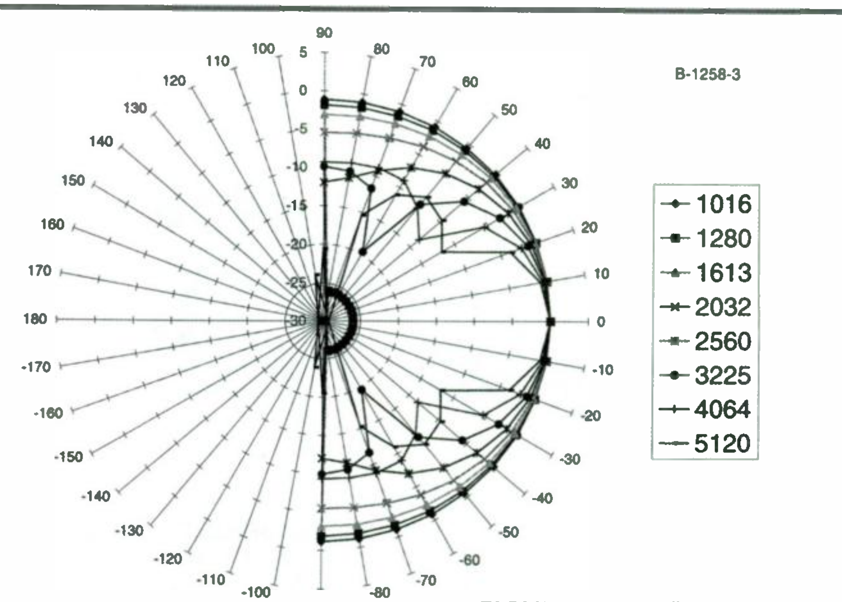


FIGURE 3: Single 6" driver (*Fig. 2A*).

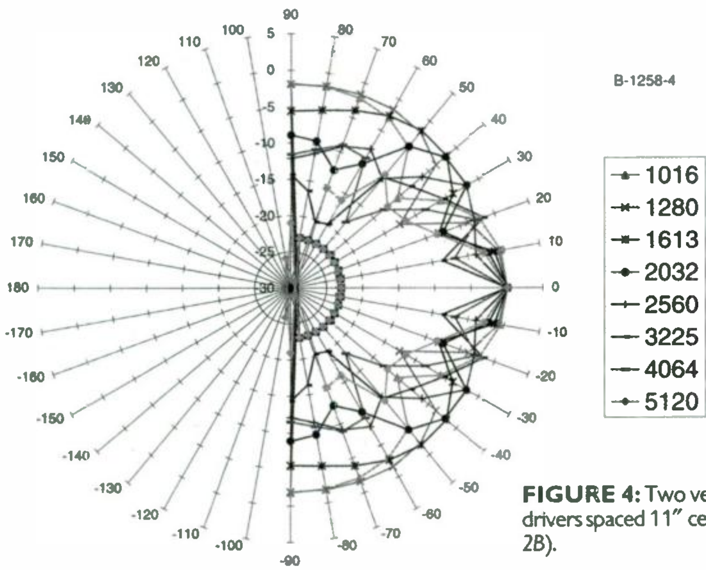


FIGURE 4: Two vertically aligned 6" drivers spaced 11" center-to center (*Fig 2B*).

becomes increasingly reverberant (as frequency decreases); hence they interact very strongly with a room. Below 500Hz, most boxed speakers will have front and rear radiation that are nearly matched, as well as matched radiation to the side.

Research has shown that noise or tones can mask a signal that extends to higher frequencies, or limit its perceivable dynamic range. You can draw the conclusion that excess reverberation at low frequencies can limit the perceivable dynamic range in music reproduction at all frequencies. Measuring the reverberant response of typical loudspeakers in most rooms indicates an obvious excess of low-frequency energy.

If you agree with our conclusions and think it is important to maximize the ratio of direct-to-reflected sound, then you might well use a dipole design, which requires smaller amounts of more localized room treatment to achieve the desired ratio of direct-to-reflected sound.

THE MTM ARRAY

We began the design of the hybrid-dipole midrange and treble portion of the speaker by examining the focused main lobe of the MTM array. As an aid in selecting drivers and optimizing the array, we developed a model that allowed us to explore filtered arrays of drivers. Early in the process we began to consider the question "If narrow vertical radiation is desirable, why not also have narrow horizontal radiation?" Bill Dudleston of Legacy Audio was able to answer this question with a demo of his top-of-the-line Whisper.

To summarize our impressions, a narrow horizontal pattern—combined with the proper toe-in—places off-axis listeners in a position so the nearer loudspeaker is more attenuated than the farther one. This stabilizes the center image for the off-axis listener.

Modeling arrays with midranges on all four sides of the tweeter yields interesting results. The radiation pattern is focused horizontally and vertically, and has less off-axis vertical lobing than an MTM. To illustrate how the vertical radiation pattern can vary with different driver arrays, the arrangements in *Fig. 2* are simulated in an infinite baffle *(Figs. 3–6)*.

The patterns are normalized to demonstrate radiation pattern independent of driver response. You can see that the single 6″ (*Fig. 3*) yields the smoothest radiation pattern over the typical operating region of 6″ drivers, while the MTM+ (*Fig. 5*) offers a good blend of smoothness and focused radiation. Given that our design must be a three-way maximum, the midrange drivers must share a crossover tap, so it was important to find an unfiltered array with focused radiation and minimal lobing inherent in the arrangement. Starting with the MTM+ array was an obvious choice.

ACHIEVING CONSISTENCY

We considered consistency (with frequency) of the radiation pattern as important as being narrow. Most designs achieve consistency at the expense of focus. We wished more of each. With greater consistency, we could expect better off-axis frequency and power response.

Consistency occurs when the radiator remains in constant proportion to the wavelength of sound. In other words, the driver needs to shrink with increasing frequency. While this is the case in almost every speaker, it is apparent that most speakers have sharp transitions between radiator sizes. In our design, we made smooth transitions between radiator sizes a priority, and thus we have interesting treatments and crossover responses.

The MTM+ array has a narrow but inconsistent radiation pattern. *Figure 7* is the measured radiation pattern of an MTM+ array of Audiom 7k drivers, 11″ center-to-center horizontally and vertically, in a 468 in^2 dipole baffle, with each driver driven equally (frontal radiation only). You can see that it is very inconsistent and quite narrow above 3kHz.

To widen the pattern at high frequencies, we sought ways to make it smaller at high frequencies without making the crossover a four-way. Our first attempt involved covering half of each Audiom 7k driver with felt to filter high frequencies from the outsides of the cone. (Felt stretched in front of a driver will slightly attenuate high frequencies.)

Figure 8 demonstrates the result. A comparison with *Fig. 7* shows that this worked, but not that well. There followed a long search for the right material in the right amount to "tune" the radiation pattern.

WHITE FOAM

Further attempts included more felt, various cushion foams, acoustic foams, and cloths covering all or part of the drivers in the array. The best results for the horizontal pattern were achieved by covering the two side drivers with 4″ of white foam from a fabric store—the type of foam used for making seat cushions.

Figure 9 is radiation-pattern data taken on center axis of the array horizontally. Again, the plot shows only frontal radiation. In it you can see that the pattern has become very consistent and wider at high frequencies. These results were almost unbelievable, so we confirmed the data more than once. On-axis, the foam attenuated the output of the two side drivers as shown in *Fig. 10*. Some may accuse us of cheating by using acoustic filters; however, the foam's attenuation differs with the angle (exactly how is not yet part of our data).

If you have an intuitive understanding of radiation pattern, you might guess that the result of this great horizontal arrangement is a compromised vertical pattern. The awkwardness of the speaker prevents a thorough documentation of the vertical pattern at this time, though we were able to take enough data to realize the vertical pattern suffered as predicted.

We applied some of the same trial-and-

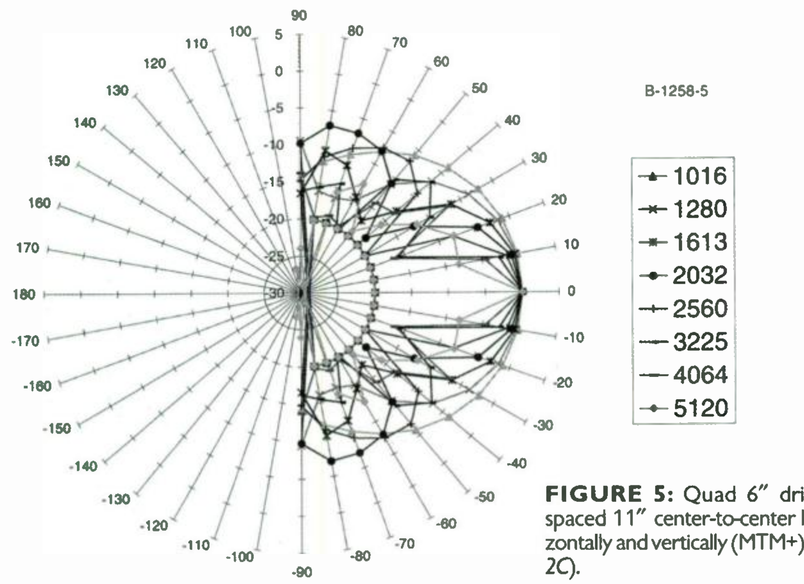


FIGURE 5: Quad 6″ drivers spaced 11″ center-to-center horizontally and vertically (MTM+) (*Fig. 2C*).

error acoustic filtering with little improvement to the vertical pattern. We then decided to eliminate the lower vertically aligned midrange, and to wire the side drivers in series with each other, and in parallel to the top midrange in the array. This significantly improved the vertical pattern with no sacrifice of the horizontal pattern.

Since documentation is not available, a simulation will illustrate this effect. For the simulation, we introduced the following simplifications:

1. The effect of the acoustic filter is approximated as a first-order filter applied at 1.5kHz to the side drivers.
2. The side drivers each receive 0.7 times the input signal of the top driver in the three-driver array (approximates series-parallel wiring).
3. The fact that the drivers are in a dipole is ignored. (Relative differences in the "main" lobe will be similar—besides, a real-world dipole is tough to model.)

FOUR-DRIVER ARRAY SIMULATION

Figure 11 is a simulation of the four-driver midrange array, and *Fig. 12* is a simulation of the three-driver array. You can see that the pattern is wider and more consistent off-axis with the three-driver array. Considering that at a distance of more than 8′, the typical vertical listening window is less than 18°, the midrange array is capable of covering this with good response.

We chose the Raven R1 ribbon primarily for its excellent transient performance, but the fact that it is a narrower horizontal radiator allows more bandwidth off-axis. To keep the pattern narrow around the crossover, we built felt horn/absorbers into the midrange acoustic filters. These rest against the faceplate of the tweeter and "flare" into the acoustic filters in front of the midranges (*Fig. 13*). The absorbers provide some loading to the tweeter, raising its efficiency around crossover, and they also provide some directivity control. We constructed and tested many clay horns, but this simple felt arrangement is the most consistent and involves no faceplate reconstruction.

Two Focal 12726S woofers per speaker first handled the low frequencies. Creative construction techniques and optimization of materials made it possible to use four of these drivers per speaker. Placing them all near the ground gained nearly a 6dB boost at low frequencies. Each driver had a published linear excursion of 9mm, and with the total surface area, these were capable of about 103dB of output at 30Hz (per speaker) within their published 9mm excursion.

The physical size of the "driver," made up by the group of 12″ woofers, determined the radiation pattern, and there was little we could do at such long wavelengths to control it. An absorber or barrier intended to control radiation must be comparable to the wavelength of sound to be controlled. At the higher frequencies (above 200Hz) the "kink" in the cabinet, which helps it stand, acts like a phase plug, widening the pattern in front of the baffle. At low frequencies, the dipole baffle is the sole controlling factor of the radiation pattern.

BRINGING IT ALL TOGETHER

Bringing the bass cluster, midrange array, and tweeter together mainly involved comparing individual radiation patterns and choosing crossover points to make a system with the most consistency. In the region from 5kHz–8kHz, the midrange array and the Raven R1 (with the various acoustic filters and pattern controls) match very closely in pattern. We chose the crossover to be around 6kHz. The drivers are time-adjusted, and the filters result in a minimum phase response.

The filter on the Raven R1 is a fourth-order, minimum-phase, high-order filter, which grows steeper to fourth-order around 2kHz for added protection. The filter on the midrange is the most complex, since it must compensate for three 96dB/W drivers operating below 2kHz, one driver above 2kHz, and dipole baffle effects.

The Audiom 7k array begins to radiate widely below 500Hz, so the 12V726S cluster is brought up to almost 500Hz to help keep the pattern narrow. Fortunately, the 12″ woofers move very little at these frequencies, and their first noticeable breakup is around 1kHz.

Below 150Hz, the dipole baffle attenuates the output of the system. The low f_3 was determined by choosing the system efficiency and then cutting the output of the 12″ drivers to compensate for 6dB/octave dipole baffle rolloff. Essentially, four 12″ woofers in our baffle

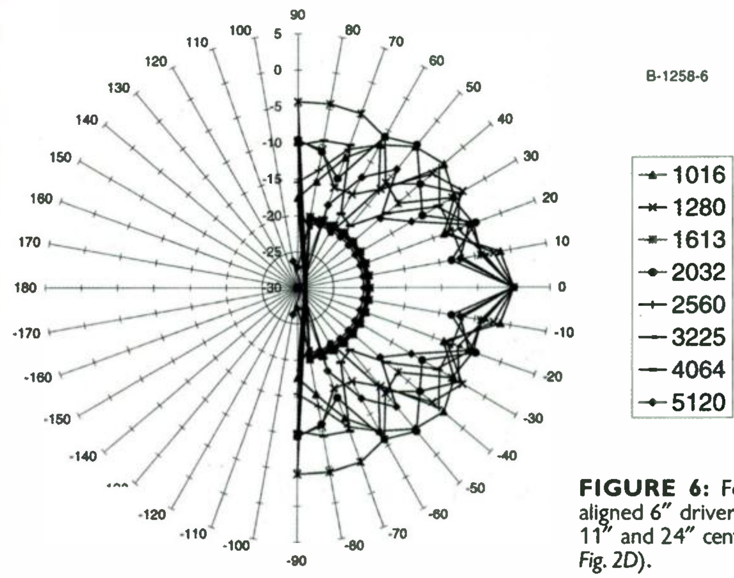


FIGURE 6: Four vertically aligned 6″ drivers (line source 11″ and 24″ center-to-center—*Fig. 2D*).

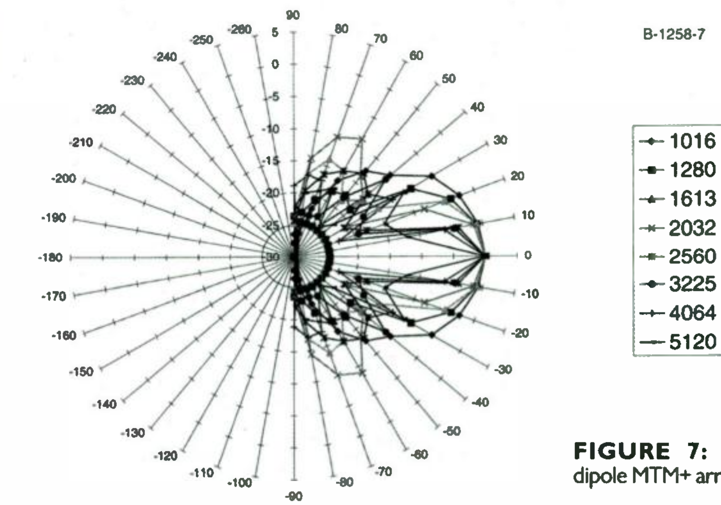


FIGURE 7: Audiom 7k dipole MTM+ array.

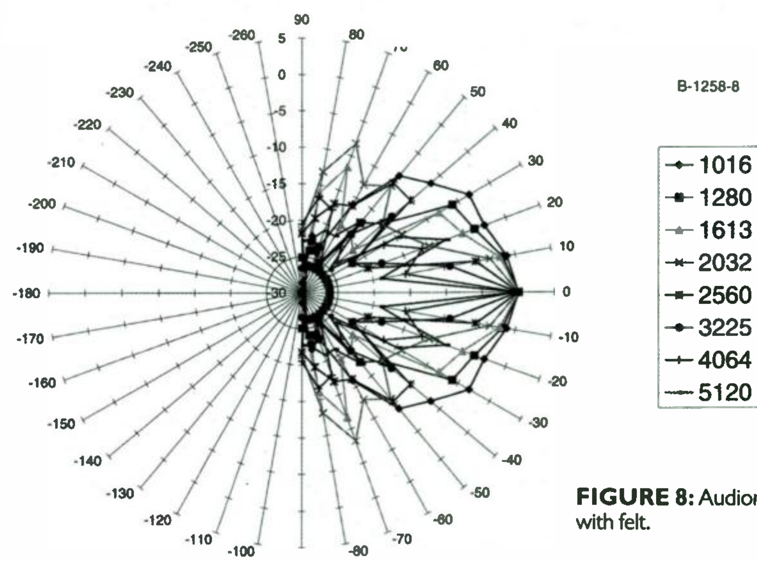


FIGURE 8: Audiom 7k array with felt.

have a voltage sensitivity (2.83V) at 30Hz of around 92dB.

Above 30Hz, the drivers are attenuated with a very large inductor to maintain the 92dB efficiency to 150Hz, where the attenuation is stopped by the addition of the right amount of parallel resistance to the inductor. The goal is to have a filter slope that exactly opposes the rolloff of the system. *Figure 14* demonstrates this graphically.

MEASUREMENT TECHNIQUES

Comprehensive measurements of a loudspeaker are always difficult. The best measurements are achieved in a good anechoic chamber. Although we have access to a large chamber, it is not an audio chamber, so its usefulness extends only to about 200Hz.

My favorite technique is to measure speakers in an open warehouse or parking lot, with a microphone on the ground 10′–20′ from the speaker. This eliminates reflections that alter the response in the midrange, in exchange for some inaccuracy above 10kHz. The complete radiation pattern was measured with LAUD in the parking lot at DCM Loudspeakers in Ann Arbor one cool day this summer after truck traffic had diminished.

We took data every 10° horizontally around the speaker (on the vertical axis of the tweeter). Since the speaker is horizontally symmetrical, we averaged the data from the left half of the speaker (190°-350°) with the corresponding data from the right half (10°-170°) to reduce error.

The radiation pattern is described by two plots, both of which are normalized to describe radiation pattern independent of system frequency response. *Figure 15* is the third-octave radiation pattern from 640Hz–16.255kHz. *Figure 16* is an octave-band radiation pattern, which covers more bandwidth.

You can see that the pattern becomes pretty wide below 640Hz. We would have

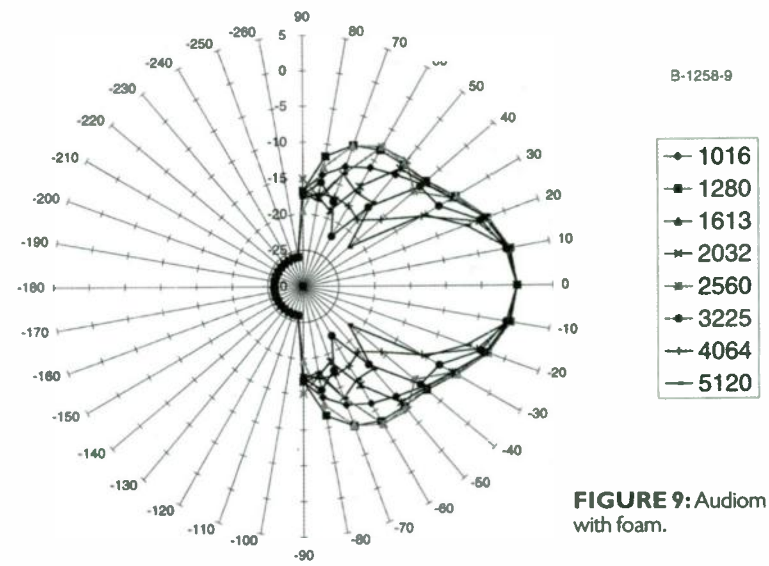


FIGURE 9: Audiom 7k with foam.

liked to narrow the pattern there, but this would have involved a larger baffle, without a kink, which would not have been possible without giving up efficiency, bandwidth, or dynamic range. In other words, we would have needed to eliminate a 12″ woofer to afford the weight of the additional structure. You will also notice—to the side of the baffle—that there aren't the usual deep nulls characteristic of a dipole. The strangely shaped baffles and acoustic filtering, as well as the outdoor noise level, all had a part in this.

Figure 17 is the on-axis frequency response of the speaker. The plot is composed of three measurements. Above 10kHz, the data is from the anechoic chamber at Michigan Technological University (MTU). Below 200Hz, data is from a sine-swept ground-plane measurement at MTU. The remaining data is from the ground-plane measurement made at DCM. *Figure 18* is the impulse response of the system as mea-

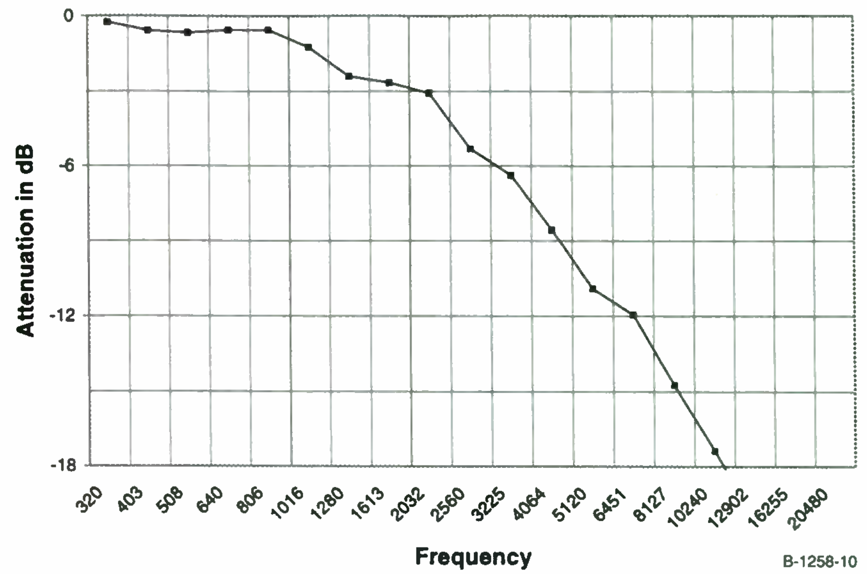


FIGURE 10: On-axis attenuation of foam on Audiom 7k.

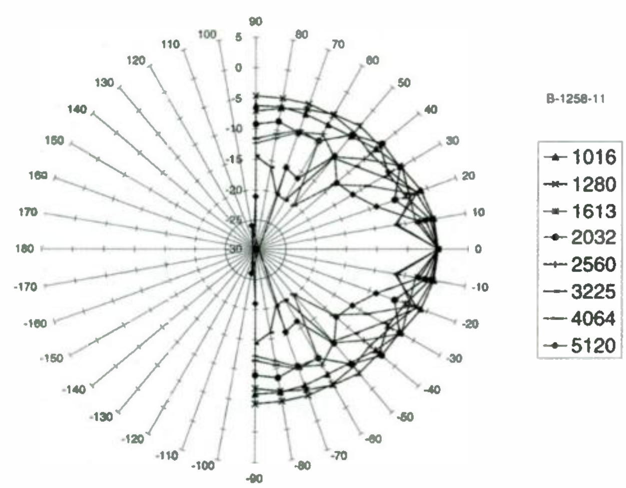


FIGURE 11: Simulated vertical four-driver MTM+.

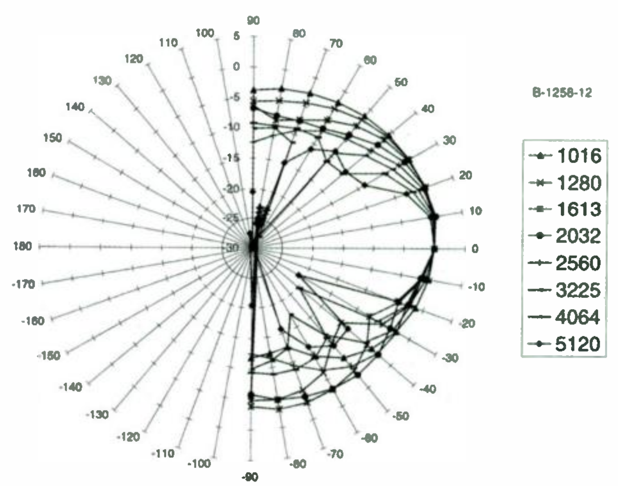


FIGURE 12: Simulated vertical three-driver modified MTM+.

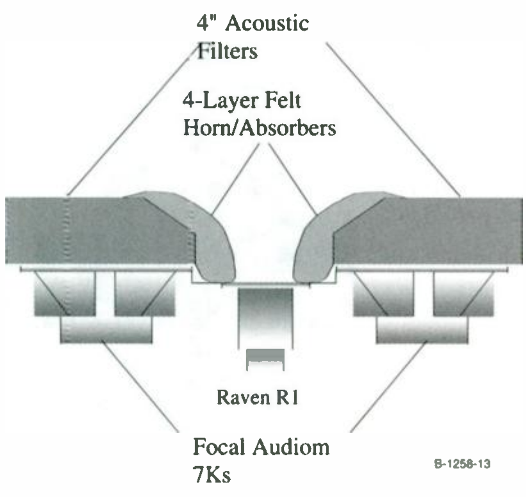


FIGURE 13: Diagram of foam and felt acoustic filters.

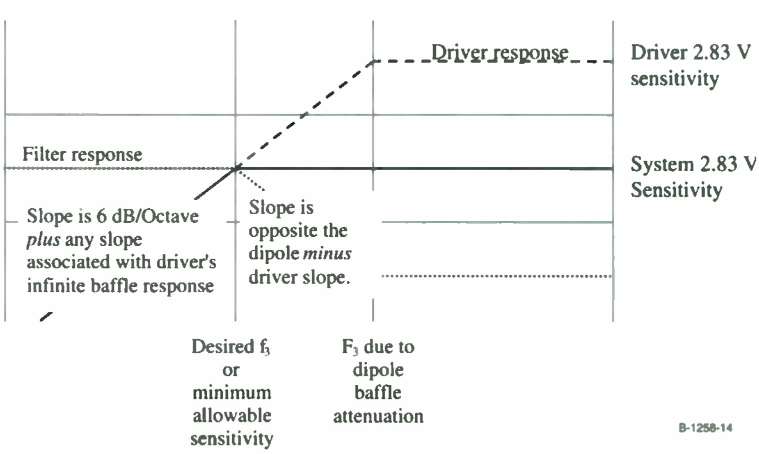


FIGURE 14: Dipole baffle compensation.

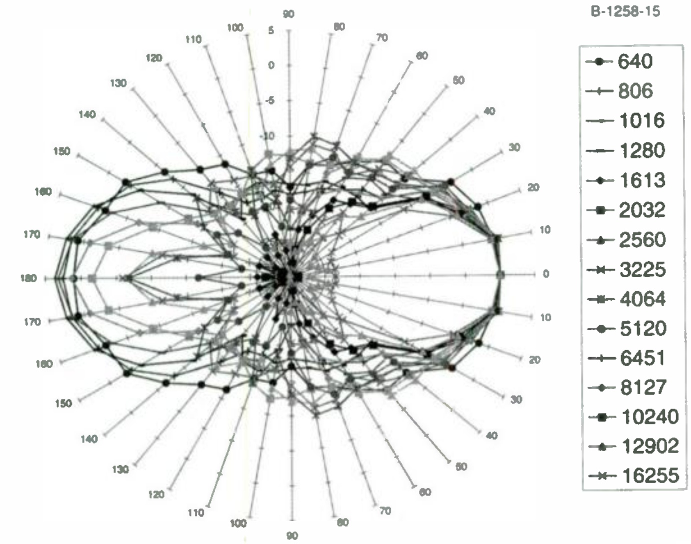


FIGURE 15: One-third octave radiation pattern.

sured outdoors. From the top plot of this figure, you can see that the presence of reflecting surfaces is minimal. The lower impulse is the same, only gated closely to show detail in the directly radiated sound.

WONDERFUL SOUND

Listening to this speaker is amazing! We're too young to consider our ears golden, and our hours logged listening to the speakers are still few, but we have heard few systems that make an impression like this one. Whether it is the limited room interaction, the awesome transient ability, or the sheer dynamic range capability, they sound great. Off-axis, the image loses a little detail, but remains unusually centered.

Originally, we had planned to provide detailed plans so you could build your own set, but frankly, we wouldn't recommend your doing so. It could be built far more simply, and something this good should be customized for the owner—we wanted ours to look like a race car. Following is a list of considerations that are keys to matching or exceeding the performance of our system:

- Space the midranges 5.5″ above and to each side of the tweeter's center.
- Recess the rear of the tweeter face plate ½″ behind the rear of the woofer frame for time adjustment.
- Find foam 3″–4″ thick to filter the side midranges with the same transfer response as *Fig. 10*. To find the transfer function of the foam, place the driver in a large baffle, measure its response without the foam, then repeat the measurement with the foam. Most measurement packages will find the transfer function between two curves. Mathematically, you'd subtract the curve without the foam from the curve with it if the curves are in dB SPL. If the curves are expressed as pressures

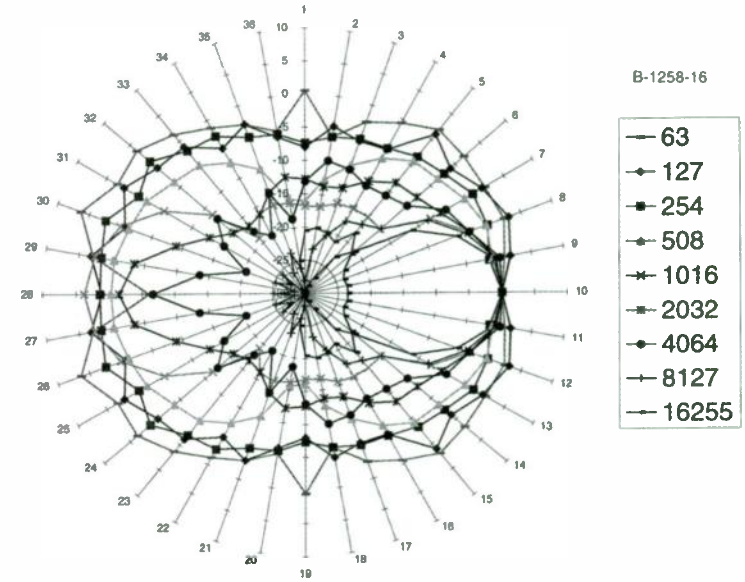


FIGURE 16: Octave-band radiation pattern.

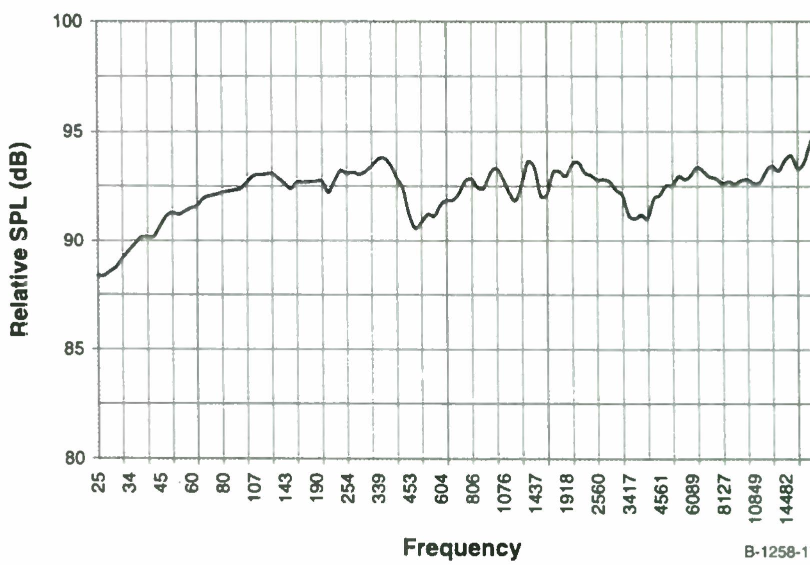


FIGURE 17: On-axis frequency response.

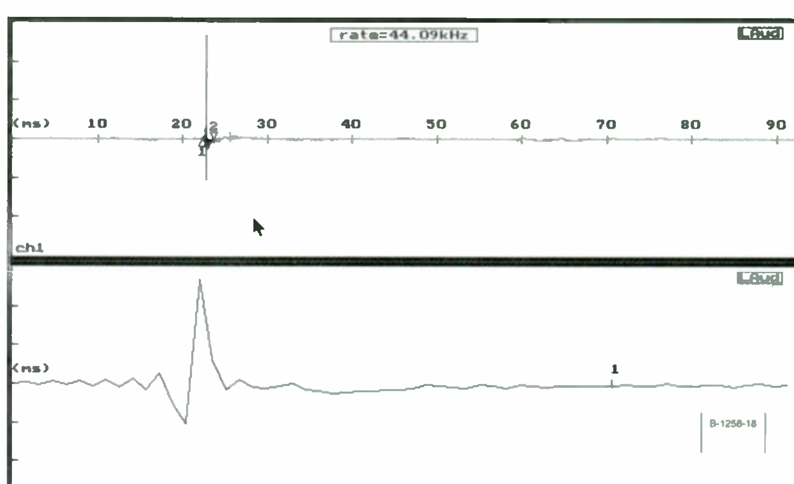


FIGURE 18: Impulse response.

TABLE 1

CROSSOVER PARTS LIST

R1, R2, R7	6Ω
R3	30Ω
R4, R5	15Ω
R6	2.5Ω
R8	22Ω
C1	3.9μF
C2, C5	30μF
C3	10μF
C4	6μF
C6	4.7μF
C7	300μF
L1	1.5mH
L2	3.0mH
L3	1.0mH
L4	0.5mH
L5	0.25mH
L6	18.0mH

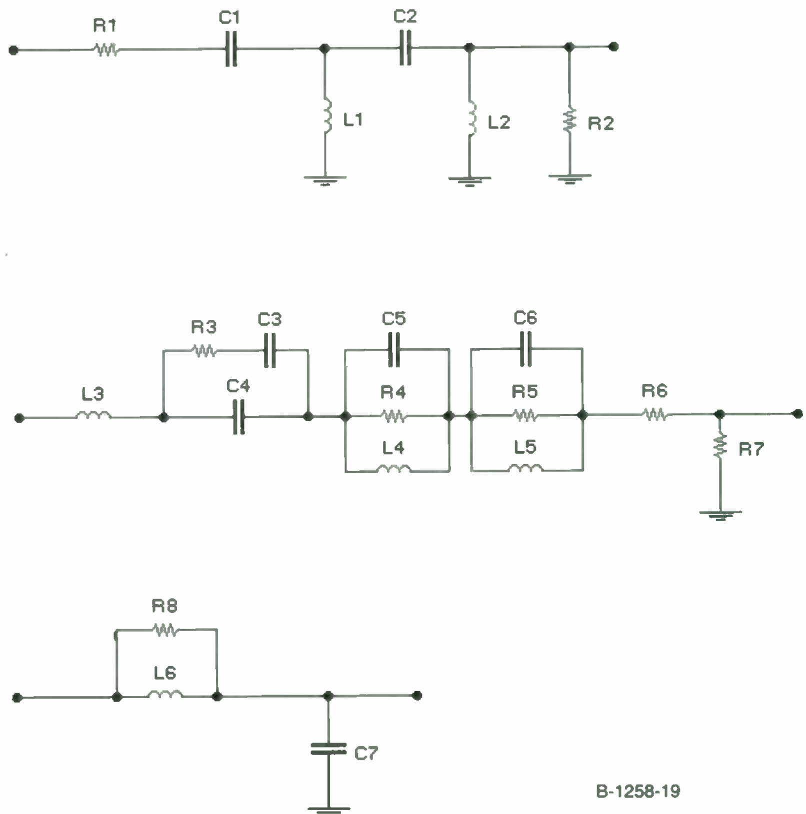


FIGURE 19: Crossover schematic.

(not dB), divide the curve with the foam by the curve without.

- Form the tweeter horn/absorber with four layers of felt, carefully attach it to the tweeter face so it is not blocking the tweeter, and secure the other side to the foam absorbers. You'll know if it's right if your response is flat.

- Use the same crossover. *Figure 19* is the crossover schematic, and *Table 1* is the parts list.

- Be sure to use an air-cored inductor for the 18mH in the woofer circuit. It would probably be more cost effective for most designers to perform dipole baffle compensation actively.

- The upper baffle is about 468 in^2, and the lower baffle is around 1300–1400 in^2.

The spacing we used between the center of the mid-tweeter array and the woofer cluster was based on the unused four-driver midrange array. The upper array can, and in most cases should, be moved closer, vertically, to the woofers. Also, an unkinked lower baffle will help to narrow the radiation pattern where the midranges cross over to the woofers—and you don't need to use two separate baffles.

A bit of final advice: take time to do it right. For us, this speaker was a learning experience, and a well-financed one. Skipping a step here and there will result in a very complicated and expensive "pretty good" speaker. Before you get started, though, get in touch with the student AES section at MTU and see what they're up to this year.

In Part 3, I'll deal with enclosure and vibration matters.

Kit Review

ARIA 5/RAVEN R-1

By Craig Stark

Aria 5/Raven R-1, Zalytron, 469 Jericho Tpke., Mineola, NY 11501, (516) 747-3515, FAX (516) 294-1943. Price: $1170 with cabinets, $770 without.

The Aria 5/Raven R-1 kit is a modification of the Aria 5 design from Focal that has been on the market for several years. Like the original Aria 5, the Aria 5/Raven was designed by Joseph D'Appolito and uses a pair of 5¼″ Focal 5k013L woofers in the MTM alignment that bears his name. In place of the Focal T-90 tweeter is a Raven R-1 ribbon driver. This most recent version of the Aria 5 is an extraordinarily smooth, detailed, and accurate speaker that has been a pleasure to review.

THE KIT

The kit as it arrived from Zalytron consists of a pair of cabinets, four Focal 5¼″ mid-woofers, two Raven R-1 tweeters, eight pieces of hookup wire, two binding posts, a sheet of Black-Hole™ self-adhesive multi-layer acoustic foam (an optional upgrade present in the review sample), a bag of assorted parts, and a schematic. While those who have built speakers before may require no further assistance, the neophyte would certainly benefit from some additional documentation.

Since novices are one of the primary audiences for a kit, this one should include, at a minimum, a parts list, a basic description of the system, and a suggested crossover layout. A printed-circuit board would be an excellent addition, as well. For those uncomfortable assembling the crossover themselves, however, Zalytron can provide you with an assembled crossover for a nominal charge.

To say that the cabinets are solid is an understatement. Though fairly small (20.5″ × 10″ × 11.5″), each cabinet tips the scales at over 40 lb without the drivers installed. The heft is the result of the 1½″ MDF (two layers of ¾″) used throughout the cabinet. Box vibrations, therefore, are not much of a concern with these cabinets, which came unfinished with an oak veneer on all but the back and bottom sides (teak and rosewood are also available).

SYSTEM SPECS

Drivers: Two Focal 5k013L 5¼″, one Raven R-1

Crossover: Fourth-order Linkwitz-Riley (acoustic) at 2250Hz, second-order electrical

Frequency response: 100Hz–20kHz ±1.5dB, –3dB at 45Hz

Sensitivity: 91.5dB

PHOTO 1: View of the Aria 5/Raven kit showing the cabinet, drivers, and assembled crossover using the supplied cable.

Two ports are located on the back of each box along with a standard five-way binding post. Grilles are black and ¼″ deep and held in place with standard fasteners. The front baffle is rounded to reduce diffraction effects, though the effects of the 1″ quarter-rounding are probably more cosmetic than acoustic. Overall fit-and-finish of the cabinets is quite good, especially given that they are hand-made to order by the folks at Zalytron (*Photo 1*).

In keeping with the sonic excellence of the drivers and cabinets, the parts quality is excellent. All coils are air-core, all caps polypropylene, and all resistors appear to be noninductive. The two resistor values included (5Ω and 7.5Ω) allow for two different tweeter response curves: "flat" and "sloped." According to the supplied response curves, the 7.5Ω resistor results in approximately 3dB worth of attenuation.

The kit I received included "high end" hookup wire to run between the binding posts and the crossover (eight solid conductors) and between the crossover and the drivers (four solid conductors). Unfortunately, the time required to prepare and attach the supplied wire increased the total assembly time by a factor of nearly three relative to using conventional high-quality zip cord. Additionally, the incredible stiffness of the wire can cause undue tension on the solder joints, potentially breaking connections (in fact, one broke during shipment).

While many will argue for the sonic benefits of various exotic speaker cables over the more pedestrian zip cord, I doubt that any such benefits could be heard in the ½′–1′ lengths used in the Aria 5/Raven. Given the added time, frustration, and the greater chance of failure inherent with the supplied cables, you would do better to simply discard them and reach for a trusty spool of more traditional cable.

ASSEMBLY

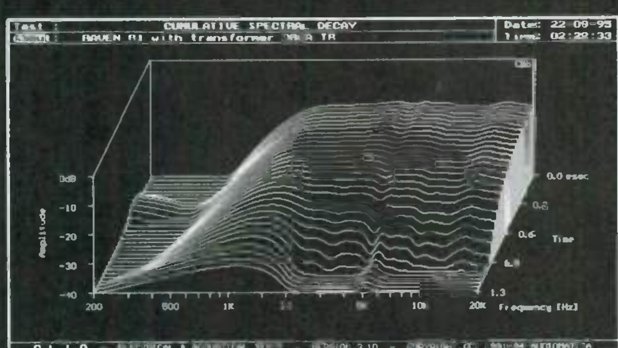
Assembly of the kit was fairly straightforward. The supplied foam was easy to cut using an X-acto knife, and it attached without difficulty using the built-in adhesive. Since this is a vented design, the foam is meant only to absorb internal reflections and not to alter the effective box size through adiabatic-isothermal conversion. Therefore, I placed the foam (as best I could) on one of each of the parallel surfaces.

With no guidance given for crossover layout, I generated my own (*Photo 2*), designed to minimize the interactions between the two coils, while remaining compact enough to fit

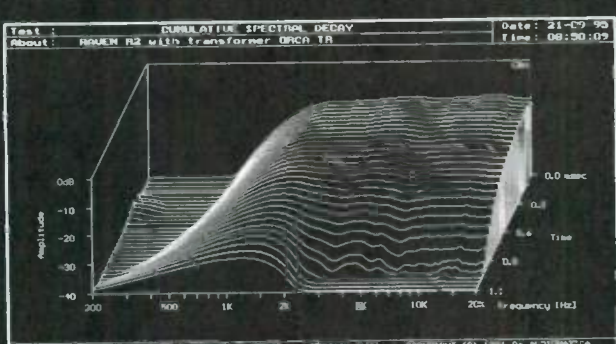
RAVEN

HIGH EFFICIENCY PURE RIBBON TWEETERS

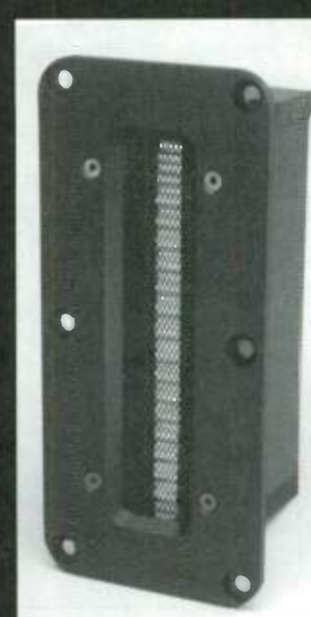
ORCA is sincerely proud of introducing these exceptional high frequency transducers from France. The RAVEN tweeter is a true ribbon tweeter, possibly the purest transducer available today. In a dome tweeter the signal is carried through the voice coil wire, and the sound is radiated by the dome attached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



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



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



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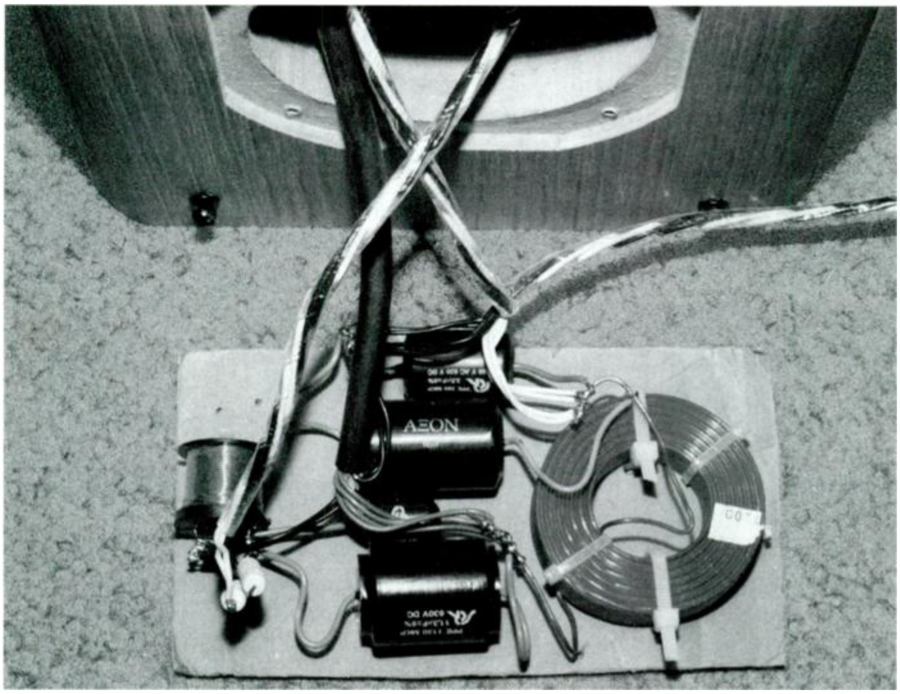


PHOTO 2: Detailed view of the assembled crossover. Coils are positioned to minimize interference.

in the small (7″ × 8.5″) space available at the bottom of the box. I mounted the components on a piece of heavy-gauge cardboard (a thin sheet of plywood or a printed-circuit board would work at least as well) using silicone to attach the capacitors and silicone reinforced with nylon ties to attach the coils.

Since I required the option of removing the crossover to fix any errors or to audition the system using the alternate tweeter resistor, I attached the assembled crossover with two small nails. Given the very limited access afforded by the mid-woofer hole, this was no easy task. While glue would provide an easy (but permanent) method of attachment, a set of pre-drilled holes for crossover mounting would be a nice improvement to the kit.

With the crossover and foam in place, all that was required to complete assembly was drilling a number of pilot holes in the cabinet for the screws used to mount the drivers and binding posts. With this done, the drivers and binding posts fit nicely into their recessed openings, and with screws in place the speakers were ready. Total assembly time was

TESTING THE ARIA 5/RAVEN 1 LOUDSPEAKER KIT

By Joseph D'Appolito

Author's note: I must state at the outset that I designed this kit for ORCA Design and Manufacturing, the distributor of the Focal and Raven drivers used in this kit. It will be interesting to compare the results of my tests with the original design data.

I ran a series of impedance, frequency response, and distortion tests on the ARIA 5/Raven 1 kit constructed by Mr. Stark. *Figure 1* is a plot of system impedance. Below 200Hz you can see the classic double-peaked curve of a vented system. The minimum impedance of 5.2Ω at 54Hz indicates the resonant frequency of the vented box.

An overall minimum impedance of 4.4Ω occurs at 240Hz. The final minimum at 2350Hz has a value of 4.9Ω, which qualifies the ARIA 5/Raven 1 as a 5Ω system. Impedance phase ranges from −40° to +29° over the full audio range. This should be a relatively easy load for most amplifiers.

FREQUENCY RESPONSES

Figure 2 shows the 1m on-axis frequency response of the ARIA 5/Raven 1 compared against the prototype built and tested during the design of this kit. Both curves are a composite of quasianechoic response data above 220Hz taken with the microphone placed on the tweeter axis at 48″, combined with near-field woofer and port data below 220Hz to get the complete curve.

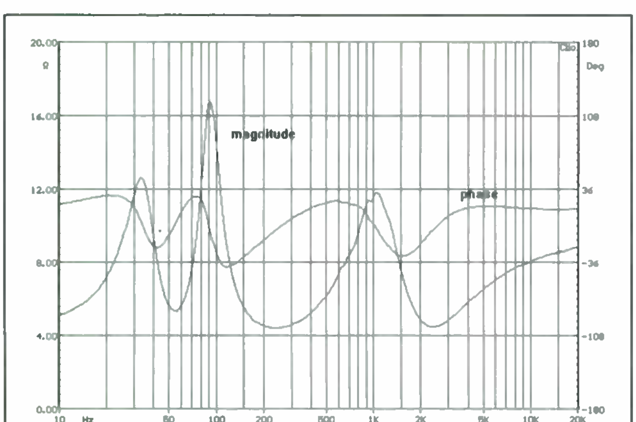


FIGURE 1: System impedance.

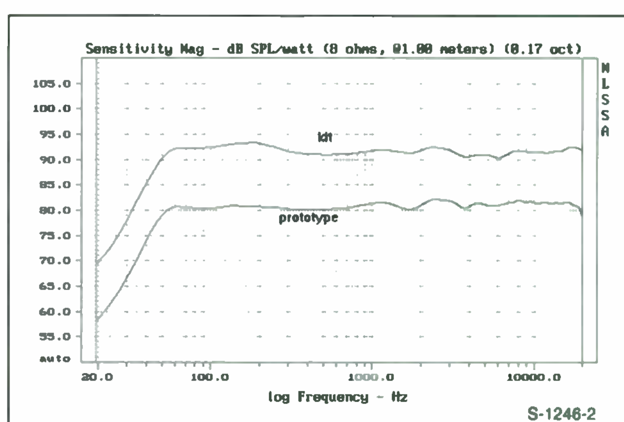


FIGURE 2: Full-range response of ARIA 5/Raven 1 kit and prototype.

approximately 45 minutes per cabinet using zip cord to connect the drivers to the crossover, and approximately two hours using the supplied cable.

LISTENING TESTS: METHODOLOGY

I primarily auditioned the Aria 5/Ravens on two systems: my home system and that of a friend. My system is located in a 14′ × 20′ room (speakers placed along one of the short walls) with a 10′ acoustically treated ceiling and a carpeted floor. No other acoustic treatments are currently in place. The system consists of a Denon CD player being used as a transport, an outboard DAC of my own design (using Burr-Brown PCM 1702s and a POOGE-5 output stage), a preamp based on the Welborne Labs hybrid linestage, and a Borbely 60W MOSFET amplifier.

For direct comparison with the Arias, my current speakers are three-ways of my own design using a side-firing NHT 12″ woofer, a pair of Scan-Speak 6.5″ mid-woofers, and a Focal 1″ tweeter. My friend's system is in a 15′ square room with plush carpet and a back wall that opens into another room. He uses a Technics CD player as a transport, an Audio Alchemy DTI and DDE, and a preamp of our design (based on AD812 op amps), and he had both a Bellas 1 (MOSFET based) and a Pass Zen amplifier on hand.

For comparison to the Arias, he provided a pair of Signet bookshelf speakers. In both listening environments, the speakers were placed on stands to raise the tweeter to approximately ear level. Source material varied widely over the weeks of listening, ranging from Mozart to The Modern Jazz Quartet to Metallica.

LISTENING TESTS: RESULTS

Phenomenal. Once we found the optimal speaker locations, the sound emanating from the Aria 5/Ravens was nothing short of phenomenal. Detailed and accurate without sounding "clinical," the Aria 5/Ravens were a joy to listen to and never became fatiguing.

While the original Aria 5s had the reputation of being excellent speakers marred only by a slight harshness in the high end, the high end produced by the Ravens was smooth and effortless. The midrange was equally as mellifluous with no hint of harshness or strain to be heard. Imaging was superb, and the system overall has a smoothness and clarity that few speakers possess, especially those that *Speaker Builder* readers can so readily build.

The low moving-mass of both drivers, when coupled with the solid boxes and a crossover designed to keep the drivers' inherent resonances and breakup modes out of play, produced excellent cumulative spectral decay and step response measurements (see D'Appolito's *Figs. 3 and 4*). No doubt these

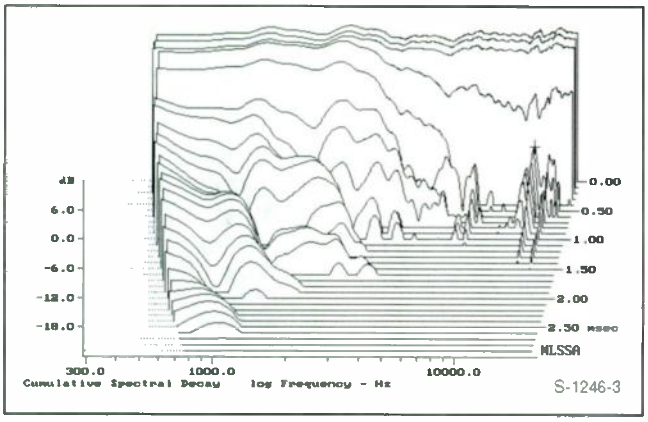


FIGURE 3: Cumulative spectral decay.

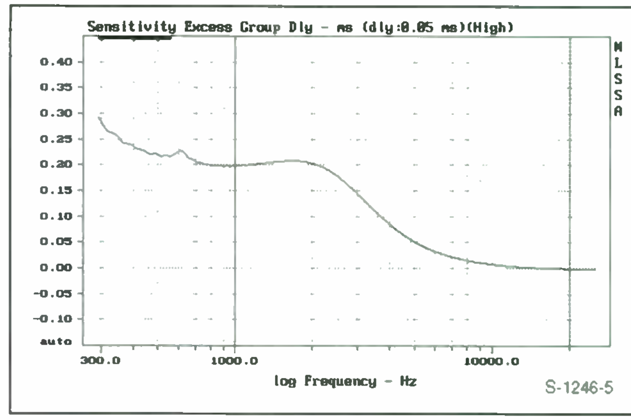


FIGURE 5: Excess group delay referenced to the tweeter.

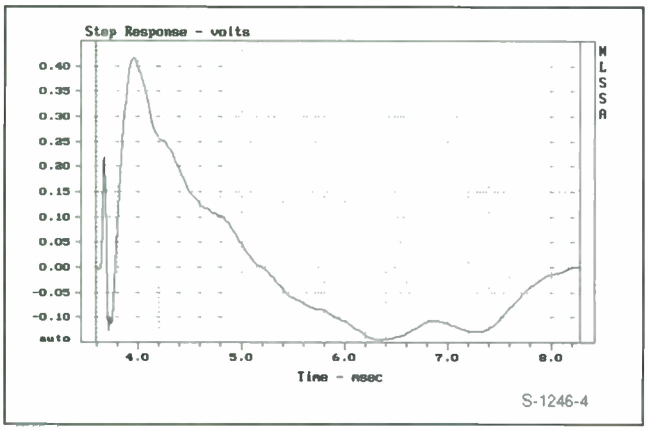


FIGURE 4: System step response.

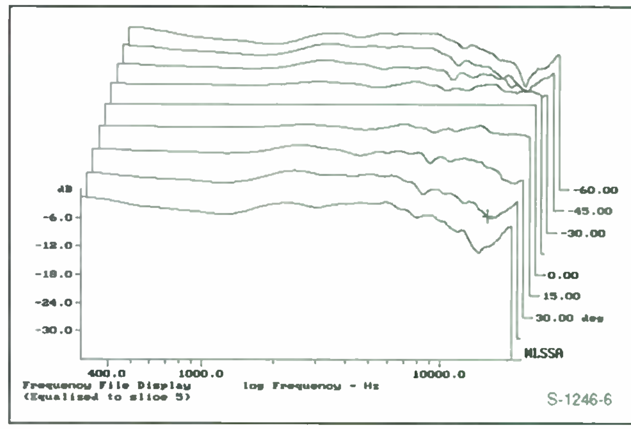


FIGURE 6: Horizontal polar response (−60° = 60° left).

The plot of the kit response is normalized to 1m distance to get system sensitivity. The prototype response is offset by 10dB for easy comparison. Both curves are admirably flat, but there are subtle differences possibly due to small differences in kit driver responses relative to the drivers used to design the ARIA 5/Raven 1.

The kit's on-axis response is flat within ±1.6dB from 60Hz–20kHz. This compares very well with the prototype response properties are one of the sources of the exceptionally clean sound.

While harmonic-distortion levels below 1% are not only typical but expected from audio components, loudspeakers rarely achieve this across the board. Perhaps this is why we rarely see distortion measurements made on loudspeakers (or at least why they're not often advertised). For the second- and third-harmonic-distortion levels (*Figs. 10 and 11*) to remain below 1% from 100Hz on up (going just above 1% at the crossover point) is quite an accomplishment. For the highest level of IM distortion to be 0.2% for the woofer and 0.5% for the tweeter is equally impressive. Both give a clue to the source of the smooth, uncolored sound.

A LOOK AT THE LOWS

The several criticisms to make about the Aria 5/Raven speaker are all relatively minor. To begin with, I would be remiss if I did not point out that deep bass is not the Aria 5/Raven's strong suit. C-0 organ pedals, *1812 Overture* cannon shots, and even less extreme examples of the lower end of the spectrum are not generated in all their glory. When compared with my current speakers, the foundation provided by the bottom two octaves was notably absent, and bass aficionados will certainly wish to augment this speaker with a subwoofer.

However, you can hardly expect a loudspeaker using two 5¼″ drivers to reach too deeply into the low-end. With approximately one-third the surface area of a 12″ woofer, a pair of 5¼″ drivers simply cannot move that much air. That said, the Focal 5¼″ midwoofers, in the vented alignment of the Aria 5, move a surprising amount. While not earth-shaking, the bass on the Aria 5s was far from anemic. When placed to take advantage of room-gain, the bass was quite reasonable for a speaker of any size—a real accomplishment for a small loudspeaker.

Despite the use of a D'Appolito alignment (designed to reduce vertical lobing), the sound quality was quite sensitive to the height of the stands used. While the horizontal polar response is excellent (*Fig. 6*), the vertical response is far less ideal (*Fig. 7*) and may contribute to this effect. Whatever the source, stands that place the tweeter at ear-level are a requirement for the Aria 5s.

Additionally, while the sensitivity was high enough to be adequately driven by the low-wattage Pass Zen amplifier, the speakers notably did not sound their best when attached to the Zen. Whether the somewhat complex load presented by either the phase (–40° to +29°) or the magnitude (4.4Ω–18Ω) of the impedance was the source, the single-ended Zen, with its 1Ω output impedance, was not the ideal amplifier for these speakers.

Bass all but disappeared and the midrange became quite thin.

Finally, as with many speakers, the grille cloths had a noticeable effect on the sound. The drivers are flush-mounted, and care has been taken to produce a smooth, protrusion-free baffle when the grilles are off. With the grilles in place, a ¼″ frame surrounds the drivers, creating diffraction effects. These effects were both measurable (*Fig. 15*) and audible.

CONCLUSIONS

There are many reasons why hobbyists choose to build their own loudspeakers. One common, practical reason is the desire to do more sonically with less money. At nearly $1200 for a pair of small speakers, you might not think the Aria 5/Ravens would fulfill this desire. To draw this conclusion would be a mistake, however.

At $1200 the Aria 5/Ravens are a bargain. More important than the fact that this represents a very good deal on the cost of the components (neither the Focal midwoofers nor the Raven tweeters are inexpensive drivers), this is a meager sum to proffer for a loudspeaker of such quality. The Aria 5/Raven's sonic competition comes from the very high end of the loudspeaker market where speakers often cost several times more. As such, it clearly satisfies this goal. The Aria 5/Raven is an outstanding loudspeaker that will greatly reward those who choose to build it.

response of ±1.5dB over the same frequency range. System sensitivity in the two octaves around 1kHz (500Hz–2kHz) averages 91.5dB/2.83V/1m.

The cumulative spectral decay (CSD) response for the ARIA 5/Raven 1 (*Fig. 3*) is a waterfall plot that shows the frequency content of the system decay response following a sharp impulsive input at time zero. Ideally, the response should decay to zero instantaneously. Real loudspeaker systems, however, contain inertia and store energy that takes a finite amount of time to decay.

The first three milliseconds (ms) of the CSD are shown with a total dynamic range of 32dB. The tweeter response is essentially gone in 0.3ms. There is some small amount of residual hash in the 12–14kHz range down about 18dB which persists out to 1.4ms, but it has no special character which might indicate a delayed tweeter resonance. This is excellent decay response.

Decay response below 3kHz is controlled by the woofer and its crossover network. Here again decay response is quite good. There are no distinct uni-modal ridges to indicate the presence of strong system resonances.

Figure 4 is a plot of system step response. It is obtained by a numerical integration of the system impulse response. The initial positive spike indicates the tweeter arrival. It is followed by the woofer arrival, peaking about 0.3ms later. The drivers are both connected with positive polarity, but the system is not time coherent.

A better view of this behavior is seen in *Fig. 5*. This is a plot of excess group delay versus frequency referenced to the tweeter's acoustic phase center.[1] In a time-coherent system this plot would be a flat line. Above 10kHz excess group delay is essentially zero as it should be, since it is referenced to the tweeter in this frequency range. The curve rises below 10kHz to a plateau starting just below 2kHz.

The difference in excess group delay between 20kHz and 1kHz points is 0.199ms, or 199µs, which is the woofer time delay relative to the tweeter. This time offset of the drivers is a direct consequence of the fourth-order acoustic Linkwitz-Riley crossover used in this system. Excess group delay is a very accurate indicator of driver time offset. The sonic effect of this lack of time coherence is a hotly debated topic with no clear conclusions.

SYSTEM POLAR RESPONSE

Figure 6 is a waterfall plot of horizontal polar response in 15° increments from 60° left to 60° right when facing the speaker. The microphone is at tweeter height for this series of curves. All off-axis plots are referenced to the on-axis response, which appears as a straight line at 0.00°.

For good stereo imaging, the off-axis curves should be smooth replicas of the on-axis response, with the allowable exception of the tweeter rolloff at higher frequencies and larger off-axis angles. You can see the expected rolloff of tweeter response at higher frequencies and larger off-axis angles. Tweeter response is down 10.1dB at 15kHz and ±45°. The corresponding figures at ±30° and ±15° are 4.7 and 1.7dB, respectively.

This is as good as many dome tweeters and better than most that I have tested. The tweeter's ribbon is recessed slightly within the slot formed by its magnetic gap. Tweeter output diffracts around this slot and the face plate, producing the uptick in response seen above

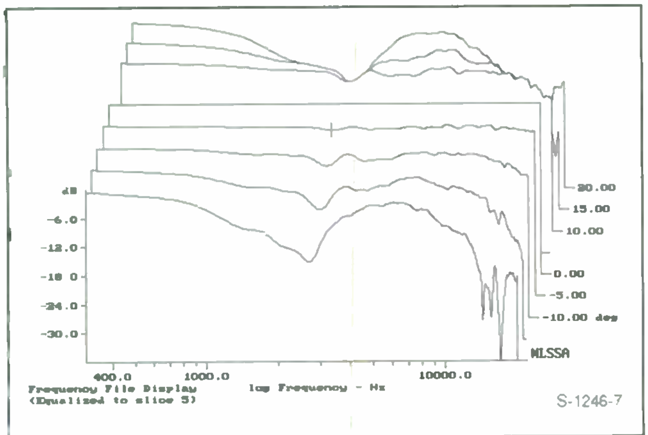


FIGURE 7: Vertical polar response (negative degrees are down).

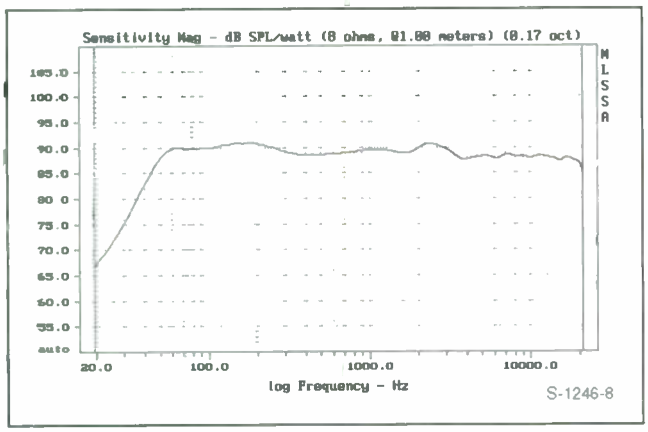


FIGURE 8: Response averaged over ±30°.

15kHz at horizontal off-axis angles of ±45° and ±60°.

Figure 7 is the waterfall plot of vertical polar response. Responses are shown in 5° increments from 20° below (−20°) the tweeter axis to 20° above it. Response at ±5° is within 1dB of the on-axis response. Worst-case response dips at ±10° and ±15° are 4.1 and 8.8dB, respectively.

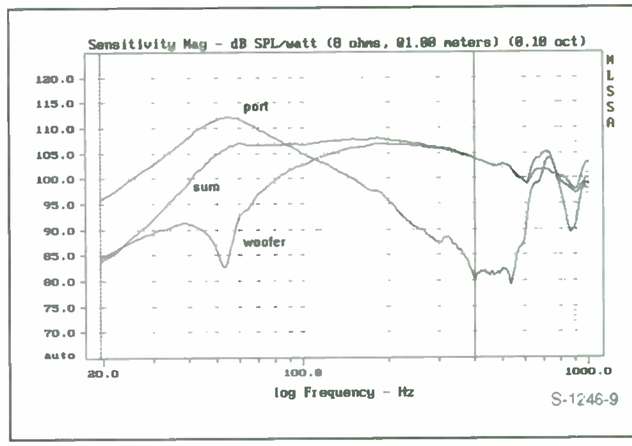


FIGURE 9: Woofer and port responses and their sum.

Both of these dips are in the crossover region. The tweeter ribbon is 6cm long in the vertical direction, which is equal to one wavelength at 5.7kHz. Above this frequency the tweeter will become more directional in the vertical. This explains the rapid drop in high-frequency response above 5kHz at ±20°.

The average response over a 60° horizontal angle (±30°) in the forward direction is shown in *Fig. 8*. The response is very smooth, sloping gently downward by about 2dB above 3kHz. This is excellent horizontal power response and suggests good direct field coverage in the primary listening area with little if any change in timbre. Image stability should be very good.

Near-field woofer and port responses are shown in *Fig. 9*. These responses are summed by the MLSSA system, giving proper weighting to the difference in areas of the combined woofers and the ports, to obtain the complete near-field system response. This response is then spliced to the quasianechoic response at 220Hz to get the complete system response, shown in

Fig. 2, without the use of an anechoic chamber.

The dip in woofer response just above 50Hz indicates the box tuning frequency. The port output is near maximum at this point. The port curve also shows a peak at 720Hz, which is the result of an "organ pipe" resonance in the port tube. This resonance causes a dip in the overall near-field response, but because the port exit is on the rear of the enclosure, it does not show up in the far-field response of *Fig. 2*.

DISTORTION TESTS

I ran harmonic-distortion tests at an average SPL of 90dB at 1m. Ideally, harmonic-distortion tests should be run in an anechoic environment. In practice, it is important to minimize reflections at the microphone during these tests.

Out-of-phase reflections can reduce the level of the fundamental while boosting the amplitude of the harmonic. In order to reduce the impact of reflections, I placed the microphone at 0.5 from the loudspeaker and analyzed only the first 40ms of data at each frequency to further reduce room contamination. This limited the lowest analysis frequency to 50Hz.

Figures 10 and *11* show second- and third-harmonic distortion levels in dB SPL versus frequency in 1/3 octave steps. System frequency response is also plotted on these figures. The second- and third-harmonic-distortion figures at 50Hz are 1.8% and 4.6%, respectively. The corresponding numbers at 100Hz are 0.6% and 0.2%. All woofer distortions are well below 1% above 100Hz. Considering the size of the mid-bass drivers, this is excellent performance.

Distortion does rise above 1% briefly at 2kHz. This distortion comes from the tweeter, which could benefit from a somewhat higher crossover frequency. But the higher crossover frequency would restrict vertical coverage in the crossover region. The world is full of compromises and there is no free lunch!

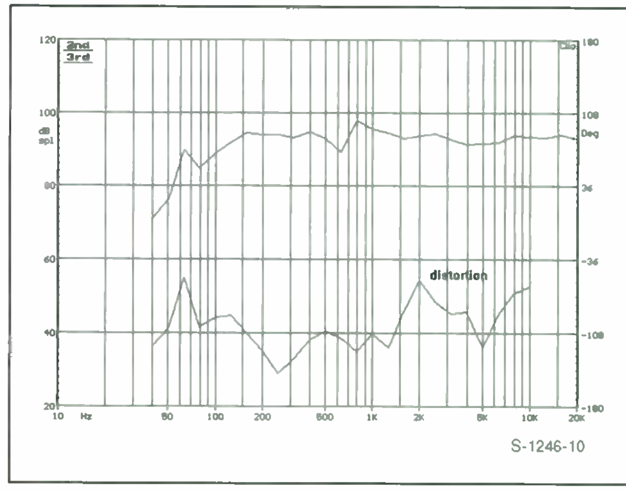


FIGURE 10: Second-harmonic distortion.

IM DISTORTION

The next area to investigate is intermodulation distortion. In this type of test two nearby frequencies are input to the speaker. Intermodulation distortion creates output frequencies that are not harmonically related to the input.

These frequencies are much more audible and annoying than harmonic distortion. Let the symbols f_1 and f_2 represent the two frequencies used in the test. Then a second-order nonlinearity will produce intermods at frequencies of $f_1 \pm f_2$. A third-order nonlinearity generates intermods at $2f_1 \pm f_2$ and $f_1 \pm 2f_2$.

I first examined woofer intermods by inputting 900Hz and 1kHz signals at equal levels. These frequencies are far enough below crossover that they will appear predominantly in the woofer output. Total SPL with the two signals was adjusted to 90dB at 1m. The ARIA 5/Raven 1 output spectrum is shown in *Fig. 12*. The two longest lines represent the input signals. The primary intermodulation distortion products and where they come from are:

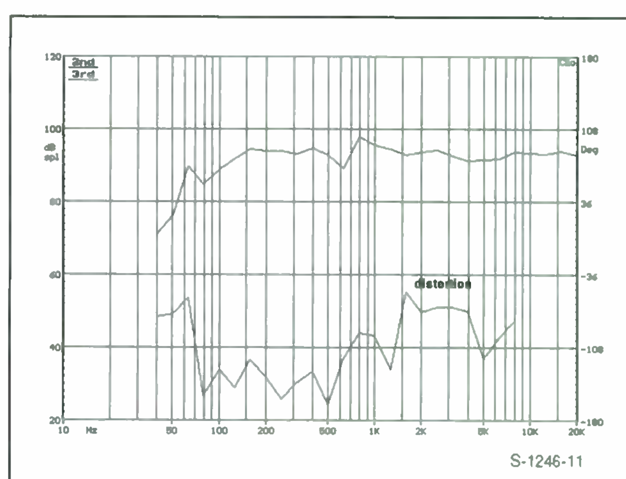


FIGURE 11: Third-harmonic distortion.

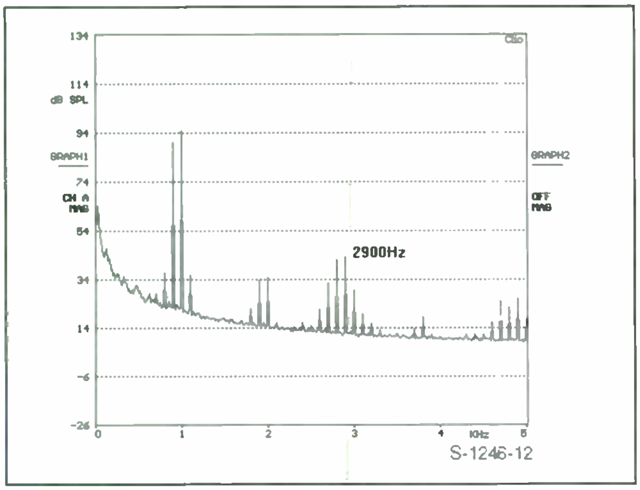


FIGURE 12: Woofer intermodulation distortion.

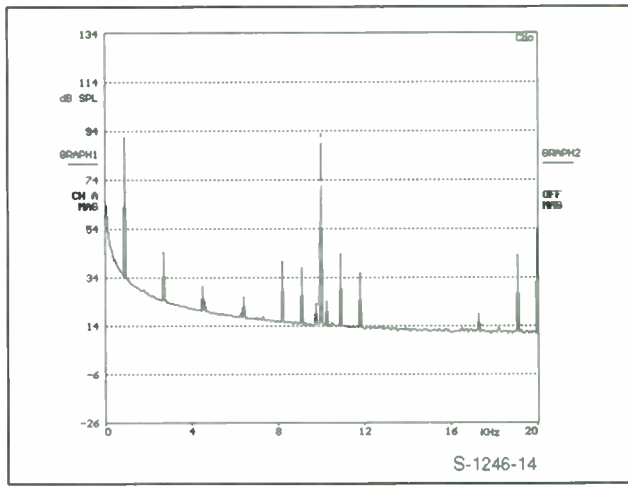


FIGURE 14: Woofer/tweeter crossmodulation distortion.

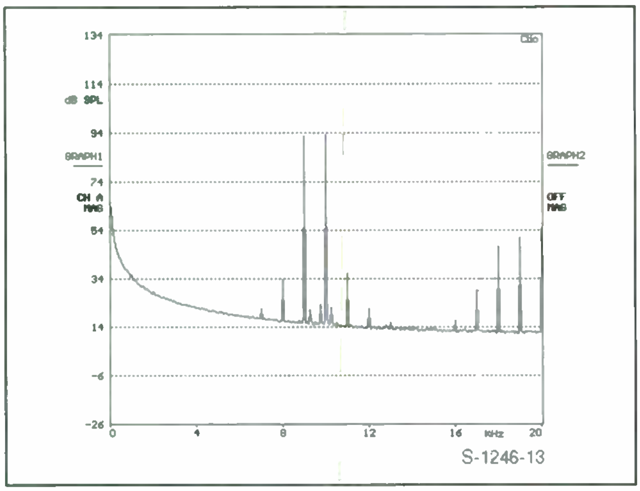


FIGURE 13: Tweeter intermodulation distortion.

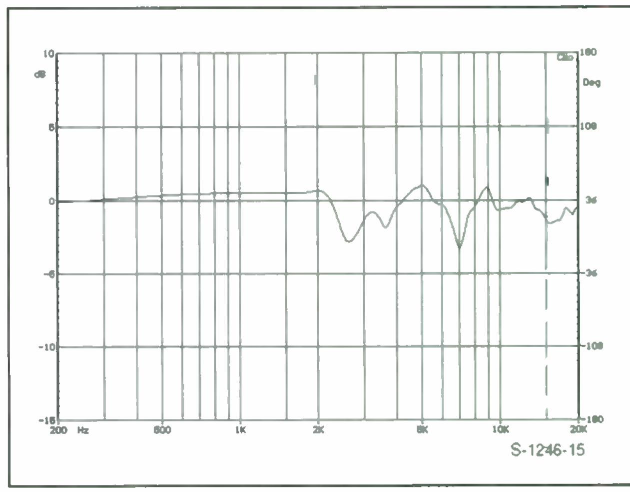


FIGURE 15: Effect of grille on frequency response.

1900Hz = 900 + 1000	(second-order)
800Hz = 2 × 900 – 1000	(third-order)
1100Hz = 2 × 1000 – 900	(″)
2800Hz = 2 × 900 + 1000	(″)
2900Hz = 2 × 1000 + 900	(″)

Other lines on the plot are due to harmonic distortion, which we have already discussed. The majority of intermods are third-order. The largest distortion product at 2900Hz is 54dB below the main output, which is equal to 0.2%. This is better than many tube amps!

I measured tweeter intermods with a 10 and 11kHz input pair also adjusted to produce 90dB SPL (*Fig. 13*). Intermods are at 9, 11, 12, and 19kHz. The worst-case tweeter intermod at 19kHz is down 46dB at 0.5%. This is very good performance.

The last intermod test examines cross-intermodulation between the woofers and tweeter using frequencies of 900Hz and 10kHz. (A 1kHz signal would produce intermods that fall on harmonic-distortion lines and confuse the results.) This spectrum is shown in *Fig. 14*. Intermods can be seen at 8.2, 9.1, 10.9, and 11.8kHz. The highest level at 11.8kHz is 51.5dB down at 0.3%.

Ideally, there is no mechanism by which 900Hz and 10kHz signals can mix. But in practice, cross-coupling in the crossover networks and the common return wire introduce low frequencies into the tweeter and high frequencies into the woofers. This makes the case for bi-wiring. (At extremely high SPLs nonlinear mixing can occur in the air, but this is not the case here.)

I conducted all of the above tests with the grille off. *Figure 15* shows the response of the ARIA 5/Raven 1 system with the grille on, but referenced to the response with the grille off. That is, it plots the *difference* in response under the two conditions. Below 2kHz the grille has no significant effect. Above 2kHz, however, the grille causes rather ragged response deviations of 4dB peak-to-peak. This grille is clearly of cosmetic use only.

References

1. J.A. D'Appolito, *Testing Loudspeakers*, Audio Amateur Corporation, Peterborough, NH, 1998.

A NOTE ON TESTING

I tested the ARIA 5/Raven 1 kit in the laboratories of Audio and Acoustics, Ltd., using the MLSSA and CLIO PC-based acoustic data-acquisition and analysis systems with an ACO 7012 ½″ laboratory-grade condenser microphone and a custom-designed wideband, low-noise preamp. I conducted polar response tests with the aid of a computer-controlled OUTLINE turntable on loan from the Old Colony Division of Audio Amateur Corporation.

Book Report

TESTING LOUDSPEAKERS

Reviewed by G.L. Augspurger

Testing Loudspeakers, by Joseph D'Appolito. Available from Old Colony Sound Lab, PO Box 876, Dept. B90, Peterborough, NH 03458, (603) 924-6371, FAX (603) 924-9467, E-mail custserv-@audioXpress.com. $34.95, plus $6.50 s/h. Published by Audio Amateur Press. 174 pp.

This book is well written and well produced, and contains a lot of information. It is aimed at the serious hobbyist—someone who knows how to use a multimeter and how to hook up a crossover network, but who has not made a career of designing and testing loudspeakers. If you have recently acquired one of the computer-based test systems, the book will provide additional insight into how these systems work. If you are considering buying such a system, read the book first; it will demystify much of the technical jargon and show you which features are really important.

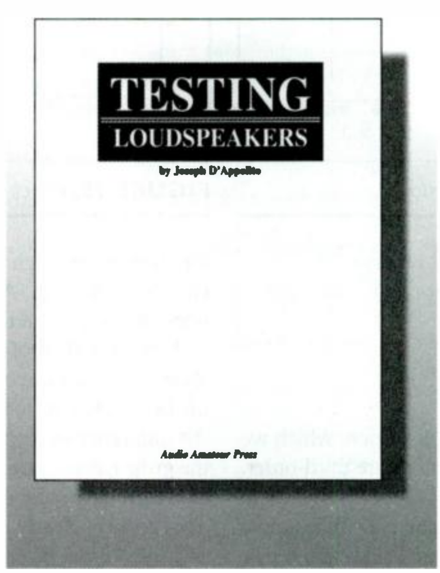
Testing Loudspeakers is not perfect. In my experience, every technical book contains errors and omissions. Since part of a reviewer's job is to find things to carp about, I will point out some of its shortcomings. But I will also try to give you a feel for the range of material covered and the author's way of presenting that material. The object of this review is not only to tell you whether I liked the book, but also to help you decide whether it belongs on your bookshelf.

FORMAT AND TYPOGRAPHY

When I first opened the book I was struck by its clean, easy-to-read page layout that is consistent from beginning to end. This is a happy contrast to some current technical volumes that seem to have been pasted together from assorted computer printouts. Pages are 8½″ × 11″, with text printed in two columns. Illustrations and math equations are held to column width, which makes it easy to compare information in various parts of the book.

But consistency can also have its drawbacks. In three or four cases the author uses a graph to make a point, but the graph in question is simply too small to identify the information in question. In those few instances, perhaps an enlarged graph could be printed on its own page at the end of the chapter. A similar complaint can be made about some of the equations where subscripts and superscripts are too tiny to make out. In some parts of the book, particularly Chapter 6, textual references are several pages ahead of their associated graphs. Devoting a page exclusively to illustrations where needed would minimize the annoyance of repeatedly flipping back and forth.

Proofreading is commendably thorough. I noticed only one typographical error, which has by now been corrected. I did not attempt to verify all of the math equations, but those few that I did check were printed correctly.



GENERAL ORGANIZATION

The book has a total of seven chapters, each of which is divided into clearly marked topics and begins with a brief outline of what follows. Chapters 2–5 are concerned with testing in the analog domain using relatively inexpensive instrumentation. The remaining two chapters contain almost 40% of the book's content and are devoted to computer-based digital test systems.

Chapter 1 is something of a novelty. The author begins with a clear statement of the book's objectives, the scope of its subject matter, and what the reader should know. This is followed by five pages of preview in which each chapter is summarized in detail. The preview is not a topical outline, but a kind of chatty description of things to come. This might be dismissed as unnecessary padding, but I found it a handy lookup reference. How many times have I thumbed through a text page by page because the paragraph I remembered could not be identified in the index or table of contents? Here in Chapter 1 I can effectively scan the entire book in about 15 minutes.

Each chapter concludes with its own list of references. These are well chosen. Most of them are important, well-known books or technical papers that you can track down without too much difficulty.

TUTORIAL APPROACH AND STYLE

In writing *Testing Loudspeakers*, Joseph D'Appolito has drawn on his long experience in loudspeaker design. The emphasis is on real-world test methods, using case histories as examples. There is a fair amount of math, but most of it is simple algebra. If the math is essential to the point being made, then the text walks you through it step by step, using a practical example. If it is really important, there are two or three examples. To me this is one of the best features of the book. It is almost impossible to be uncertain about a particular procedure after reading through one of these detailed examples.

The prose style is direct and informal. The book is meant to be read, not just used as a reference. In any given section, the main point is almost always clear and correct. The tone is lightened by numerous tangential observations and comments. These are interesting but occasionally misleading, and in a few instances they are dead wrong. For example, in Chapter 1, Kellogg and Rice are credited with the invention of the direct radiator loudspeaker. The situation is clarified somewhat in Chapter 2, where it becomes apparent that the modern, moving coil, direct radiator is what the author had in mind.

I confess that I am bothered by acronyms. Using x in an equation is appropriate: dropping SUT into an otherwise understandable English sentence is not. However, this is a matter of taste. If you enjoy reading that the *LDC* suggests using a DMM to measure the EMF of your DUT, then this is the book for you.

ANALOG TESTING

Chapter 2 is a self-contained short course covering the basic electromechanical loudspeaker model, the factors that affect a speaker's impedance curve, how to measure impedance versus frequency, and how to use the impedance curve to derive most of what else you need to know.

Chapter 3 continues the analysis, explaining what you can learn about loudspeaker-/box combinations by simply measuring their impedance characteristics. The author does a good job of explaining how the effective cone mass includes air mass, and how to estimate air mass under different loading conditions. However, the importance of this factor may be overemphasized, especially since the sources quoted do not agree with other equally reliable authorities. Also, I wish there had been a little more discussion of mutual coupling—how do I calculate the air mass of *two* loudspeakers in a box?

Chapters 4 and 5 move on to acoustical testing, with many good recommendations for loudspeaker testing in ordinary rooms. I found a few nits to pick, but this is good, hands-on information. Almost any reader, regardless of experience, will find useful tips in this section.

Unfortunately, Chapter 5 also includes a couple of lapses. One is a confusing presentation of low-frequency power response, in which it supposedly differs from near-field response, but somehow relates to room response at various listening locations. This was the only place in the book where I simply could not tell what the author was trying to say.

The other is a case history involving a conventional woofer crossing over to a long, mid-range line source. Here the author's point is clear and well taken: the relative level between low- and high-frequency sources varies with distance.

As a sidelight, the author then suggests that the coverage pattern of the line source takes the form of an expanding cylinder, which seems to make sense. Additional measurements are presented to support the thesis. Unfortunately, this is a case where common sense is wrong, as anyone who has spent much time with one of these devices knows. The supporting measurements are flukes, probably contaminated by strong room reflections.

I bring this up not to embarrass the author, but to point out that even a seasoned professional can be misled now and then. Murphy's law prevails.

COMPUTERS AND THE DIGITAL DOMAIN

Chapter 6 attempts to cover a lot of ground in relatively few pages. It starts by explaining the relationship between time-domain and frequency-domain data via the Fourier transform. Then, without pausing for breath, we are treated to discussions of impulse response, periodic signals, discrete frequency spectra, the sampling theorem, the discrete Fourier transform, and the fast Fourier transform.

All of this is pretty standard material that must be covered somehow. To the author's credit he does not simply retrace the footsteps of a standard textbook presentation. For one thing, almost all of the math is given in terms of sine and cosine calculations, which makes it much easier for the average reader to understand. Also, the order in which concepts are presented is different from most other sources. For me it was initially a little confusing, but after I had read the chapter three times I could follow the logic of the presentation without difficulty.

Becoming familiar with impulse-response measurements and the power of the Fourier transform requires much more than one reading. You need three or four sources to explain things in different ways so you can compare one presentation with the others, going back and forth until the concepts gradually gel. I would definitely include this book in my list of recommended sources.

At this point we are only two-thirds of the way through Chapter 6. The remaining pages are devoted to a good explanation of maximum length sequences, then definitions of phase response, minimum phase, excess phase, group delay, and what is labeled the acoustic phase center. The difference between group delay and true time delay is explained at some length. This section should be required reading for at least one professional loudspeaker reviewer.

Reading Chapter 6 is a little like watching a juggler keep 20 plates spinning without dropping any of them. It would be easier for both the author and the reader if the material were separated into two chapters and allowed to expand a little.

I found one significant error in the discussion of minimum phase systems. The mathematical definition is correct, but the graphical interpretation ("If the slope is positive, phase will increase...") obviously is not. Perhaps the author was trying to condense Heyser's rules on this topic and got carried away in the trimming process.

The final chapter is devoted to features offered by several computer-based measurement systems and examples of the kinds of measurements that are possible. Examples are confined to two systems with which the author is familiar: CLIO and MLSSA. Most of the information is interesting and the examples are useful, but it seems to me that the author's approach is needlessly restrictive.

On the one hand, I really don't care what keystrokes are required to initiate a particular test. On the other hand, I would be interested in a brief comparison of MLSSA processing versus the TEF20 system. Alas, there is not even a hint that time-delay spectrometry exists, even though Richard Heyser is referenced in other contexts.

SUMMARY

The author has accomplished about 97% of what he set out to do. The book fills a real void in technical literature and is packed with useful information. The few errors I noticed are mostly concerned with peripheral comments rather than main topics. Chapter 2 alone is worth the price of the book. If you are seriously interested in loudspeaker testing, order a copy.

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SB Mailbox

BLUEPRINT BLUNDERS

I noted three mistakes in the plans for the Snail II speaker (*SB* 3/98, p. 20).

1.) Part #3 in Table 1 is missing from the list, and all the parts from 3 on are mislabeled (i.e., part 3 is really part 4, and so on). 2.) The measurement for part 12, the rear horn braces, is incorrect (p. 24). It should be 22¼″, rather than 21¼″ (the author forgot the ½″ at either end which is cut off later). 3.) The measurement is also incorrect for part 14, the front horn braces (p. 25). It should be 26″ long, not 25″. Again, the ½″ at either end is not included.

Al Amaral
alamaral@bit-net.com

Bill Fitzmaurice responds:

Oops! Reader Amaral is correct to some extent. The actual size of the piece of plywood from which the back and rear horn braces is cut is 23.5″ × 9.5″; the pattern for the front horn brace is cut from a 27″ × 15.5″ piece. Part 3 is correctly labeled in the list, but part 4, the upper back, is omitted.

These errors are not as serious as one that I found: the total of vertical measurements on the side view (p. 22) add up to 37″, while the cabinet is supposed to be only 36″ high! That shows what can happen when you're transferring notations from original blueprints, which have been changed and scribbled over during the building process, to a clean sheet for the graphics guys to attempt to decipher.

To clear up the source of the error, the measurement from the horn "point" to the inside of the bottom of the cabinet is 9.5″, not the 10.5″ as shown (Fig. 2). I apologize for the goof, which I attribute to working too late in the pm, and the influence of a good grape with dinner.

This points out the absolute necessity to be positive of both positions and exact measurements by drawing the cabinet parts positions on the cabinet sides prior to cutting. Anytime you attempt a project as complicated as this one, it's best to be very sure of your course before starting to cut.

I noticed another problem in the text. I didn't make it clear that after initial cutting, ¾″ of material should be removed from the rear horn brace to allow for the position of *the baffle. Most readers who are experienced builders wouldn't need to be told this, but that's no excuse for my omission. With readers watching over my shoulder, I'll need to be more careful in the future.*

One final error I noticed was a pure typo: two Carvin TR1801s paralleled have an output capacity average of 130dB from 1200W input, not the 120dB incorrectly stated on p. 22. Correctly stated, the Snail II will put out 132dB with only 300W in. As I said before, horns rule.

PEAK REMEDY

I noted with interest "Low-Budget 2-Way with Top-Mounted Tweeter" (*SB* 7/97), by Carl Richard, since I had built with this same woofer. His measured frequency response (Fig. 2 in his article) shows a nasty peak at 4k, which I'm sure the woofer is causing. My *Fig. 1* shows the response of the woofer with Mr. Richard's crossover values, and the 4k peak is clear.

G.R. Koonce
Liverpool, NY

Carl E. Richard responds:

I thank G.R. for taking the time to respond with what appears to be an excellent recommendation for flattening the 4kHz hump in the response curve of the Parts Express woofer referenced in my article. It's an amazing coincidence that he had been working on the same woofer that had such limited availability. I believe I will follow his recommendation for the Zobel.

HELP WANTED

I am currently involved in a subwoofer design, and am looking for literature that may help. My design places the subwoofer (two 15″ drivers) section in a ceiling configuration. I have a multiamp setup and more space available there than in the area once a living room but now simply equipment and speakers. I first dismissed that option, but the more I thought about it, the idea seemed very effective and natural...but, after a major failure using a Isobarik configuration-type cabinet, I wish to avoid more setbacks.

I was hoping someone else had attempted a similar mounting and could reply with some comment or advice.

John Straub
3604 Hermosa Ct., NE
Salem, OR 97305

I am looking for information about RCA-type MI 11411 speakers, which look like the Olson speakers but have a coaxial tweeter.

J-L Delvaux
CE, Cort 80 4/11
200 rue de la Loi
B-1049 Bruxelles
Belgium

Readers with information on these topics are encouraged to respond directly to the letter writers at the addresses provided.—Eds.

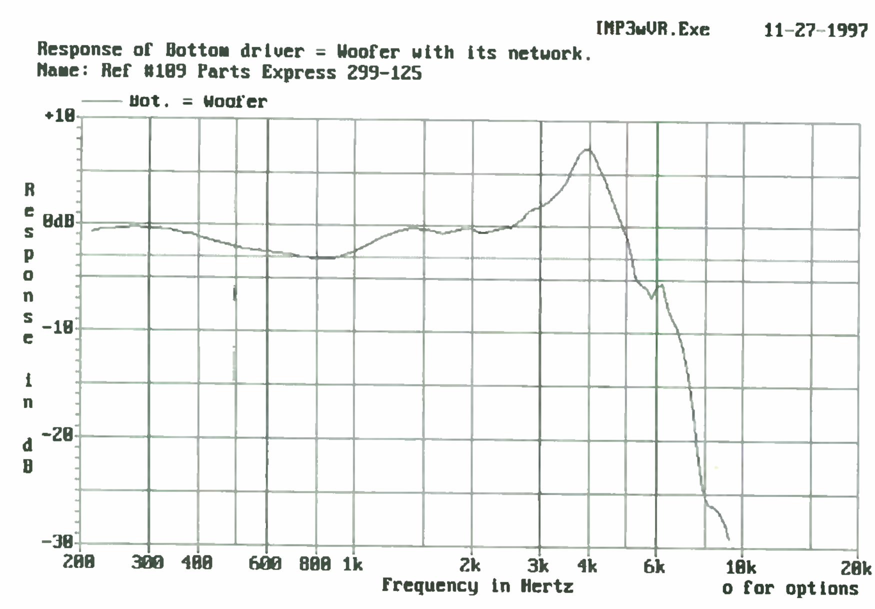

FIGURE 1: Woofer response.

Ask SB

By G.R. Koonce

AN "IMP"ORTANT QUESTION

My question concerns the procedure for measuring the frequency response of individual drivers for use with a network optimizing program. I am modifying the network of an existing two-way speaker system so the drivers are mounted on the enclosure baffle. Also, I am using IMP and the program IMPRoom. This program gives an option of measuring driver responses or enclosure response.

Since I wish to measure the individual woofer and tweeter responses, but mount them in the enclosure, should I use the enclosure or driver option? Is it best to measure directly on each driver's axis, repositioning the mike after each measurement, or take both measurements from one point—say, midway between the woofer and tweeter—at the distance specified by the enclosure option? Is it preferable to measure driver responses on the finished enclosure baffle, or remove the drivers and measure on a circular test baffle as specified by IMPRoom's companion program, IMPSpace?

Robert Mallory
Sedalia, MO

G.R. Koonce responds:

The basic question you ask is how to test your drivers to get data files representing the drivers to use with a crossover (CO) optimization program. The correct answer, while not very useful, is that you should test them in the way your optimization software specifies. This is because different programs may require the driver tested in different ways. (I will describe some similar considerations later.)

You refer to the IMPRoom and IMPSpace programs which are on my IMP-Aid disk distributed by Old Colony (PO Box 876, Peterborough, NH 03458, 603-924-9464, FAX 603-924-9467, E-mail custserv@audioXpress.com). Both programs are aimed at helping you establish a valid test setup for conducting quasi-anechoic testing with a time-gated tester, such as IMP or Audiosuite by Liberty Instruments, Inc. (6572 Gretel Ct., Middletown, OH 45044, Voice/FAX 513-755-0252).

The basic rule is that to obtain a valid far-field test that incorporates all acoustic aspects of a device under test, the microphone must be back at least three times the "extent" of the device under test. Thus, to test a lone 8″ driver the test microphone should be back at least 24″. This is the calculation that the IMPRoom program makes when in the "driver response" mode.

If you had a pair of drivers to be tested simultaneously (with their CO), then you would have to put the microphone back three times the distance spanned by the two drivers. This would still be the "driver response" mode in IMPRoom, because you are looking for just the response of the driver pair, and for driver diameter, you would enter the total distance to span the two drivers.

If you want to ensure your test is accurate for the two drivers with CO mounted in an enclosure including all enclosure effects, use the "enclosure response" mode of IMP Room. You enter the enclosure dimensions to determine the minimum test distance; this makes sure your microphone is back far enough to include edge diffraction effects and other aspects of the enclosure.

So in theory, to test individual drivers mounted in an enclosure, you must use the "driver response" mode; if you want just to ensure that all effects of the driver are included, use the "enclosure response" mode.

The problem is that the size of your room limits the maximum test distance before echoes from the room boundaries severely restrict the low-frequency limit of your test, so your room size may not support valid testing for the enclosure size. I will give you my recommendation on whether or not I would test the individual drivers in the enclosure after I discuss some of the basic problems.

The function of a good CO network optimization or modeling program is to take in the frequency response and impedance of each driver (generally as test developed files)

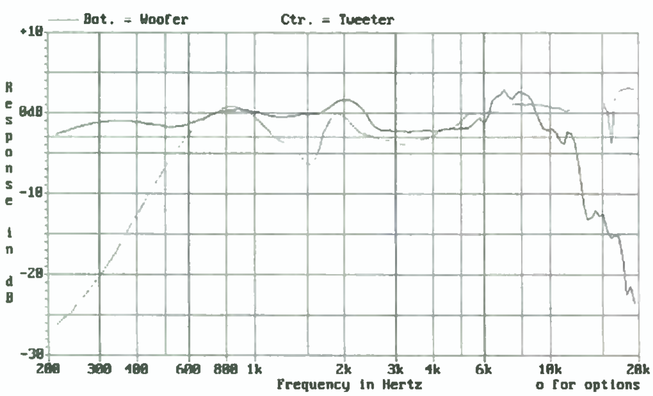

FIGURE 1: On-axis responses of small woofer and dome tweeter at 1m.

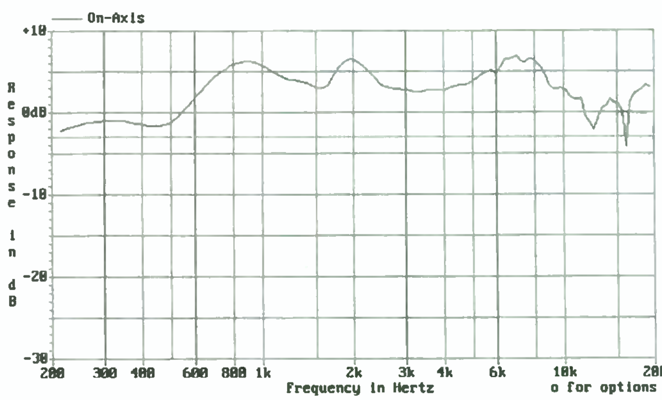

FIGURE 2: Summation of woofer and tweeter using Hilbert phase shift.

and develop a CO network that produces an acceptable acoustic response on and about your listening axis. It is important that you test the drivers in a way that is compatible with what the particular program wants. The driver input-impedance files are not a problem, so I will not discuss them. The problem is how you handle the frequency-response files. There are three important items relative to these files that we should discuss:

1) The program may require the response files in true absolute decibels or in relative decibels. Absolute means the files report the actual sensitivity of the drivers, but the testing is more difficult, because you must be sure to test the drivers at the same test distance and driven with the same test voltage level. Setting the same test voltage level can be very difficult if you are using an MLS or other "noise like" signal. To produce absolute response files, you must also have a calibrated microphone for which you know the sensitivity.

For relative response files, you simply set the basic response of each driver near 0dB. Relative response files are much easier to generate, but you lose any ability to set the padding required on the drivers, because the program does not know their relative sensitivities. The program may allow entry of factory-rated sensitivity, but I have found this data to be unreliable.

2) How you handle the driver phase shift in the file. When you measure the acoustic response of a driver far field to be sure you include all aspects of the driver response, you are actually measuring two components of phase shift. The first is from electrical input to driver acoustic output (which you want), and the second is the phase shift caused by the transit time from the driver to the microphone (which you don't want).

Unfortunately, the transit-time phase shift is a function of frequency and is not easily corrected. When working with a single driver, the transit time phase shift is not a problem, but it is when summing the response of two or more drivers. If, for example, you test at 1m, the transit-time phase shift is about 10,470° at 10kHz! In theory, if you could accurately position the microphone, the transit-time phase shifts would all subtract out, but this is not practical. An error of only 0.1" represents a 27° phase error at 10kHz, and you probably do not know the driver's acoustic center location to that accuracy.

To fully correct the transit-time phase shift, you must accurately know both the velocity of sound and the distance from the driver acoustic center to the microphone. The normal approach to this problem is to replace the measured phase shift with the Hilbert phase shift for the driver. This is the phase shift a minimum-phase network would have if it had the same amplitude response of your driver. The minimum-phase network is a reasonable assumption for individual drivers (without a whizzer), but is clearly not valid for a full system with a CO network.

When you remove the transit-time phase shift by using the Hilbert, you unfortunately also lose information about the location of the driver acoustic center. Thus a design program using the Hilbert phase shift requires entering a horizontal offset for each driver to correct for the relative acoustic-center position (or zero delay plane). IMP and Audiosuite are capable of generating output files that have the Hilbert phase shift substituted for the measured phase shift. It is likely that some optimization and modeling programs are actually generating the Hilbert phase shift, internally, which you must take into account in your work.

What is the problem with using the actual measured phase shift containing the transit-time phase shift for a pair of drivers? Since this phase shift is a function of frequency, the phase shifts of the two drivers constantly go in and out of phase, causing dips and peaks in the acoustic summation response that do not occur when you measure the two drivers together with their CO network.

Figure 1 *shows the on-axis responses of a small woofer and low-resonance tweeter that have a considerable overlapping frequency range.* Figure 2 *shows the acoustic on-axis summation of these two drivers without any CO, and the result looks as you would expect these two drivers to sum.* Figure 2 *used the Hilbert phase shift, in its file for each driver.* Figure 3 *shows the results for the same conditions when the files used the actual measured phase shift, and the dips and peaks caused by the transit-time phase shift are clearly evident.*

I have compared the results of modeling when using the Hilbert phase shift for drivers with actual tests of the driver pair with CO many times, so I know this approach produces agreement. The poor results shown in Fig. 3 *are a product of the test technique and do not really exist.*

3) On what axis do you measure the response? If the CO design program produces only the system acoustic response on-axis, then you need to test the drivers on-axis. If the program will plot the system acoustic response at other angles, then you may need to generate files at these other angles unless the program is capable of modeling the driver-response change as you move off-axis. My experience is that knowledge of only what the system acoustic response does on-axis is not sufficient to produce a good CO design. You need to know how the drivers with their CO will sum on and about the axis on which you will listen.

Now back to the question of whether you should test your drivers in the actual enclosure or in a test baffle. It would seem that testing in the actual enclosure makes good sense because it would include the effects of the cabinet on the actual system. However, I

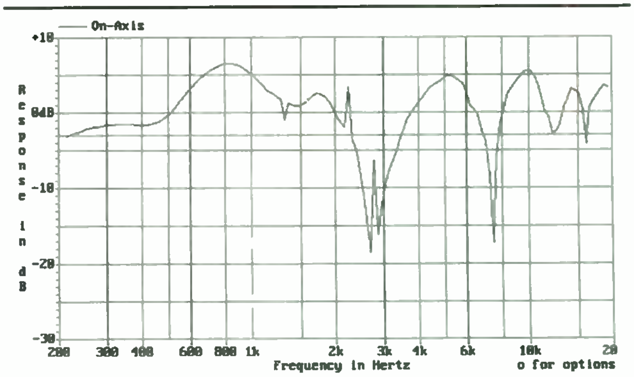

FIGURE 3: Summation of woofer and tweeter using measured phase shift.

would not test the individual drivers in the enclosure for the following reasons:

1) Trying to include the enclosure effects will greatly increase the minimum test distance as developed earlier. My room size will not support measurements of any reasonably-sized enclosures down to a usable lower frequency limit. If you test in the enclosure without meeting the minimum required test distance, you will have results that may include some "box" effects, but not others. This is not a prescription for good, consistent results.

2) From an engineering standpoint, you always want to break a problem into the smallest possible pieces for analysis. If an anomaly appears in a response, I want to know if it is due to the driver or to the enclosure. You can work on enclosure problems by front-panel damping material and for new designs via cabinet shaping. Therefore, I would test the drivers in a way that showed only their own response (if possible).

3) The inclusion of enclosure effects (edge diffraction, and so on) may cause the assumption of each driver being a minimum-phase network to be violated. Thus, with a program using Hilbert phase shift (in the data files or generated internally), the plots the program produces will not represent what you would actually measure or have when the CO the program develops is used with the drivers and enclosure. This would defeat the purpose of using such a program to develop the CO network.

With all these problems, does an optimization or modeling program that takes in test data files produce usable CO designs? The answer is definitely yes! I have been working since IMP first arrived to develop good testing techniques. The best approach I found is to mount the driver in a large round test baffle. If you make the center portion of such a baffle a removable insert, you can mount multiple drivers in the baffle for testing together with their CO. It is possible even to mount small enclosures flush into the baffle to measure their responses with edge effects removed.

How do you size such a baffle? Your room size will set a limit on what maximum time period you can use FFT analysis before boundary echoes cause a problem; for my room, it is in the 3–4ms range. Size the baffle so that the time for the sound to move from the driver to the edge of the baffle and then diffract to the microphone exceeds the main arrival time by the length of FFT time you can use. For my room, a 6′ diameter baffle works well (see Photo 2 in SB 6/96, p. 11).

For the last year, I have been working with a CO modeling program that takes in the impedance and frequency-response files (with Hilbert phase shift) generated by Audiosuite. My program is not an optimizer; you must manually change the CO values. I find this a plus because you see just what effect each component has on the system response. COs developed by this approach have been tested with multiple drivers in the test baffle, and the agreement is excellent.

Since you can develop the CO just as well by testing, why do you need a program? To develop the CO by testing, you must design a variety of COs, gather the parts for each of them, and then test each at various angles. This process may take weeks, and you generally give up before hitting the best design. Once you have tested the drivers with a modeling program, you can play on the computer and in hours find out if a driver pair has any chance of working together. If so, what is the best CO to use, and what is the optimum listening angle? You can quickly develop systems that have worked out well when built.

If the program that you plan to use will support it, I recommend that you test the drivers in a baffle, on-axis, and at a convenient distance, such as 1m. This should allow design of a CO that will work with the driver spacing dictated by your existing enclosure that performs well—drivers permitting.

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M13SG-09-08, 5" Shielded Cast Frame Woofer

This magnetically shielded 5" woofer is perfect for A/V applications. It features a cast magnesium basket, high damping rubber surround, treated paper cone and a very smooth frequency response. Best results will be obtained in vented enclosures with .10 to .45 cu. ft. of internal volume.
◆Power handling: 50 watts RMS/75 watts max ◆Voice coil diameter: 1" ◆Voice coil inductance: .70 mH ◆Nominal impedance: 8 ohms ◆DC resistance: 5.6 ohms ◆Frequency range: 54-5,000 Hz ◆Magnet weight: 12 oz. ◆Fs: 54 Hz ◆SPL: 88 dB 1W/1m ◆VAS: .42 cu. ft. ◆QMS: 1.50 ◆QES: .46 ◆QTS: .35 ◆XMAX: 2.0 mm ◆Net weight: 2.6 lbs. ◆Dimensions: A: 5-1/2", B: 4-1/2", C: 2-3/4", D: 3-3/8", E: 1-3/8". ***Ask for part #297-304.***

M17SG-09-08, 6-1/2" Shielded Cast Frame Woofer

This affordable 6-1/2" shielded woofer is well suited for any A/V application. It features a cast magnesium basket, treated paper cone, rubber surround and a very smooth frequency response. Designed for vented enclosures with .35 to 1.00 cu. ft. of internal volume.
◆Power handling: 50 watts RMS/70 watts max ◆Voice coil diameter: 1" ◆Voice coil inductance: .7 mH ◆Nominal impedance: 8 ohms ◆DC resistance: 5.6 ohms ◆Frequency range: 34-5,000 Hz ◆Magnet weight: 12 oz. ◆Fs: 34 Hz ◆SPL: 89 dB 1W/1m ◆VAS: 1.87 cu. ft. ◆QMS: 1.36 ◆QES: .47 ◆QTS: .34 ◆XMAX: 3 mm ◆Net weight: 2.6 lbs. ◆Dimensions: A: 6-5/8", B: 5-3/4", C: 3-3/8", D: 3-1/2", E: 1-1/2". ***Ask for part #297-307.***

P17SJ-00-08, 6-1/2" Shielded Cast Frame Woofer

The shielded version of the famous P17WJ-00-08, this woofer is perfect for A/V applications. It utilizes a cast magnesium basket, rubber surround, mineral filled polypropylene cone and a very smooth frequency response. Intended for use with vented enclosures with .35 to 1.00 cu. ft. of internal volume.
◆Power handling: 70 watts RMS/100 watts max ◆Voice coil diameter: 1-1/4" ◆Voice coil inductance: .84 mH ◆Nominal impedance: 8 ohms ◆DC resistance: 5.6 ohms ◆Frequency range: 35-5,000 Hz ◆Magnet weight: 25.4 oz. ◆Fs: 41 Hz ◆SPL: 87 dB 1W/1m ◆VAS: 1.16 cu. ft. ◆QMS: 1.28 ◆QES: .48 ◆QTS: .35 ◆XMAX: 4.0 mm ◆Net weight: 3.8 lbs. ◆Dimensions: A: 6-5/8", B: 5-3/4", C: 3-1/2", D: 4", E: 1-3/4". ***Ask for part #297-308.***

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The 1978 C.E.S. in Chicago was the very first time that Morel Acoustics USA, Inc. presented their product to the public. It became clear, early on, that the loudspeaker industry was in need of high quality speaker drivers. Shortly thereafter we introduced several drivers and established the MDT-28/30 as one of the most popular and highly demanded tweeters on the market.

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

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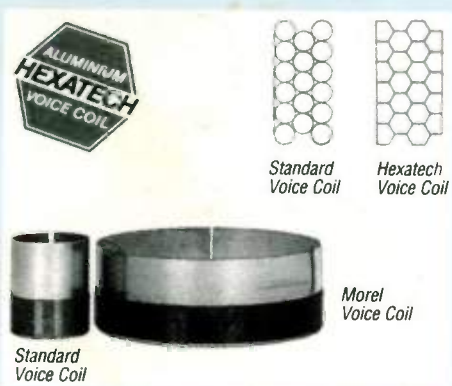

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Double magnet tweeter

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