SPECIAL FOCUS ON SPEAKERS, p. 23–41

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From the Editor's Desk

Focus on Speakers

Each year's "speaker special" issue is a favorite among *audioXpress* members and clients. It features in-depth articles about speaker technologies and design projects, without completely ignoring other interesting topics (e.g., tube amps).

This year's special speaker section features informative articles about a detailed loudspeaker design project and an intriguing electrostatic speaker experiment. We also included a third must-read speaker-related article, but it isn't in the special section because it focuses on enclosures rather than speaker design. Let's review each article.

On page 23, George Ntanavaras shares his Horizon loudspeaker design. Inspired by Siegfried Linkwitz's work, Ntanavaras designed an open-baffle active loudspeaker featuring two woofers connected in series used as dipoles, a woofer/midrange used as a dipole, and a dome tweeter used as a monopole. He details everything from the baffle cabinet design to frequency response adjustments.

Turn to page 36 to read about Richard Mains's excellent electrostatic loudspeaker (ESL) project. Although he started experimenting with hybrid electrostatic speakers about two decades ago, Mains hasn't lost his passion for them. His article covers techniques relating to ESL fabrication, and includes the materials needed to complete your own project.

The third speaker-related article is about enclosures (p. 8). Mike Klasco and Steve Tatarunis cover the practical uses for enclosures, as well as wood alternatives and experimental design techniques.

The rest of the issue's content is a nice mix of past and present audio technology. We feature an interesting turntable redesign project and the second part of a series on tube sound.

In the article "A Vintage Turntable Revisited," Ron Tipton explains recent modifications he made to a 2011 turntable restoration project (p. 12). He describes lengthening the tone arm and why he used carbon fiber tubing.

If you're into tube sound, flip to page 18 for Richard Honeycutt's insight on distortion. He covers topics such as distortion signatures and testing.

Regards,

C. J. Abate
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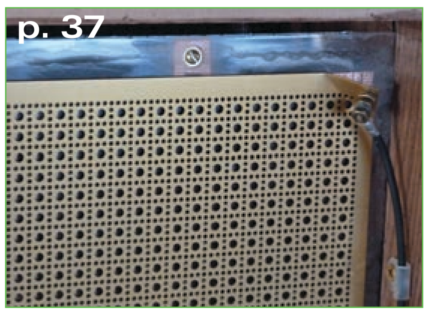

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By Mike Klasco and Steve Tatarunis (United States)

Speaker Enclosures

Selecting the right cabinet prevents unwanted resonance

Our Speakers—"Parts is Parts" series has focused on various speaker components. Now, let's take a break and look at the enclosure. Assuming you have already selected the drivers for your latest project, your attention is most likely focusing on the enclosure. No matter how much effort was spent on the drivers' design, the enclosure characteristics are still a significant factor in achieving smooth response, controlled bass, a defined stereo image, clarity, and definition.

PRACTICAL USES

Speaker enclosures are mainly used to isolate the backwave of the woofer from the front. This is needed to avoid the two equal but out-of-phase outputs from canceling each other. Wood cabinets are traditionally used for speaker enclosures. Other materials have been tried, but they have not been generally accepted. Enclosures need to provide a rigid structure to support the woofer with minimum panel resonances. Most boxes are simply medium-density fiberboard (MDF) that are vinyl covered and "groove folded" out of a larger sheet. The grove-folding process cuts a V-groove into the wood, down to, but not through, the vinyl covering. A bead of glue is dispensed into the V-groove and the enclosure is folded into a box shape (see **Photo 1**).

Conventional particle-board boxes are inexpensive, readily available, and a known commodity. But, wood boxes have intrinsic flaws. The sharp baffle edges diffract sound where it is not wanted. The flat baffle with the woofer and tweeter directly mounted means the acoustic centers are most likely not aligned, and perhaps most importantly, your box will look just like the other dozen models on the sales floor. How can you blame the customer for assuming little thought was spent on engineering if the exterior appearance gives this impression?

Photo 1: This loudspeaker enclosure shows a basic MDF construction. (Photo courtesy of Wikipedia)

WOOD ALTERNATIVES

Over the years, there has been great interest in finding an alternative to wood enclosures. The material should be as strong and light as plywood, as easy to work with as particle board, and have a simplified construction (i.e., moldable materials that eliminate joint assembly and enable curved edges and shaped front baffles for time alignment, waveguides, and esthetic flexibility).

Dramatic enclosure styling that combines unique materials and forms takes courage. But, the rewards can be great. Bose launched the five-sided 901 in 1968. B&W's distinctive Nautilus 800 series was a breakthrough in the late 1990s. And, today's B&O Beolab 5 and the B&W Zeppelin Air continue to offer innovative enclosure shapes.

The usual injection-molded plastic thin-wall construction found in outdoor speakers or low-cost 5.1 systems is marginally adequate for commodity consumer speakers. Most speaker engineers are unhappy with the hollow, buzzy sound that comes from funky, plastic home-theater satellite speakers. From the most inexpensive styrene to filled acrylonitrile butadiene styrene (ABS) to talc-filled poly, plastic enclosures have high levels of breakthrough (the driver's backwave passing through the shell and excessive bass loss because of the lack of rigidity). When driven hard, these plastics contribute too much of their own noises, further degrading sound quality. None of this is news to speaker designers. And, audio engineers have been looking to glass-fiber, Kevlar, and carbon fiber composites for rigid and well-damped enclosure parts such as Klipsch's outdoor speakers (see **Photo 2**).

Some technologies give designers more flexibility to achieve aesthetic and technical goals. Enclosures should suppress vibrations that ripple through them and provide higher sound transmission loss (a measure of how good a barrier the enclosure is to prevent sound from passing through it). Higher stiffness is needed to drive resonances into higher frequencies where they are more easily controlled. At various times, Polk, JBL, and others have offered flagship products with laminated, damped baffles using two reduced-thickness layers of wood or particle board with a damping layer of adhesive in between.

Another consideration is reducing edge diffraction with curved edges. The curved corners result in reduced dents and rework during manufacturing and shipping. Response irregularities contributed from sharp front cabinet edges can result in peaks and dips of up to 4 dB. Audiophile speaker

Photo 2: This is an example of a Klipsch outdoor speaker with an ABS cabinet. (Photo courtesy of Klipsch)

systems in Japan, from Sony and others, and world-class recording studio monitors from Genelec, use rounded edges to maintain the accuracy of their precision drivers (see **Photo 3**). Sharp edges cause diffraction, which means the edge of the cabinet acts as a second sound source, destructively interfering with the original signal. Curved edges are made through direct molding, cutting of the relief on the back edge of rigid materials, or routing the edge.

To achieve an optimum engineered design for a more precise or "focused" soundstage, consider mechanically aligning the drivers' acoustic centers, which is especially beneficial in the vicinity of the crossover frequency. The molding of horn/waveguides can control directivity, increase sensitivity, and provide for the time alignment of the acoustic centers, as the woofer tends to lag the tweeter.

EXPERIMENTAL TECHNIQUES

Over the years, numerous techniques were introduced to revolutionize the speaker enclosure business, but not all of them are practical. In the mid-1990s, Cubicon used recycled paper and fabricated enclosure tubes with fancy finishes (as well as carpet cover, etc.). These tubes were square, round, D-shaped (autosound), trapezoid (disco mobile), and so forth. The multilayered paper/resin composite offered a well-damped material for enclosures and the radiused edges minimized diffraction effects. But, the tube shapes were not consistent enough for mass production. End cap fittings were problematic, although a few firms managed to devise workable production techniques. In any case, while recycled materials are "in," you won't see many speaker enclosures made from recycled cardboard tubes these days, although sound bars could benefit from this approach.

Curved panels, sculpted shapes, and all sorts of stuff that usual enclosure suppliers cannot provide, could be laser precut and shipped to the speaker factory's enclosure shop for assembly. Advantage Cutting and Gasket had big plans but closed down about 10 years ago, as speaker-enclosure assembly shifted offshore.

Another technology that did not gain acceptance, at least for speaker enclosures, was Gridcore. Thin, but light, stiff panels made of inexpensive molded honeycomb wood fiber enabled curved panels and thin-wall construction. The material originally developed for interior door panels is also used by the military for emergency shelters and other structures because it is light enough to be easily transported (see **Photo 4**).

A more sophisticated honeycomb technique was used in Celestion's SL600. Introduced in the mid-1980s, the speaker enclosure used a rigid honeycomb alloy called Aerolam instead of wood. Honeycomb had its advantages, but attaching drivers, joining panels, and sound breakthrough were problems not easily or inexpensively resolved.

Nu-Stone, a bulk-molded compound (BMC) composite (compressed of glass fiber, talc, mica, etc.) can be molded into enclosures. BMCs are typically compression-molded, so the forming time was 10 minutes or more. BMC offers great acoustic performance, in part because of the high-density material, easily molded ribs, radiused edges, and the baffle. Unfortunately, the thicker walls and denseness increased the cost per unit. Sharp edges chipped and high aspect-ratio details like internal posts had poor production yields. Polk and NHT, among others successfully used this material (see **Photo 5**). These days, cost-sensitive products (e.g., docking stations) cannot defend the higher cost of goods versus inexpensive ABS and other commodity injection-molded, thin-wall plastics. Industrial Dielectrics, which develop Nustone, still makes huge quantities of BMC materials, and production has expanded to China. Globe Plastics is a vendor of BMC-molded speaker enclosures, horns, and other parts, and its products are established in the market. BMC has also found a home for outdoor and higher grade products (e.g., automotive body parts, electrical boxes, etc.). From these applications, it is apparent the material has extreme temperature tolerance and is weather resistant enough for speakers permanently installed outdoors.

Photo 3: The Genelec 8260a monitor utilizes a more engineered design. (Photo courtesy of Genelec)

POWDER–COATED WOOD

Powder-coated paints for wood enclosures have been under development

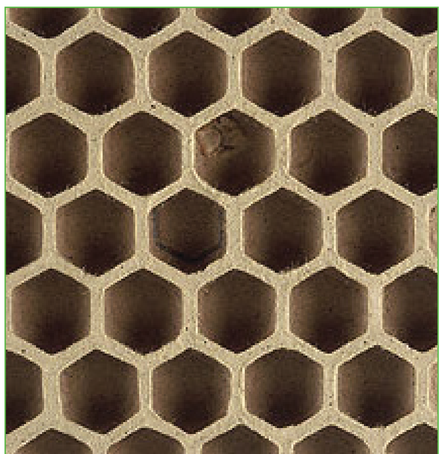

Photo 4: The Gridcore panel detail helps decrease the enclosure's weight. (Photo courtesy of Gridcore Products)

and refinement for more than a decade. Until the last decade, powder coating, which provides a strong, attractive finish, was only possible for metal parts. UV-curable powder coating or low-temperature curable powder coatings are now common for wood furniture. When used on wood, it is possible to get a "show car" or lacquer paint-like finish on a speaker enclosure. Success on wood components usually occurs when a single piece is powder coated. Complete boxes do not lend themselves to straightforward powder coating. Early processes tended to char the wood and over time, cracking resulted, particularly at the joints.

HIGH-QUALITY LAMINATE ENCLOSURES DISAPPEAR

From the 1950s through the 1960s, audiophile speaker enclosures were handcrafted wood and wood veneers. By the 1970s, the cost of conventional domestic woodworking and the popular walnut veneers were too high for mainstream consumer audio products.

Photo 5: These Polk RM6600 speakers with Nu-Stone enclosures use BMC materials. (Photo courtesy of Polk Audio)

Vinyl-wrapped particle board dominated audio stores' shelves. In the late 1990s, high-end wood grain and black lacquered speakers reappeared because of quality wood speaker enclosure factories in China. Now, most of the Chinese woodworking shops have done one of three things: expanded into other businesses (e.g., furniture for luxury hotels and retail store display furniture for high-grade products); found other endeavors; or closed. Several factors have driven these firms away from high-end speaker enclosure work (e.g., the atrophying of the global audiophile market, wage costs for such labor-intensive and time-consuming work (in some shops the multilayer lacquer finishing process took almost a week), and loss of interest in such tough and arduous jobs.

WHAT'S NEXT?

If cost of goods was not such a sensitive issue, BMC enclosures for docking stations, sound bars, and related products would be ideal. While some BMC formulations can be injection molded, this is a special process not mastered by most offshore factories. ABS plastic resins dominate, but they have mediocre acoustical characteristics. Filled ABS requires painting, and the surface finish can be an issue. Higher-grade resins like polyetherimide (PEI) are extremely costly and the injection-molding temperatures require expensive special steel tools. An interesting upgrade from the commodity ABS plastics is compounding wood flour with plastics. Wood flour is finely milled saw dust with a sizing (coating) and, when mixed with various plastics at about 30%, imparts a somewhat wood-like acoustic characteristic. Wood-flour plastic composites are injection molded. While the raw material costs about the same as ABS, the process costs a bit more. The process requires humidity control of the materials and humidity control at the injection molding area. While this is common in the U.S., humidity control is rare in China. Also, parts molded with compounded wood-flour require a primer and a finish-paint coat.

CURRENT STATUS

MDF boxes with vinyl wrap or veneered wood surfaces continue to dominate the home theater and stereo audiophile enclosures. Unfortunately, these product categories have shrunk over the last few years. Quality speaker systems, considered the realm of audiophiles in the 1960s, then became mainstream stereo in the 1970s and 1980s. Home theater was mainstream in the 1990s. Today, we are faced with sound bars often made of plastic construction, sometimes with a MDF baffle inside. Perhaps large-screen TVs provide enough distraction from sound quality. Docking stations and inwall and ceiling speakers are typically injection-molded plastic. In general, current sound reproduction enclosures leave a lot of room for improvement, with the exception of increasing obscure higher-end brands.

As some manufacturing comes back to the U.S. many factors could affect "tuning" products.

Next month, we will think "inside the box" and discuss box stuffing. *aX*

By Ron Tipton (United States)

A Vintage Turntable Revisited

Additional modifications improve classic equipment

In my original article, “Restoration of a Vintage Turntable” (*audioXpress*, February 2011), I described the restoration of a Netronics 350D turntable from the 1960s. It had a direct-drive DC motor with a 5-lb cast aluminum platter, two speeds (33-1/3 and 45 RPM), and a ±5% speed control. It also had a terrible suspension system, which I improved during my initial restoration. (Netronics has long been out of business, but it was a good turntable for its time and well worth restoring.)

I rebuilt it for two reasons. First, I wanted to show that making the tone arm longer reduces the tracking angle error. Second, I wanted to experiment with using a fairly new material, carbon fiber tubing, to construct the new 20” arm. That is, 20” from the arm pivot point to the stylus tip. The base unit measures 28.75 × 18.5” to accommodate the longer arm (see **Photo 1**).

TRACKING ERRORS

In the 1950s, Dr. John D. Seagrave wrote two articles on minimizing pickup tracking errors. They were published in the December 1956 and January 1957 issues of *Audiocraft* magazine. Fortunately for most of us, they were reprinted in *The LP Is Back* (Audio Amateur Press, 2000). For those interested in the information, the book is available from the Circuit Cellar CC-Webshop at www.cc-webshop.com. My tracking error analysis is based on the information in these articles.

A 12” LP record has an outside radius, R_1, of a maximum of 6” and a minimum inside radius, R_2, of 2”.

If ϕ is the tracking angle, D is the overhang (i.e., the stylus distance beyond the turntable center), and L is the arm length (20”) then:

$$\sin\phi = \frac{R}{2L} + \frac{D}{R\left(1 - \frac{D}{2L}\right)}$$

$$\text{if } D = 0,\ \sin\phi = \frac{R}{2L}$$

$$\sin\phi_1 = \frac{R_1}{2L} = \frac{6}{40} = 0.15 \text{ and } \sin\phi_2 = 0.05$$

ϕ_1 = arcsin (0.15) = 8.63° and ϕ_2 = arcsin (0.05) = 2.87° tracking error $\alpha = \phi - \beta$ where β is the cartridge offset angle.

If β is set to about 6° (easy to do by adjusting the cartridge angle in the headshell) then α_1 is about 2.6° and α_2 is about −3.1°.

For a more common arm length of 10” (228 mm), α_1 is about 5.5° and α_2 is about −6.3° for a β of 12°. I think this clearly shows the advantage of using a longer than usual tone arm. Also, many 12” LP = s have an inside radius closer to 2.5”, so α_2 is smaller than −3.1°. However, to halve the tracking error again requires an L of 40”, which is probably impractical.

CARBON-FIBER TUBING

This is a composite material composed of carbon fiber and epoxy. It’s available in three basic ”flavors”: longitudinal fibers, longitudinal plus wrapped fibers, and woven fibers (a carbon fiber mat). It has a density of 68% of 6063 aluminum with a stiffness-to-weight ratio of nearly five times that of aluminum. It should be excellent tone arm material. The CarbonFiberTubeShop website provides considerable information, but it doesn’t have a 0.375” outside diameter (OD) tube, which I wanted for the arm. I found it (0.375” OD × 0.290” ID) at DragonPlate (www.dragonplate.com) but it only carries “pultruded” tubes (in this size), which have longitudinal fibers. This manufacturing process produces inexpensive tubes, but they have low torsional strength, which makes them difficult to use. Machining with a lathe is a torsional operation and tubes tend to shatter. But, the longitudinal fiber tubes are a bit stiffer than the others so it’s a trade-off.

I built the machining aids shown in **Photo 2,** which helped. The split wooden tube with the center hole applies nearly even pressure around the tube circumference when it’s held in the lathe chuck. Turning the wooden rod to fit helps stabilize the inside diameter. Even with these aids, and taking very light cuts of only a few thousandths of an inch, the tube still split. However, I was able to cut the tubing on the lathe with a fine-tooth hacksaw blade and then squarely face the cut end.

I solved my construction problem using plastic collars to couple the carbon-fiber tube to the gimbal tube and to the pickup cartridge connector (see **Photo 3** and **Photo 4**). I will have more to say about this in the Tone Arm Construction section.

Photo 1: This is the completed second Netronics 350D rebuild. The only remaining original parts are the direct-drive DC motor, the platter, the aluminum nameplate with switches, and the 33-1/3 and 45 RPM speed control.

Photo 2: These are key carbon-fiber tube machining aids. The split hardwood tube, about 2.5" long, spreads the pressure on the tube circumference when held in the lathe chuck. The inside wooden rod stabilizes the tube for cutting.

Photo 3: Epoxy connects the gimbal tube to the carbon fiber tube with a 1.5"-long acrylic collar. Masking tape is used to hold the small gauge solder wires that go through the gimbal.

THE BASE

At 28.75” × 18.5”, the base unit is admittedly large, but that’s the price of using a 20” arm. I used 0.5”-thick MDF with 4”-width sides for rigidity. I used only glue and wooden dowels for assembly, so I needed 36” glue clamps. This wasn’t a problem because I’ve built quite a few speaker enclosures. Three coats of clear, satin polyurethane varnish completed this part of the project.

I had good results in the original restoration with suspending the motorboard at each corner on a stack of art gum erasers. So, I reused the idea. I used rubber cement to lightly glue the erasers together, and to the wooden supports (see **Photo 5**). They are always under compression, so there isn’t a tendency for them to wander. The four wooden blocks, made from scraps of 1.1” thick walnut, were sized so that one-third of the motorboard thickness was inside the base unit. The turntable’s four feet were originally salvaged from a defunct Stanton turntable. I also used them in my original restoration (see **Photo 6**).

THE MOTORBOARD

The motorboard is a 27.25” × 17” sheet of 0.75” thick, 11-ply Baltic birch plywood finished with a contrasting stain and three coats of the satin polyurethane varnish. All parts, including the direct-drive DC motor, are mounted on the motorboard, which “floats” on the art gum erasers in the base unit. The board was sized so there is about a 0.25” clearance from each side of the base unit.

Photo 7 shows the underside of the completed motorboard. Although the birch plywood is rather rigid, I added the diagonal brace to eliminate the possibility of “sag” over time. Brass hardware is used to attach the 0.125” × 1.5” × 1.5” aluminum angle to the board in five places. I reused the original Netronics nameplate with its switches and speed control, but I didn’t recess it into the motorboard as was done in the original design—again for better rigidity.

The loop of wire at the upper left corner is the shielded signal cable from the tone arm to the cast-aluminum connector box on the top side of the board. In the original restoration, I used an aluminum bracket on the underside of the motorboard for the connectors, but that required a cutout in the rear base unit side wall. I didn’t want to do that this time because it may have impaired the rigidity of the longer side wall.

TONE ARM CONSTRUCTION

Although it added some additional weight to the tone arm, I used clear acrylic collars to mate the carbon-fiber tube at each end. I bought a length of round acrylic tube (0.5” OD × 0.25” ID) from McMaster-Carr and

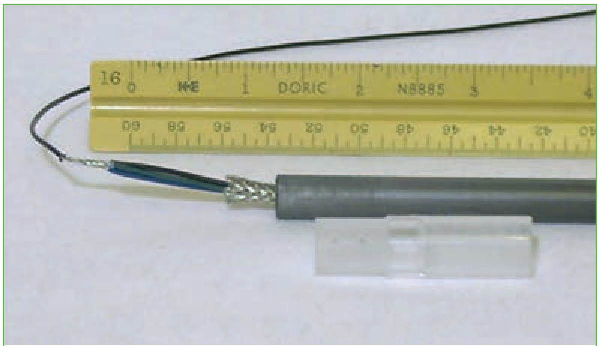

Photo 4: I used wires that were bundled and soldered to another wire to pull the wires through the copper-braid shell. The shielded wires were then pulled through the tube.

Photo 5: One corner of the base unit with a wooden block and stack of three art gum erasers was used to support the motor board. Rubber cement was used to lightly connect the erasers. The small circle in front of the block is an embedded nut the foot screws into.

machined the required collars. Acrylic is quite easy to machine. The collars also increase the torsional strength of the tone arm by providing strength at the carbon-fiber tube ends, where cracking would begin. (This is probably not an issue in this light-duty application.)

The machining must be carefully done with only a few thousandths of an inch of clearance between the mating parts because the arm must be as straight as possible. Fractional and “letter” size drills will get you close, but the final cut or cuts will need a boring bar. In addition, a good micrometer or caliper is needed to measure the actual diameter of the carbon-fiber tube, headshell connector, and gimbal tube. For example, the carbon-fiber tube I used has an OD manufacturing tolerance of ±0.010”. Collar length is not critical, but a good compromise between weight and fit is about 0.75” between the mating parts (about 1.5” overall). Slightly roughen the carbon-fiber tube ends with fine sandpaper and use rubbing alcohol to clean off your fingerprints or other residue. The machined collar surfaces are rough enough, just clean them too with a little alcohol. Spread a thin layer of epoxy and push the parts together with a twist to ensure a good joint. Use a clean cloth to wipe off any seepage. I used Devcon Clear 2-Ton 2-Part epoxy, which is available at many hardware stores, but I doubt that it’s critical.

Photo 4 shows the shielded bundle of five wires that will go to the headshell connector, a standard 0.5” mount that accepts dozens of headshells. (**Photo 1** shows a Sumiko HS12 Azimuth adjustable model.) The acrylic collar will be epoxied to the carbon fiber tube. After the wires are soldered to the headshell connector, it will slide into the other end of the collar and be secured with two 1-72 × 0.125” machine screws. Unfortunately, all the headshell connectors I’ve seen have only four solder terminals: signal and common for left and right. So I had to improvise to connect the ground wire.

I filed a shallow groove in the side of the brass connector wall, soldered the ground wire in the groove, and filed the solder joint smooth so it would slide into the collar.

Photo 3 shows the gimbal end with the collar already epoxied in place. Care is needed when applying the epoxy because you don’t want to cement the wires to the inside of the arm—they need to slide freely. All of the signal wiring is shielded 28-AWG stranded wire except for about 3” of very small-gauge solid wire that goes through the gimbal. Thus, there are several wire splices, each insulated with shrink tubing and covered by

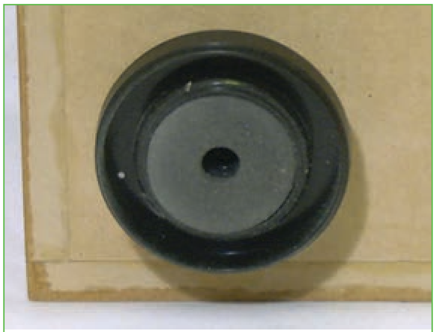

Photo 6: One corner of the base unit shows a foot salvaged from a defunct Stanton turntable.

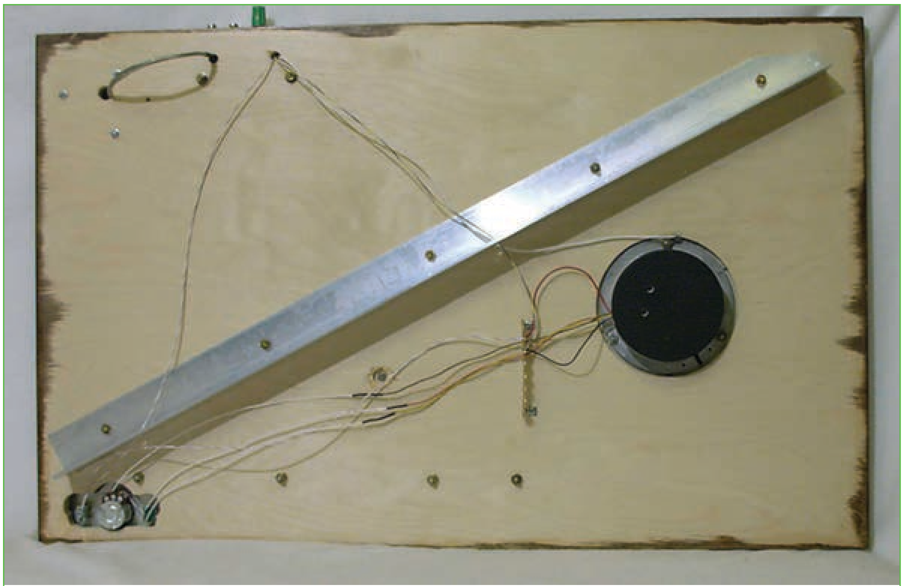

Photo 7: A router and circle-cutting template were used to cut the stepped hole for the direct-drive DC motor.

the shield. The shields are only connected at one end to the continuous ground wire that runs from the headshell connector to the RCA output connectors (see **Figure 1**).

The tin-plated, copper-braid shield is easy to retrieve from a length of RG-58A/U coaxial cable. Just slit the outer covering for a few inches at one end, fold the slit covering away from the shield, and pull. The covering easily strips away from the shield. Then use both hands to "pucker" the shield from one end to the other and the center conductor will easily slide out. The puckered shield will enable you to pull the new wiring through—it's nearly impossible to push it through!

The gimbal was salvaged from the Stanton turntable that contributed its feet. (The whole tone arm was used in my original restoration.) With the longer arm, the counterweight size had to be increased. With the arm level, I weighed the cartridge end of the arm with a digital postage scale. The 45 grams at 20" needed to be balanced by a new counterweight at 3.25" (277 grams). The original counterweight is 85 grams so I needed to add about 192 grams. Using 8.5 grams/cm^3 as the average weight of yellow brass, I calculated the size needed and built it from a length of 1"-diameter brass rod. It attaches to the original counterweight with two set screws at right angles to each other through the sides of the original weight (see **Photo 8**).

With an average-depth pickup cartridge, the arm is level when playing a record so the vertical tracking angle (VTA) is the best you can do without a height-adjustable gimbal. The azimuth angle is easily set with an adjustable angle headshell (such as the Sumiko HS12). The overhang was set to zero and the cartridge offset angle was set to about 6° (in the headshell) as this minimized the tracking error to about 3° at each end of the playing arc.

SOUND QUALITY

Does the longer arm sound better? That's hard to answer. It certainly

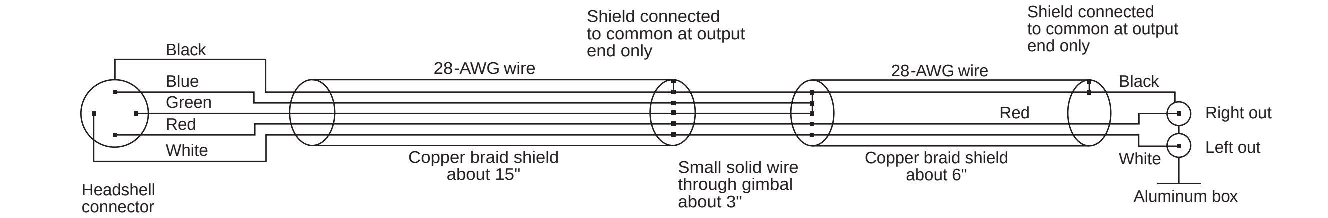

Figure 1: This is the 20" tone arm wiring. The shields are floating and do not carry any signal current from the cartridge. The small-gauge solid wire through the gimbal was salvaged from the original Stanton tone arm.

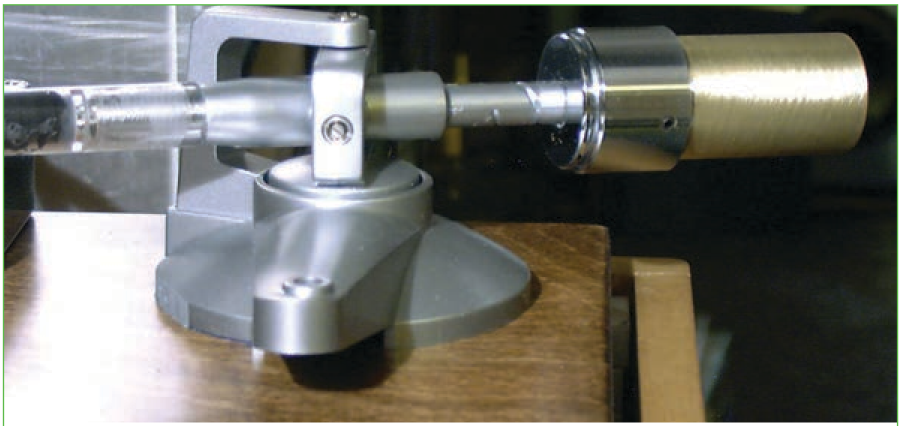

Photo 8: The gimbal salvaged from the Stanton turntable was rebuilt with a heavier counterweight. The added weight slides into the rear of the original weight and is secured by two set screws The original calibrated dial was removed because it's not accurate with the weight increase. A gauge must be used to set the stylus force.

sounds fine and the tracking error has been cut in half. The sound quality also depends on the cartridge and the rest of the audio system, but I am satisfied that this turntable is as good as it can be given the practical limitations of its components. I suspect commercial versions this size aren't produced because of the size. But I have a solid wooden desk right next to the rack that contains the audio system, so its size is not a problem. I think this article includes enough detail that many turntables could be rebuilt with a longer arm if you have the inclination and tools to take on the project. *aX*

By Richard Honeycutt (United States)

The Tube Sound (Part 2)

The Distortion of Different Types of Amps

A look at the aural differences arising from distortion

In my last article, I examined the likelihood that many, if not most, of the perceptible differences in various amplifiers' sounds result from different distortion signatures. In an earlier article, I briefly discussed the distortion differences between pentode and triode amplifiers. This time, I'll look at the details of these differences. In a future article, I'll examine the likelihood of perceptible differences when amplifiers are operated well below the threshold of perceptible distortion.

Many design differences can be found among commercial audio amplifiers. Power amplifiers are the most common. These can use single-ended triodes or pentodes, push-pull triodes or pentodes, push-pull bipolar transistors, or push-pull junction field effect transistors (JFETs). (Bipolar transistors and JFETs are rarely used in the single-ended configuration.) There are also the pre-amp/voltage amplifier stages and the phase splitters (in push-pull amps). Depending on the specific amplifier design, the threshold of clipping or other nonlinear distortion can occur in any of these stages. In most hollow-state amplifiers, the phase splitter is responsible for the earliest clipping, although in some instrument amplifiers (e.g., guitar), one of the pre-amp stages is designed to clip first. In transistor amplifiers, the design used to provide DC stability causes the entire amplifier to remain linear up to the clipping threshold, at which point clipping occurs pretty much everywhere at once. Also, the short-circuit/overcurrent protection circuits in solid-state amplifiers often determine the clipping point.

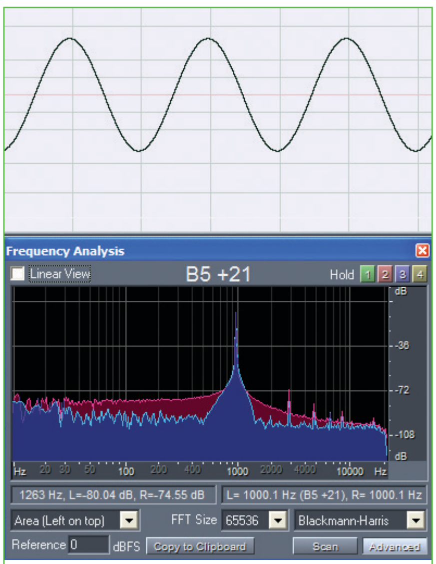

Figure 1: This test shows the the NTI signal generator's 1,000-Hz sine-wave output.

DISTORTION SIGNATURES

To provide real-world examples of distortion signatures without testing dozens of amplifiers, I tested a two-stage triode pre-amp, a single-stage bipolar transistor amplifier, a commercial bipolar transistor power amplifier, a commercial push-pull pentode amplifier (using a pentode pre-amp stage and push-pull 6L6GC pentode output tubes), and a McIntosh MI-75 power amplifier using a triode pre-amp and push-pull 6550 pentode output tubes. I tested each amplifier on my Tektronix oscilloscope, using a lab-grade, sine-wave generator, with the input set 1 dB below the onset of visible distortion in the amplifier's output wave.

Figure 1 shows the 1,000-Hz sine-wave output from the NTI signal generator. The fundamental level is about –8 dB. The third and fifth harmonics at about –72 and –84 dB, respectively, are predominantly from the sound card in the test computer. The widening of the 1,000-Hz trace at lower levels represents phase jitter in the digitally generated sine wave, and is almost 40 dB below the fundamental (about 1%). **Figure 2** shows the distortion spectrum of a two-stage triode amplifier. There is a noticeable 60-Hz hum component probably caused by amplifier layout. The fundamental level is about –8 dBV, but this time there is a significant second harmonic component about 26 dB lower (roughly 5%), along with a third harmonic 60 dB below the fundamental and a fourth harmonic at about –68 dB.

Figure 3 shows the spectrum of a bipolar transistor pre-amp. Here, some 60-Hz hum also shows up, due to induction from power lines. Again, the fundamental is about –8 dBV, and there is a significant second harmonic component at about –36 dBV (roughly 1.5%), plus third and fourth harmonics monotonically decreasing as the harmonic number increases. Notice that there is less second harmonic in the transistor amplifier than in the triode amplifier, and that the transistor amplifier's harmonics extend to the eighth; whereas the triode's harmonics essentially disappear after the fourth.

Figure 4 shows the distortion spectrum of a commercial bipolar transistor power amplifier. This amplifier incorporates a high degree of negative feedback. So, although the total harmonic distortion at 1 dB below visible clipping is only about 0.1%, the harmonics' amplitudes

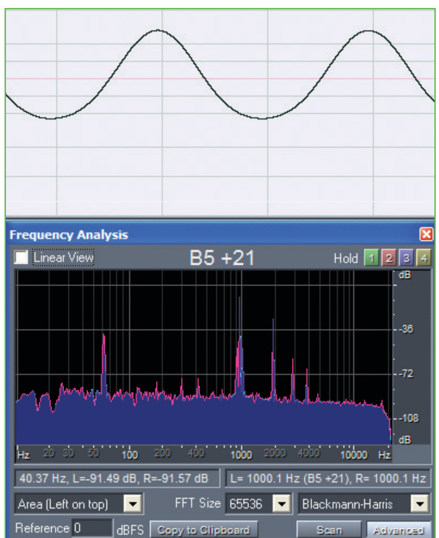

Figure 2: A two-stage triode amplifier's distortion spectrum shows the second harmonic is much larger than the third.

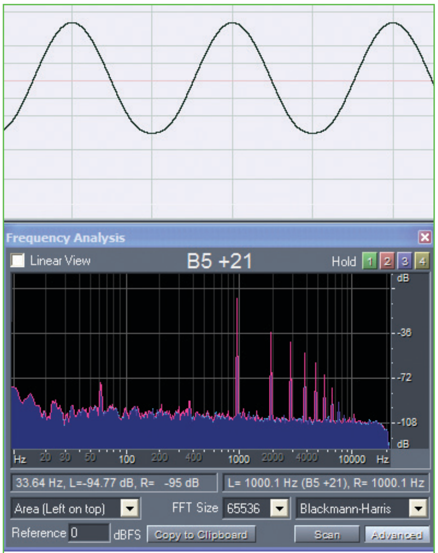

Figure 3: The bipolar transistor pre-amp shows significant second harmonic, but it's still less than in the triode pre-amp.

(both odd- and even-order) are essentially constant up to about the 18th.

Figure 5 shows this same amplifier driven into hard clipping. The input level was only about 2 dB greater than for the test shown in **Figure 4**. But as shown in the waveform, the sine wave is symmetrically clipped, and the spectrum shows the same rich array of harmonics extending pretty much to the limits of audibility.

The two remaining amplifiers are both hollow-state models that use similar push-pull pentode output stages. The first is a 50-W commercial public-address amplifier with a pentode input stage. The second is designed as a studio monitor amplifier with a 75-W push-pull pentode output stage, cascaded triode pre-amps, and a phase splitter.

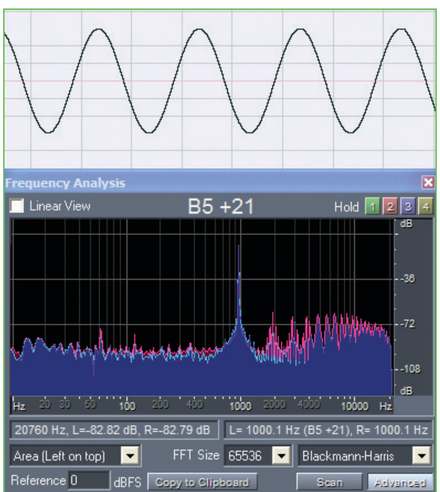

Figure 4: This bipolar transistor amplifier shows small harmonic amplitudes but they do not decrease at higher frequencies.

Figure 6 shows the distortion spectrum of the PA amplifier. Notice both 60-Hz and 120-Hz components, which result from the voltage-doubler power supply, whose filtering and voltage regulation weaken at high-power outputs. The fundamental is about –8 dBV, and the second harmonic is about 40 dB lower, so the total harmonic distortion under these test conditions is a bit more than 1%. Significant harmonics extend up to the 11th, with about a 14-dB dropoff from the second to the 11th, and a more rapid dropoff of the lower-amplitude harmonics above the 11th. The presence of significant even-order harmonics even with a push-pull output stage indicates clipping is not occurring mainly in the output stage, but rather in pre-amp and/or phase splitter stages.

Figure 7 shows the distortion spectrum of a McIntosh MI-75 studio monitor power amplifier with a push-pull pentode output stage and triode preamp and phase splitter stages. The spectrum is not very different from that of the commercial PA amplifier, except that harmonics from the fifth to the 11th are about 12 dB lower in the McIntosh. Notice the absence of line frequency-related hum components (60 Hz and multiples), resulting from the excellent power supply design and chassis layout.

DISTORTION RESULTS

Remember from last month's article that lower levels of higher harmonics provide a less irritating sound and probably less perceptibility of distortion. So, we can surmise the worst sounding amplifier at or near clipping is the commercial bipolar transistor power amplifier, assuming that it is actually driven into clipping. However, at the 1-dB below clipping level, its distortion is extremely low—roughly one-tenth that of the McIntosh amplifier. This sudden onset of distortion is a characteristic of amplifiers using a lot of negative feedback.

The two-stage triode amplifier and the bipolar transistor pre-amp share a similar architecture, in which an unbypassed cathode or emitter resistor provides the only negative feedback. These two amplifiers have the lowest levels of upper harmonics, so we would expect their roughly 5% and 1.5% distortion levels (respectively) to be less objectionable than the

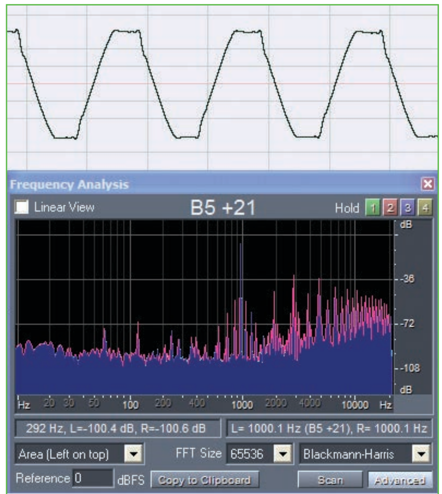

Figure 5: The spectrum shape stays the same when the amp is driven into hard clipping, but all the harmonics' amplitudes are increased.

somewhat lower total distortion levels of the push-pull pentode amplifiers. In spite of the higher distortion in the triode pre-amp compared to the transistor amplifier, many recording studio engineers prefer triode microphone pre-amps, insisting that they sound better. As you can see by carefully examining the waveforms of these two amplifiers, the triode amplifier begins to visibly compress the negative half-cycle before the positive half-cycle is affected. I should mention no great care was taken to try to match the dynamic characteristics of the bipolar transistor pre-amp and the triode pre-amp, so this difference is not always as great as appears in this case. However, the triode provides some gentle amplitude compression before clipping

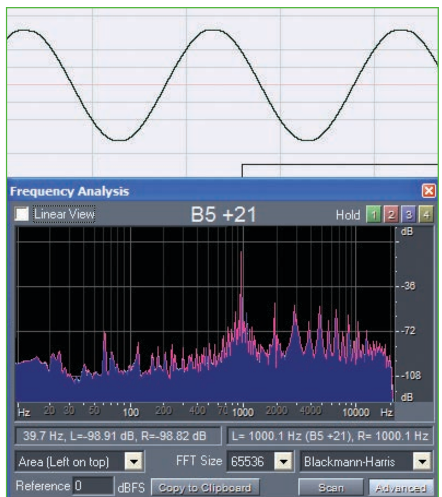

Figure 6: The pentode push-pull PA amplifier shows significant even- and odd-order harmonics up to the 11th.

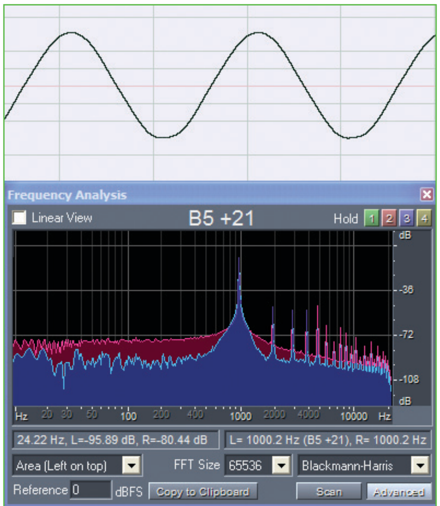

Figure 7: The McIntosh MI-75 pentode push-pull power amplifier displays lower amplitudes of harmonics 5 through 11.

produces truly objectionable distortion. Given that a condenser recording microphone may put out about 7 mV when picking up a vocalist—or perhaps 440 mV or more when picking up a trumpet—a microphone pre-amp must handle a wide range of signal voltages, while at the same time producing a very low output noise. Under these conditions a bit of compression before clipping can help, regardless of the characteristic sound of the distortion components.

It would not be accurate to assume that the performance seen in the triode pre-amp can be extrapolated to a triode power amplifier, since the latter will have an output transformer that can add some distortion, and since the load line (collection of all plate currents and their corresponding plate voltages) will not necessarily be very similar for a pre-amp and a power amplifier. Many audiophiles prefer triode power amplifiers, but I did not have a commercial model available for this set of tests. In the absence of tests, we can say triode amplifiers can provide gentler clipping (depending on the degree of negative feedback), and thus a less irritating sound when overdriven, than other types of amplifiers.

In this article series, we have focused on the differences in the distortion signatures of different amplifiers. We have not paid any attention to other significant performance characteristics, such as frequency response, power-supply stiffness (voltage regulation of the power supply under varying amplifier output conditions), and hum and noise. All of these characteristics can be controlled by carefully designing of any kind of amplifier. Thus, none of them specifically relates to any "tube sound," although they are important in the evaluation of amplifier performance.

FACTOR POSSIBILITIES

Damping is one possible factor that is specific to hollow-state amplifiers. In electrical theory, there is a concept called "an ideal voltage source." This is a source whose output voltage is unaffected by the load resistance connected to it (and thus, by the output current drawn from it). There are no true ideal voltage sources, but some come closer than others. The difference is in the "output resistance" (or more generally, "output impedance") of the source. For example, as an old-technology carbon-zinc dry cell ages, it develops an internal resistance to current flow. This internal resistance will cause the output voltage to be lower when more output current is drawn from the cell. Likewise, internal resistance in an amplifier (usually called "output resistance" or "output impedance") will cause the output voltage to decrease as the load current increases when a lower-resistance load is connected. In either case, this internal resistance acts as if there was a discrete resistor inside the source (cell or amplifier), connected in series with the load. Although this sounds like a problem with the available output voltage, that is not really the important issue. A speaker system acts as a complex circuit containing inductance, capacitance, and resistance. When it is driven by a pulse (drum stroke), it will act as a sort of resonant circuit and the speaker cone will ring at the resonant frequency, smearing the impulse response. The duration of this ringing depends on the resistance in series with the speaker. In fact, the speaker voice coil ringing in the magnetic field acts as an AC generator. A low resistance in series with this generator enables a lot of current to flow in the voice coil, damping the motion. A higher resistance provides less damping. The load impedance of the speaker (e.g., 8 Ω) divided by the resistance in series with the amplifier output, is called the "damping factor." Thus, an amplifier with a high output resistance that provides little damping is said to have a low damping factor.

Unfortunately, the damping factor is

sometimes defined as the speaker impedance divided by the amplifier's output resistance. However, in a series circuit, it does not matter where a resistor is located in the circuit. Any series resistor behaves in the same way as any other. Thus, the voice coil resistance of the speaker is also calculated into the damping factor. So an amplifier with a 1-Ω output resistance driving an 8-Ω speaker, whose DC voice coil resistance is 6 Ω will have a damping factor of:

$$\frac{8\,\Omega}{(1\,\Omega + 6\,\Omega)} = 1.14$$

Amplifiers with little negative feedback tend to have low damping factors, depending upon the plate resistance, the output transformer turns ratio, and the output transformer's secondary resistance. Speaker systems with low internal damping tend to have poor transient response. This is particularly true of some of the older bass-reflex speakers that tended to be somewhat boomy. When these speakers are driven by amplifiers with a high damping factors, the boominess is controlled, to some extent. But when these speakers are driven by amplifiers with low damping factors, boominess will be emphasized. This may not necessarily be undesirable, as some listeners prefer an over-emphasized bass sound. Thus, depending on the speakers used, some listeners may prefer a hollow-state amplifier with a low damping factor for reasons having nothing to do with the relative presence, absence, or nature of the distortion. By the way, as was so well explained by Howard Tremaine in the second edition of *The Audio Cyclopedia*, there is no benefit in trying to make a damping factor higher than about 1.23 (amplifier output resistance of 0.5 Ω and voice coil resistance of 6 Ω), because, above that value, voice coil resistance almost completely dominates circuit behavior.

MAGICAL "X" FACTOR

In the next article, I'll take a look at the careful studies that have been done on the difference in amplifier sound for home-theater and home-music listening under unclipped conditions, to see whether any "magical factor X" has been found to explain differences in listener perceptions of pentodes versus triodes versus bipolar transistors versus JFETs. *aX*

The Horizon Loudspeaker

A loudspeaker design inspired by Siegfried-Linkwitz's work

By George Ntanavaras

The Horizon loudspeaker is an open-baffle, three-way, active loudspeaker consisting of four units: two woofers connected in series and used as dipoles; one woofer/midrange which is also used as dipole; and one dome tweeter, which is used as monopole (see **Photo 1**).

The loudspeaker was inspired by Siegfried-Linkwitz's work on open-baffle loudspeakers, especially the Orion. (See www.linkwitzlab.com for more details.)

My objective for this project was to design a loudspeaker with a dipole radiation for most of the frequency range. This was a difficult task. I ended up with a loudspeaker with a 2-kHz dipole radiation. Above this frequency the loudspeaker turns to monopole since only one tweeter was used.

Another equally important objective was the loudspeaker's off-axis horizontal frequency response. It had to be as similar as possible to the on-axis frequency response of the loudspeaker. A block diagram of the Horizon loudspeaker is shown in **Figure 1**.

The Horizon could be used to reproduce the full-frequency spectrum, but since I had two 12" subwoofers available, I decided to use them for the very low frequencies and limit the Horizon's bandwidth to frequencies above 62 Hz.

For this reason, I used the analog active crossover. This sums the left and the right channel to produce a mono-bass signal. This mono signal is then passed to a fourth-order Linkwitz-Riley low-pass filter at 62 Hz, which drives the subwoofer. Also, the left and right channels are buffered. This analog active crossover is similar to the subwoofer crossover described in my article "A Subwoofer for the Reflection," (*audioXpress*, September 2010). Refer to this article for more detailed information.

The Horizon loudspeaker's active crossover of the is realized with a Behringer DCX-2496 loudspeaker management system, which is a versatile device with three analog inputs and six analog outputs. It uses 24-bit/96-kHz A/D and D/A converters. The six outputs of the DCX-2496 drive six power amplifiers, one for each speaker unit. The Behringer crossover also includes the high-pass filter for frequencies above 62 Hz.

LOW-FREQUENCY, OPEN-BAFFLE CABINET DESIGN

The most important parameters for woofer selection for an open baffle are the large effective piston area, the large linear excursion, and the low distortion.

My woofer choice was the Peerless 830452 XLS. According to the Peerless, this woofer has an effective piston area of 352 cm^2 and a 25-mm peak-to-peak excursion. Additionally, it has low distortion and high-power handling capability.

I used two of these woofers in the Horizon. The woofers were connected in series and mounted in push-pull mode to reduce the distortion.

They cover the lower part of the spectrum down to the 62 Hz (–6 dB) frequency. A subwoofer is used below this frequency.

The main factor for the open-baffle cabinet design was the distance that separates the positive and negative sides of the woofer radiation.

Several different shapes can be used as open-baffle speakers. The purpose of all of them is to increase the distance between the two opposites sides of the woofer radiation in such a way that the total size of the cabinet will remain as compact as possible.

The H-frame is one of the simplest shapes to perform this task.

The woofer was placed on a baffle that extends its sides in the front and in the back. The result was the distance between the woofer's positive and negative sides radiation became larger, while the loudspeaker's total size remained acceptable.

Assuming the dipole loudspeaker's near-field response is absolutely flat, the dipole loudspeaker's far-field frequency response in the free-field is shown in **Figure 2**.

Photo 1: The Horizon loudspeaker (a) with front grilles and (b) without front grilles.

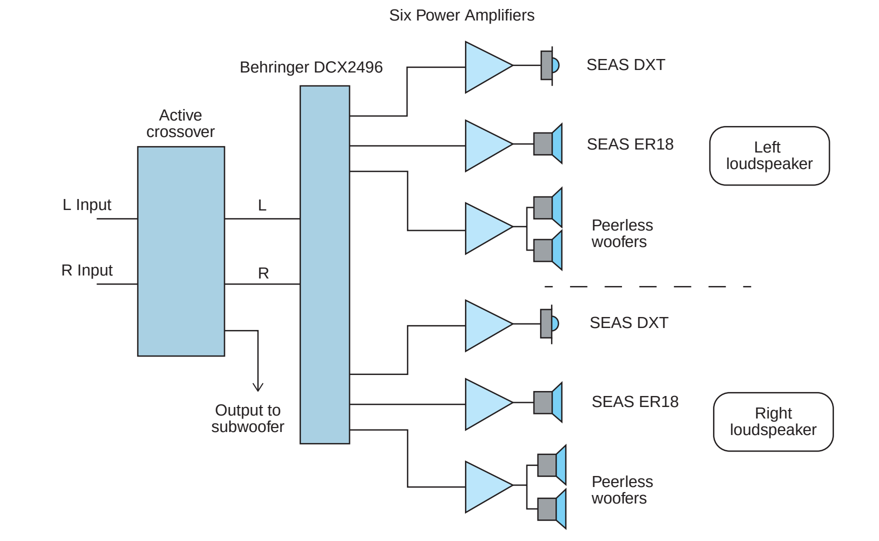

Figure 1: Take a look at the Horizon's block diagram.

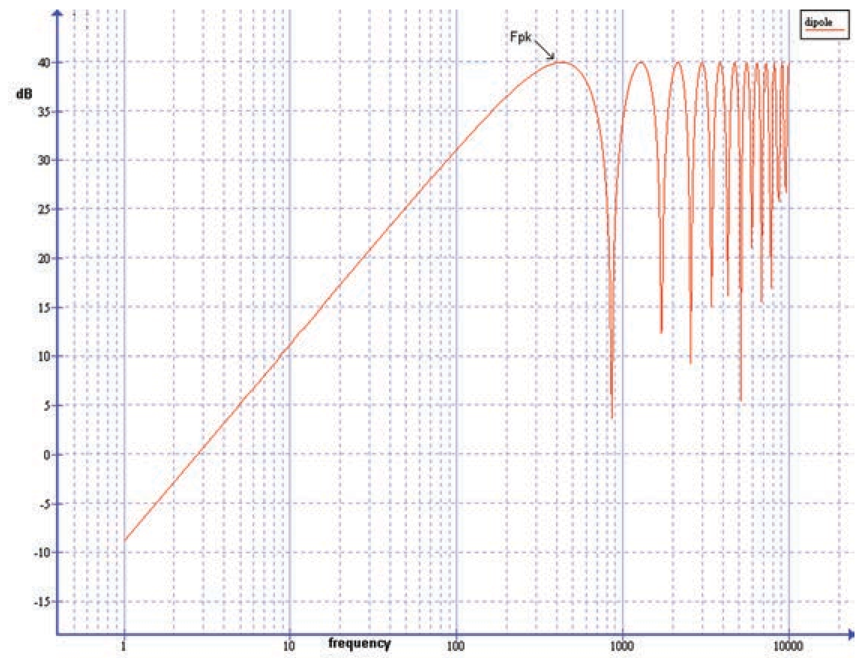

Figure 2: This plot shows the far-field frequency response of a dipole loudspeaker in the free-field.

The frequency F_{PK} where the first peak occurs is provided by:

$$F_{PK} = \frac{(0.5 \times c)}{D}$$

C is the sound velocity and D is the distance between the two dipole sources.

This peak in frequency response results when the negative polarity source after the 180° of phase shift due to the distance D added to the positive source and the resulting pressure is twice that of an single source (6 dB). Above the peak frequency, there are also a lot of other peaks and notches.

Below the peak frequency F_{PK}, the sound pressure falls with 6 dB/octave. This is due to the sound's front-back cancellation. At low frequencies, the distance D is small compared to the length of the sound, so it cannot provide any isolation between the positive and negative sources.

The frequency at which the open-baffle speaker has the same level output as the similar monopole (e.g., a closed-box woofer) is provided by:

$$F_{EQ} = \frac{(0.17 \times c)}{D}$$

Above this frequency, the open-baffle speaker has more output than a single monopole. Below this frequency, it has less output that also falls with 6 db/octave where the single monopole has a flat response.

The far-field, low-frequency rolloff has to be equalized with a circuit that boosts the low frequencies with 6 db/octave in order to flatten the frequency response.

The distance D for the woofer's cabinet including the front and back covering is estimated about 270 mm. Thus, the F_{PK} will be:

$$F_{PK} = \frac{(0.5 \times c)}{D} = \frac{(0.5 \times 344)}{0.27} = 640 \text{ Hz}$$

and:

$$F_{EQ} = \frac{(0.17 \times c)}{D} = 216 \text{ Hz}$$

The woofer cabinet's near-field response of the is shown in **Figure 3**. The response was taken at the middle point between the centers of the two woofers.

A very broad peak exists at about 280 to 300 Hz. This is due to the cavity resonance, which exists in front (and similarly in the back) of the woofers. This peak should be equalized so the near-field response of the baffle is flat.

Also, due to the woofer's low Q_{TS} value, the response is rapidly falling in the low frequencies and should also be equalized.

MIDRANGE & TWEETER BAFFLE DESIGN

For the woofer/midrange I chose the SEAS's ER18RNX-H1456-08 speaker, which is a 6.5" cone driver with a reed/paper pulp cone. According to SEAS, it has a 12 mm peak-to-peak linear excursion, while its maximum excursion goes up to 22 mm.

The ER18 is mounted on an open baffle and used for midrange frequency reproduction. The baffle is

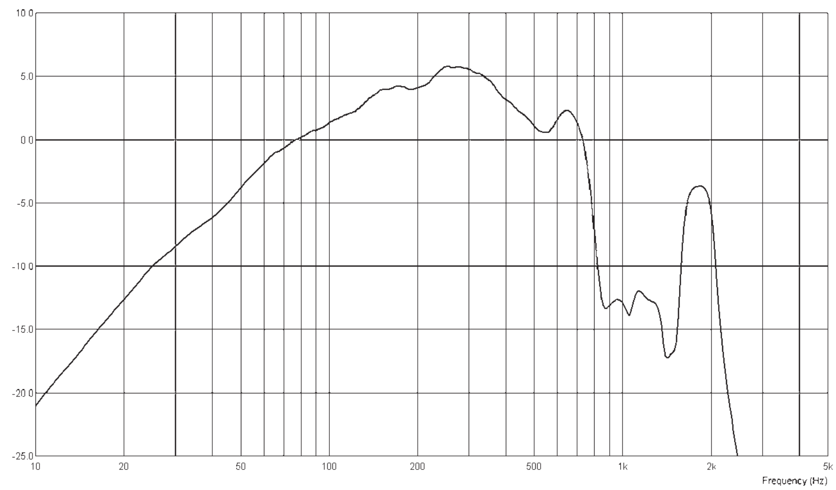

Figure 3: This graph shows the near-field response of the woofers taken at the middle.

folded back to increase the front-to-rear path length.

The baffle's size is a trade-off between the better extension at the low frequencies, which offers a large-size baffle, and the more constant directivity that offers a small-size baffle.

The low-frequency part's design theory is also applicable for the midrange panel's design.

A good estimation for the distance D between the front and rear part of the midrange dipole is made by taking half of the total width of the baffle plus the depth of folded-back parts of the baffle, which equals:

Photo 2: This is the midrange and tweeter's baffle.

$$D = \frac{(33.5 + 9 + 9)}{2}$$

This provides about D = 26 cm. So we have:

$$F_{PK} = \frac{(0.5 \times c)}{D} = 661 \text{ Hz}$$

and:

$$F_{EQ} = \frac{(0.17 \times c)}{D} = 225 \text{ Hz}$$

For the high-frequency reproduction, I used the SEAS 27TBCD/GB-DXT-H1499 dome tweeter, which is an aluminum/magnesium alloy dome tweeter with a DXT acoustic lens. The DXT technology optimizes directivity by carefully arranged diffraction edges in the drive unit's surface. The edges are placed precisely in certain distances from the cone.

The midrange and the tweeter's baffle is shown in **Photo 2**. The baffle's shape is unusual and needs an explanation.

As mentioned, one of my objective for this loudspeaker was to have a well-controlled polar response over the frequency spectrum. So I carried out a long investigation to find it.

I took a lot of measurements with different baffle shapes, with different drivers, and with the drivers placed on different positions. Some of the baffles I investigated are shown in **Photo 3**. The more controlled horizontal polar response was measured with the shape of the tweeter baffle shown in **Photo 2**. Actually, there is no baffle around the tweeter since it was eliminated as much as possible, to support the tweeter.

Figure 4 provides more details. The measurements indicate the polar response of two baffles with different shapes but with the same loudspeaker units.

The measurements in **Figure 4a** were taken with the typical rectangular baffle around the tweeter as indicated (see inset) below the curves.

The measurements in **Figure 4b** were taken without baffle around the tweeter (see inset).

Comparing the two measurements, notice that the results with the rectangular baffle have a broad directivity in the 1.5-to-4-kHz range. By

Photo 3: Here are three examples of different midrange and tweeter baffles.

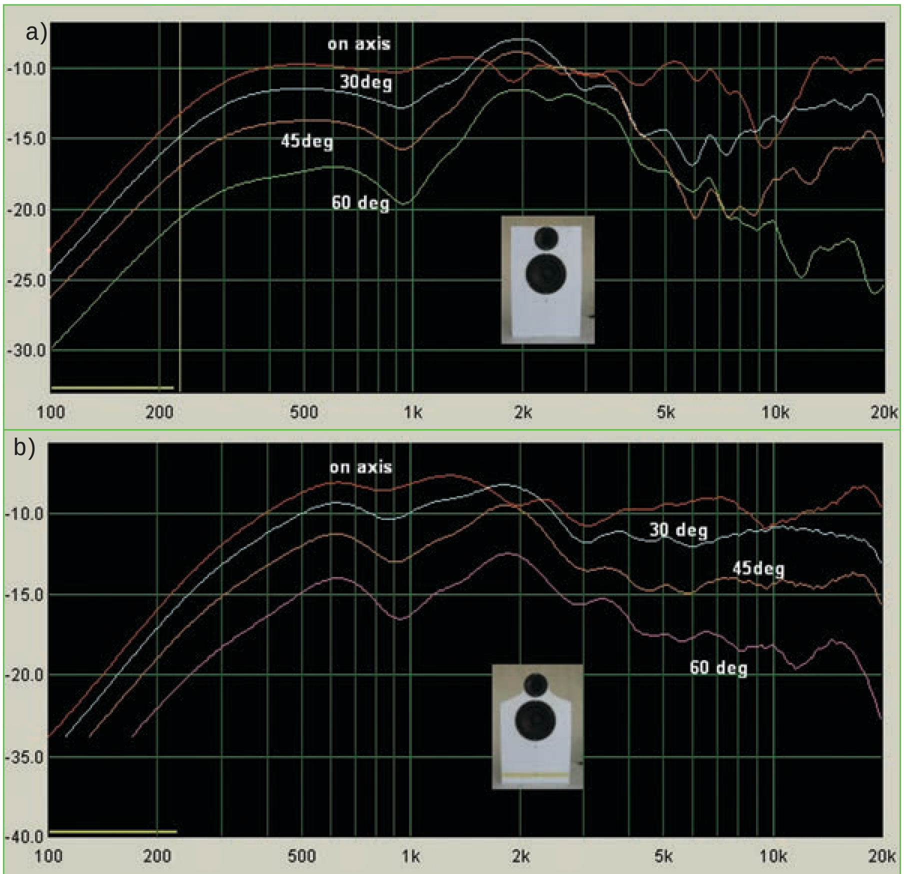

Figure 4: These graphs show the prototype loudspeaker's polar response with a baffle (a) and without a baffle (b) around the SEAS DXT tweeter.

eliminating the baffle around the tweeter, the directivity is much better controlled in the same frequency area.

SPEAKER CONSTRUCTION

The complete loudspeaker is shown in **Photo 1**. The loudspeaker and the midrange-tweeter panel's dimensions are shown in **Figure 5**.

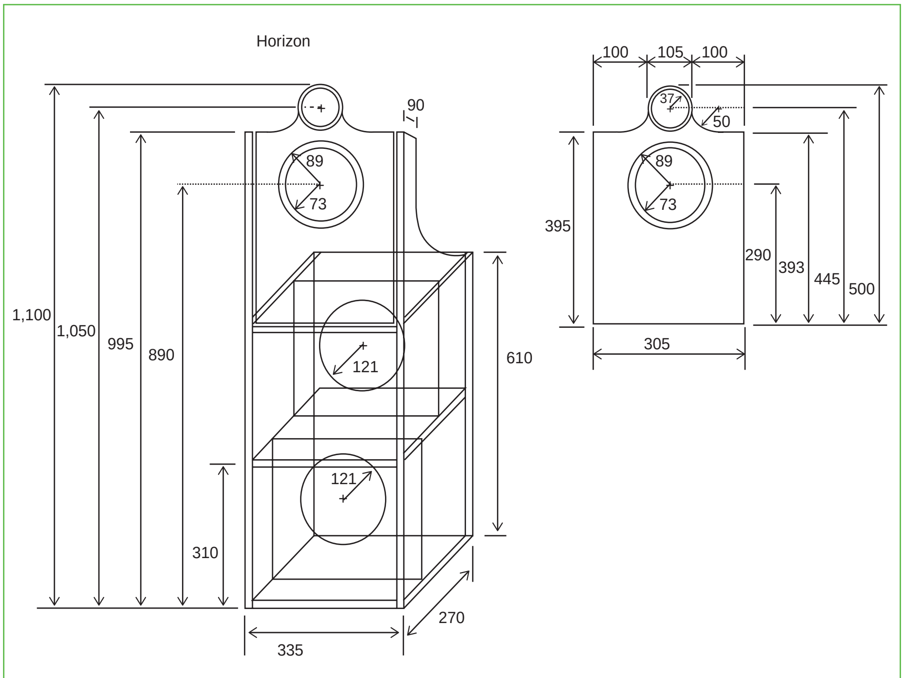

Figure 5: This diagram details the Horizon loudspeaker's dimensions.

The two woofers are mounted in the bottom of the box in a push-pull arrangement to reduce distortion. I connected them in series to make an easy load for the power amplifier that drives them since their nominal impedance is 4 Ω. Of course, they could be connected in parallel or be driven by two separate amplifiers if they are available.

The midrange and the tweeter's baffle was made to be removable from the main box for development reasons. Several M6 screws were used to afix it to the main box. The box was constructed with 20-mm plywood, screws, and glue, and covered with a wood imitation varnish. The decorative grilles were constructed from 4-mm plywood and acoustically transparent cloth. The top grille of the midrange-tweeter panel is usually removed during critical listening.

CROSSOVER DESIGN

The Horizon's crossover is realized using the DCX2496 loudspeaker management system.

The crossover frequencies selection is one of the most important loudspeaker's design decisions. The first step for crossover design is to have each unit of the loudspeaker's frequency-response measurement. This frequency response will be taken into account for the crossover and the equalizer computation.

It is obvious that the measurement should be taken with great accuracy.

I don't have access to an anechoic room. Also, I live in an apartment in a big city, so outdoor measurements are not possible. My only option is to perform the measurements in my living room, which fortunately is large enough. This obviously implies some serious limitations, mainly in the low midrange, and bass frequencies. Over the years, I have used several methods to overcome this problem.

I had some acceptable results with two different set ups. The first set up is used for measurements at frequencies above 500 Hz (see **Figure 6**).

The loudspeaker is placed above the floor and about 1.5 m away from the sidewalls. The microphone is at a 2-m distance from the speaker. The microphone is 1.35 m above the floor, which puts it in the middle of the distance between floor and ceiling. With this set up, it is possible to have a reflection-free impulse-response measurement of about 4 ms long. This gives a resolution of only 250 Hz, which obviously is not sufficient for assessing a loudspeaker's the lower frequency performance.

The second set up is used to measure frequencies below about 1 kHz.

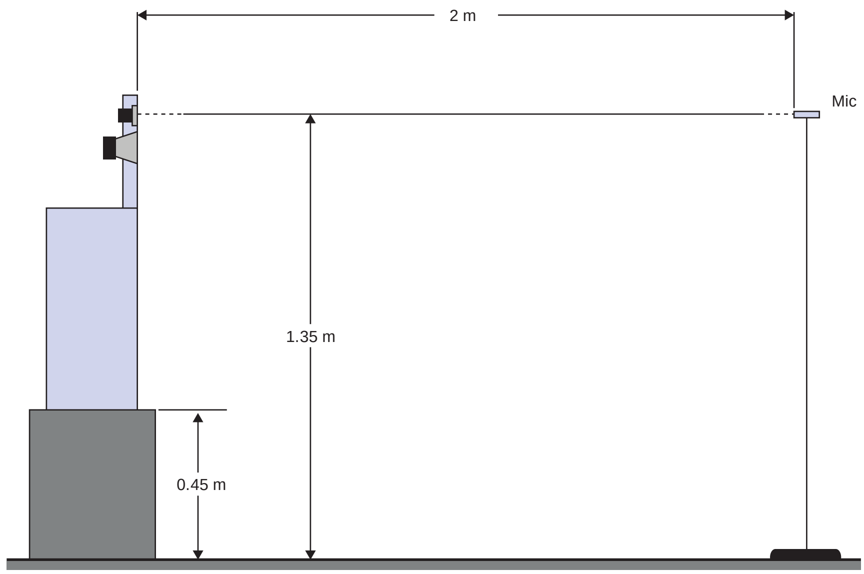

Figure 6: This is the first setup for the loudspeaker measurement.

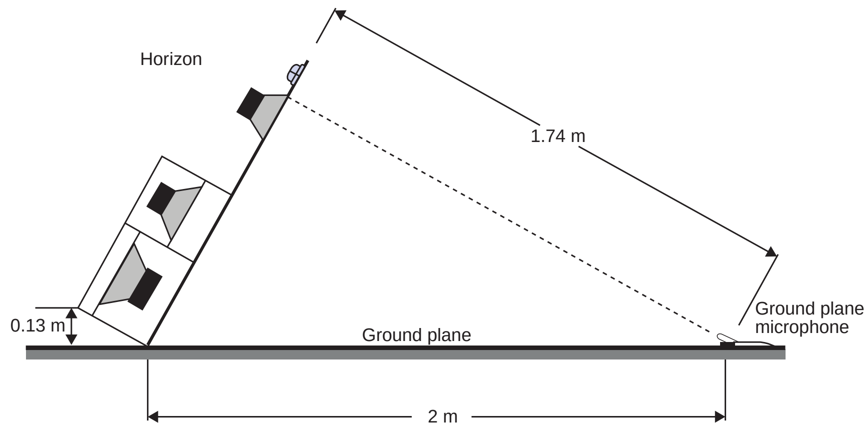

Figure 7: This diagram shows the second setup for the loudspeaker measurement.

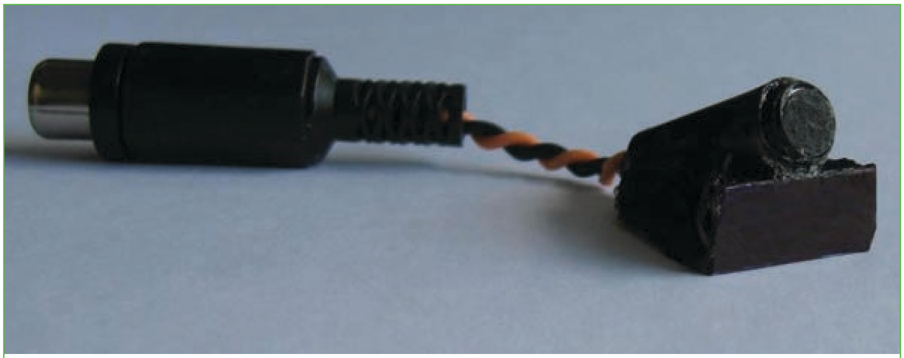

Photo 4: This is the microphone used for ground plane measurements.

The loudspeaker is placed on the floor and at least 1.5 m away from the sidewalls (see **Figure 7**). Additionally, the loudspeaker is not perpendicular to the ground but is inclined to the front so the midrange and the woofer's distance from the microphone is about the same as it will be in the listening position.

The microphone is at a 2-m distance from the speaker and flush mounted with the floor. For this reason, I used a special microphone (see **Photo 4**). This consists of a Panasonic WM-61A omnidirectional back electret microphone cartridge that is glued in a small piece of wood so that it forms an angle of about 28 to 30° with the floor. This keeps the microphone's axis on the same axis with the loudspeaker and there is no high-frequency loss.

This set up cannot eliminate the room reflections, but it provides results that could be useful after some thinking. For example, it is obvious that the peak at 80 Hz in the woofer response of **Figure 10** is caused by room reflection. It does not have an anechoic chamber's accuracy or an outdoor measurement, but it is the best I could do.

MEASUREMENT PROCESS

All the project measurements were performed using ARTA software's demo version (www.fesb.hr/~mateljan/arta), which is a fully functional program, except for saving or loading of files.

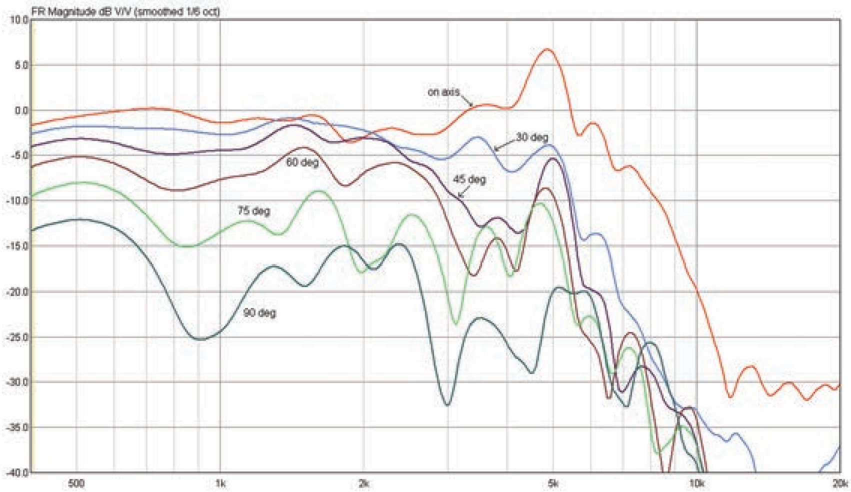

Figure 8: This plot shows the polar response of the direct-driven SEAS ER18 unit.

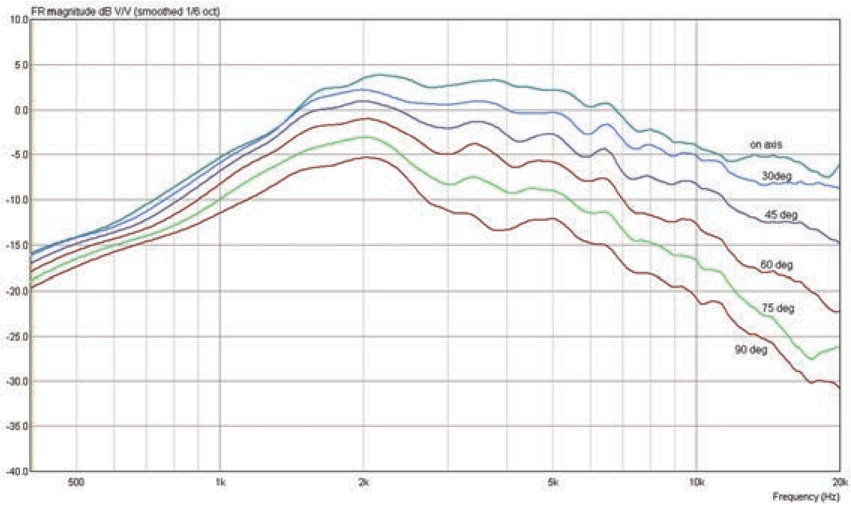

Figure 9: This plot shows the polar response of the direct-driven SEAS DXT speaker.

I used the PC's Print Screen function and the Paint program to further process the captured image.

Now, I will describe the measurements I took using the first set up.

Figure 8 shows the polar response of the SEAS ER18. **Figure 9** shows the SEAS DXT tweeter. Both are direct-driven without any equalization.

For the ER18, the response is well controlled up to the 1-to-1.2-kHz frequency. For the DXT tweeter, the polar response is very well controlled above 2.5 to 3 kHz.

But, there is a 1-to-2.5-kHz frequency range where the polar response of the midrange and the tweeter are broadening. This cannot be fixed with this design. It would be possible to have a more uniform polar response if I used an additional driver with a 10-cm cone diameter and a very small baffle around it. Of course, this means a four-way system, which was not possible for this project.

I chose the 2.03-kHz crossover frequency, which seemed like a good compromise concerning the units' polar response and the tweeter's reproduction capability.

After selecting the crossover frequency, the drivers should be equalized to have a flat on-axis frequency response. I used the ARTA software to generate target curves and to overlay them with the current measurement. I set the target curve for the tweeter as a high-pass, fourth-order Linkwitz-Riley filter with a 2,030 Hz-crossover frequency.

Then, I adjusted the DCX2496's equalization settings until the response of the tweeter as measured was the same as the target response.

The equalization capabilities of the Behringer are very powerful and the above procedure is easily performed.

I used the same method for the midrange equalization for the frequency range above 500 Hz. I set the midrange target curve as a low-pass, fourth-order Linkwitz-Riley filter with the crossover frequency set at 2,030 Hz and I adjusted the Behringer equalizers to match the two responses.

An important issue for the correct operation of the Linkwitz-Riley crossovers is to set the delay between the units in the crossover frequency range so they will have the same phase response. This is easily accomplished with a DSP processor (e.g., the DCX2696). I used the tweeter output's Short Delay function. I adjusted this delay until I measured the greatest notch to the frequency response measurement with the tweeter polarity inverted. For this notch, the delay was set at 160 µs.

Then I used the ground plane set up to adjust the crossover frequency between the woofer and the midrange (see **Figure 7**). Here the measurements need some mental interpretation since they include some room reflections.

In this range, both the woofer and the midrange are radiating as dipoles, so the polar response is not a problem. The main issue here for the crossover frequency's selection is the midrange unit's volume capability in the lower frequencies.

Considering the ER18 unit's characteristics (Sd = 136 cm^2 and X_{MAX} = 6 mm), I decided to set the crossover frequency at 161 Hz.

The midrange unit's equalization was performed following the same procedure as with the tweeter.

The target response was set as a high-pass, fourth-order Linkwitz-Riley filter at 161 Hz. The DCX2496's equalization was adjusted again until the midrange response and the target response were as close to one another as possible.

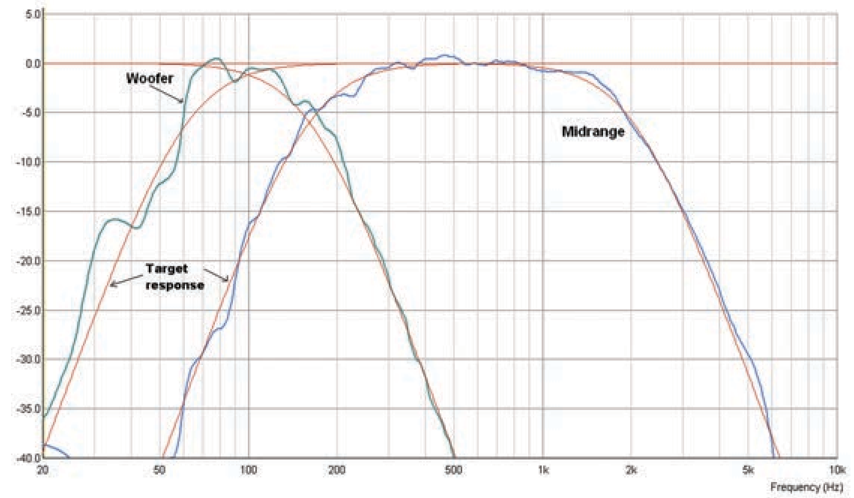

Figure 10: The woofer and the midrange's frequency response is compared with their target responses (ground plane setup).

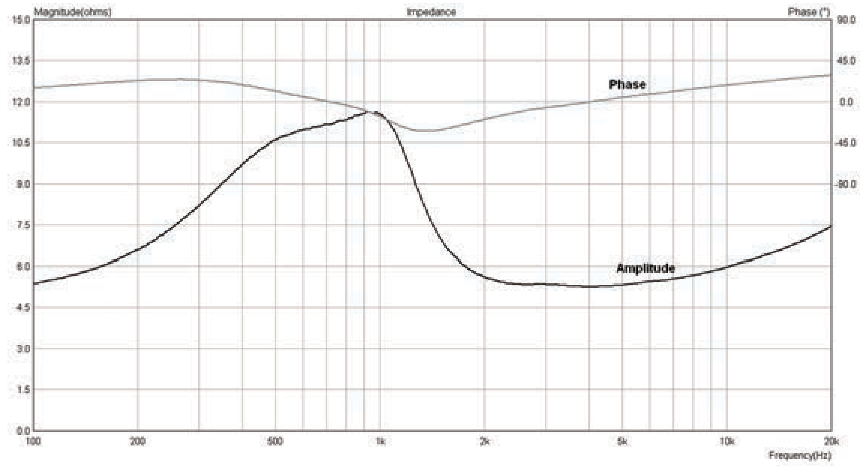

Figure 11: The SEAS DXT tweeter's impedance is shown.

The woofers' equalization was also performed following the same procedure. Here the target response was set as a high-pass, fourth-order Linkwitz-Riley 62-Hz filter and a low-pass, fourth-order Linkwitz-Riley filter with a 162-Hz crossover frequency.

The woofer's final response in comparison with the only target response is shown in **Figure 10**.

To set the delay between the woofer and the midrange, I inverted the midrange's polarity and I adjusted the midrange output's Short Delay function of the SCX2496 until I measured the crossover region's maximum notch, which uses a 500-µs delay.

This means an additional 500 µs should increase the delay of the tweeter, so the delay was finally set at 660 µs.

MORE MEASUREMENTS

The following measurements were taken as a final Horizon loudspeaker final design verification. Measurements for the loudspeaker's horizontal polar response were taken on-axis and off-axis at 30°, 45°, 60°, and 90°. Except for a broadening between 1.5 and 2.5 kHz, the loudspeaker's directivity is well controlled due to the midrange's dipole radiation and the SEAS tweeter's DXT acoustic lens.

Each of the Horizon loudspeaker's drivers has its own amplifier, and it is important to know what kind of load each amplifier will be asked to drive. This can be determined by measuring each driver's impedance.

The SEAS tweeter's impedance is shown in **Figure 11**. The minimum 5.3-Ω impedance of occurs at around 4 kHz. The frequency region's phase angle that covers the tweeter (above

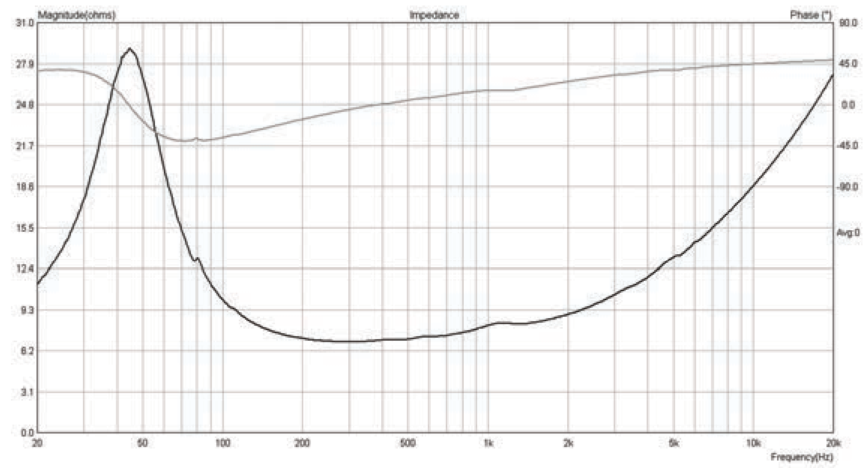

Figure 12: The SEAS ER18 midrange's impedance is shown.

1.5 kHz) is ±30°, which is a relatively easy load for a power amplifier.

The SEAS midrange's impedance is shown in **Figure 12**. The minimum 6.9-Ω impedance with a −4° phase angle occurs at 315 Hz. The 29-Ω maximum value occurs at 44.5 Hz, the resonance frequency. The frequency range's phase angle that covers the midrange is ±30°. This indicates an easy load for a power amplifier.

The two Peerless woofers' impedance, connected in series, shows a 9.7-Ω minimum impedance with a −7°phase angle occurs at 125 Hz. The 115-Ω maximum impedance with a 0.7° phase angle occurs at 22 Hz. The frequency range's phase angle that covers the woofer is between −45 and 15°, which is not a difficult load for a power amplifier.

The SEAS DXT tweeter's nonlinear distortion shows an input with a 10-cycle tone burst signal at 2 kHz at a voltage level of 8.9 V_{RMS} or 40 V_{PP}. This corresponds to a 105 dBSPL at 1 m sound pressure level. The total test signal lasted 5 s. The result is good with the level of the second harmonic frequencies down at about −35 dB.

The nonlinear distortion at 1.6 kHz shows the level set again at 8.9 V_{RMS} corresponding to a 103 dBSPL at 1 m sound pressure level. The test signal lasted 5 s. The level of the second harmonic frequency is about −31 dB down. This result is good, keeping in mind that this is a low frequency for a tweeter.

FINAL FREQUENCY-RESPONSE ADJUSTMENTS

After the completing the measurements, it was time to evaluate the loudspeaker's acoustic performance and adjust the frequency response accordingly. I installed the loudspeakers in my living room with a 2.3-m distance between the left and right loudspeaker. The listening distance was 2.7 m.

By listening to known program material, it was obvious that the loudspeaker sounded a little bright. This was very noticeable in the reproduction of the upper range of violins and trumpets. To compensate for this brightness, I adjusted the tweeter down to a lower level. After a lot of trials, I found the right balance and the correct level for the tweeter.

I was happy with this setting for some time until I read about the Orion revision 3 on Siegfried Linkwitz's website (www.linkwitzlab.com). There, Linkwitz proposes a low-pass shelving filter at the input of the loudspeaker to adjust the flat on-axis loudspeaker response. The filter is almost flat up to 500 Hz and then gradually reduces the high frequencies by 3.2 dB at 20 kHz.

I restored the tweeter at the flat on-axis response level, and I set up a similar low-pass shelving filter to the DCX2496 processor. This was an improvement. Since then, I use this filter.

WORTH THE EFFORT

The Horizon was not an easy project. I spent a lot of time and effort designing, manufacturing, measuring, and finally acoustically adjusting the loudspeaker. But I think the result was worth the effort. They sound really great. *aX*

SOURCES

DCX2496 Loudspeaker management system
Behringer | www.behringer.com

WM-61A Omnidirectional microphone cartridge
Panasonic Corp. | www.panasonic.com

830452 XLS Woofer
Peerless, Ltd. | www.peerlessaudio.com

ER18RNX-H1456-08 Speaker and 27TBCD/GB-DXT-H1499 dome tweeter
SEAS | www.seas.no

Experimenting with Electrostatic Speakers (Part 1)

Tips for selecting the proper materials for ESL fabrication

By Richard K. Mains

In the early 1990s, I began constructing electrostatic speaker cells based on instructions from Roger Sanders for his compact system. I have continued experimenting off and on with hybrid electrostatic speaker systems. I still consider the Sanders articles required reading for anyone interested in these speakers. However, since these articles first appeared, new material has become available that makes electrostatic cell construction easier and more reliable. I have also developed some techniques and insights that have proven helpful in my work with these systems. By presenting this information, I hope to make electrostatics more accessible to audiophile hobbyists and to generate some renewed interest in these marvelous speakers.

Photo 1: These are essential materials for making one pair of electrostatic loudspeakers (ESLs).

FABRICATION

An electrostatic cell consists of a thin plastic film sandwiched between two metal stators. The film must be separated from the stators by insulating spacers. A high DC voltage is applied between the stators and film. The AC music signal is applied between the stators, which causes the film to vibrate.

I used the same materials as originally recommended by Sanders to make the stators and plastic supports. (For an approach using different stator and spacer materials, Charlie Mimbs's *audioXpress* article, "Hybrid Electrostatic Loudspeakers Constructed with a Twist," April 2012.) I used 2' × 3' Lincaine perforated aluminum sheets in the Albras finish for the stators. I cut 65-mils Lexan Sheets to size for the plastic insulating spacers. **Photo 1** shows the materials used for one pair of electrostatic cells. In this photo, the Lincaine and the plastic spacers have been cut to size and contact holes have been drilled. Also, items of the same size have been stacked on top of each other.

I used the same spacer dimensions as Sanders, except I made the center supporting strip 0.5" instead of 0.25". **Figure 1** shows the overall dimensions of the spacers that I used.

I drilled holes in the corners and used #4 screws to bend them back to make electrical contact with the stators. If you use #4 screws, you don't need to cut a notch in the stator to clear the bolt for the diahragm contact. I made the diaphragm contact holes in the two 13" x 0.75" spacers at the top of the cell. **Figure 2** shows the holes I made for the diaphragm contact. The smaller hole provides clearance for a #4 screw. The larger hole is sufficient for a #4 flat washer that contacts the copper foil on the diaphragm surface.

First, drill the smaller hole shown in **Figure 2** with the two spacers stacked on top of each other, to ensure the holes line up. Then, enlarge the hole in one of the spacers. **Photo 2a** shows how the contacts look on the rear of the cell. **Photo 2b** shows the front view.

POLYURETHANE GLUE

When I first made electrostatic cells in the 1990s, I used Devcon 2 Ton epoxy as recommended in

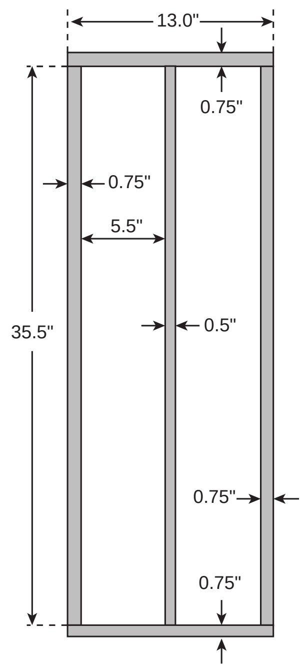

Figure 1: The dimensions of 65-mil spacers used for the cell are shown.

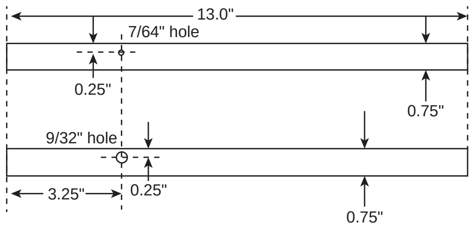

Figure 2: Holes are placed in top spacers for the diaphragm contact.

Sanders' articles. This works well for bonding metal stators to the plastic insulators, but I was unable to achieve a good bond between the mylar film and the plastic insulators. I was able to construct some working cells that lasted for years, but there was a high failure rate. This may reflect partly on my construction abilities, but I have also heard of problems from other builders who used epoxy.

Instead, I recommend using polyurethane glue, which has advantages and disadvantages. Don't get it on your hands, and don't breathe the fumes. Wear protective gloves if there is any chance you may come in contact with the glue. Also, use a respirator whenever you will be exposed to significant amounts of the glue. The big advantage, however, is the bond to the mylar film is much better than it is with epoxy. You can still break the bond with enough effort, but I consider this as an advantage in case you ever need to take the cell apart to fix it.

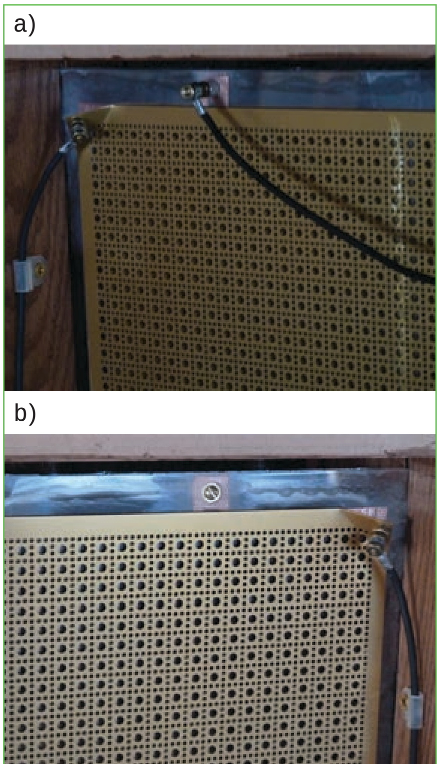

Photo 2: This image shows the diaphragm and stator contacts on the back of the cell (a) and on the front of the cell (b).

These days polyurethane glue is available in several different varieties. Unfortunately, I have not experimented with many different types. I use ER Audio's polyurethane glue, which is expensive but works well. Two kinds of glue are available, Polyurethane, which comes in a 100-ml bottle, and Diaphragm Adhesive, which is also "polyurethane-based." I found the Diaphragm Adhesive difficult to control. It seemed to come out in large droplets rather than a small consistent bead. On the other hand, I found the Polyurethane in the 100-ml bottle worked well for bonding spacers to stators and bonding the diaphragm to the spacers. It also has a shelf life of at least a few months, if kept in a sealed container. I have found one bottle sufficient for constructing up to 10 or so cells. The glue is slow curing, so there is plenty of time for each step. The downside to this is that you need to let the glue cure for at least a day before continuing.

There is a problem using this glue as it is delivered. ER Audio provides its 100-ml polyurethane in a plastic bottle with a cap with no applicator tip. The reason is it expects you to apply this glue by pouring it out and using rollers. For my application, this would be wasteful and messy. I only need a thin bead of the glue to make electrostatic loudspeakers (ESLs).

My solution was to buy ER Audio's polyurethane glue but transfer it to another container as soon as I received it. This must be done quickly since you don't want to expose the glue to air for long. I used a funnel to pour the glue into the new container. Use as large a funnel as possible since the glue is rather thick and will flow very slowly if constricted. I recommend you do this step in a well-ventilated area while wearing a respirator, since a large quantity of the glue will be exposed.

I used Howard Electronic Instruments's bottles the polyurethane glue (see **Photo 3**). The 4-oz bottle with a Luer Lock cap was just right for transferring the polyurethane glue from the other bottle. You can also buy extra caps, which is a good idea since the cap gets plugged up after a few gluing steps. The 16-gauge, plastic tapered-tip needles were the right bead size for my application. But if you need a different size, they have several to choose from. (I don't think I would use a smaller needle, but you may prefer a larger opening than I used.) Get extra needles, since you have to replace the needle after almost every gluing step. I also bought the rubber-tip cap to seal the ends of the needles.

COATING THE DIAPHRAGM

The diaphragm must slightly conduct for the ESL to work. In my first cells, I used graphite as recommended by Sanders. I didn't

Photo 3: A plastic bottle and needle from Howard Electronics is used to apply the ER Audio polyurethane.

mind the work required to rub the graphite into the mylar surface, but there are other disadvantages to this method. Theoretically, there is less distortion in ESLs if the diaphragm has a high-resistance coating. Unfortunately, powdered graphite yields a low resistance. Also, with high-resistance coating, there may be less possibility of arcing that can damage the diaphragm. In the 1990s, there were other coatings besides graphite available that would yield higher resistance, but there was concern they would not be permanent.

Techspray's Licron Crystal Permanent ESD Coating is an option that provides a very high-resistivity coating and will remain permanently on the diaphragm. I started using it last year. So far, there is no sign it is coming off the speakers I have used daily. It can come off if you scrape it enough, but it requires some effort. It can also come off if you don't give it enough time to cure.

Although Licron offers a permanent, high-resistance coating that doesn't require nearly as much effort as powdered graphite, there are also some disadvantages associated with it. I find it difficult to evenly apply, since it tends to sputter. As a result I often have to spray the diaphragm two or three times before the entire surface is covered. Between coats I wait at least a day for the previous layer to dry, so this can become time consuming. Also, you should probably wear a respirator when applying Licron. And, nearby surfaces should be masked off.

In my early ESLs, I used a tinfoil contact and conductive paint to contact the diaphragm. This works


Photo 4: Use acrylic adhesive to tack the plastic spacers together.

if you are careful, but it is easy to damage the tinfoil. For a diaphragm with a high-resistivity coating, it is a good idea to contact the diaphragm everywhere along its periphery, rather than just at the top. For these reasons, I started using copper foil from ER Audio. It is self-adhesive, 25-µm thick, and available in 4- and 6-mm widths. The 4-mm width was best for my cell size. I used the copper tape, which comes in 33-m rolls, to form a contacting ring around the outer edge of the diaphragm. I also used it to make contact with the diaphragm where the contact screw is mounted. I will show how I did this in the second article in this series.

CELL FABRICATION

One of the first steps in cell fabrication is to glue the perforated metal stators onto the spacers. It is a good idea to "tack" the spacers together at the corners before doing this step, to keep the spacers flat and to prevent them from moving. I used Weld-On 4 Acrylic Adhesive to do this (the 4 indicates fast setting). I applied the adhesives to the seams where the spacers fit together with a syringe from a local medical supply store. **Photo 4** illustrates this procedure.

If you want to heat shrink the diaphragm, I recommend buying a temperature-controlled heat gun. You can get by without one if expense is an issue, but it is nice not to worry about possibly burning a hole in the diaphragm. I bought a Bosch 1944LCDK programmable heat gun kit, which comes in a carrying case. I set the heat gun at 390°F (199°C) and I've had no problems burning holes in 12-µm film. I usually make two or three slow passes during the heat shrinking process.

Finally, I want to discuss the most difficult material to obtain for ESL fabrication, the diaphragm's plastic film. One important decision the electrostatic speaker builder must make is what diaphragm thickness to use. Since suitable film is usually hard to find, we often settle for whatever is available. But most of

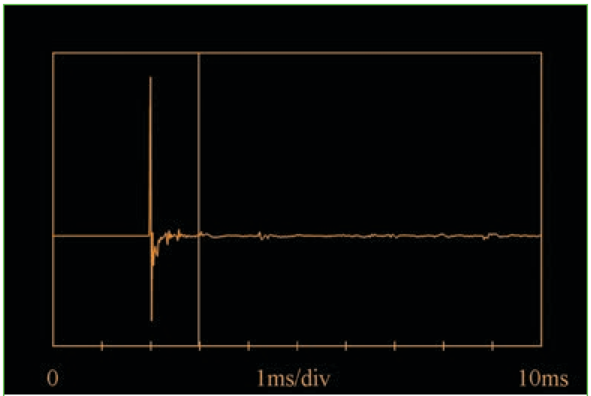

Figure 3: The MLS image was used to obtain my ESL's 10' on-axis impulse response.

conditions include performing the measurement on-axis and in the far-field of the speaker, and driving the speaker with constant current. This last requirement is contrary to the way that most amateur builders drive electrostatics. We use constant voltage output amplifiers and step up the voltage through a transformer. But according to the conclusions in the advice I see is to use the thinnest diaphragm material possible. For example, ER Audio sells a 3.5-μm Mylar Type C film, which it describes as the best performing material for ESLs.

Why is this the case? One of the often cited virtues of electrostatic speakers is the moving mass is so much lighter than conventional speakers. It frees electrostatics from mechanical resonances that plague other speakers and enables them to reproduce sound more accurately. It makes sense that the diaphragm should also be as light as possible. And, if we can ignore the mass of the diaphragm, a very elegant method of analyzing electrostatic speakers becomes possible. Peter Walker of Quad fame has shown that for an ideal electrostatic speaker and under certain conditions, the on-axis response of the speaker should be perfectly flat and independent of frequency. One of the conditions he cites in his article, "New Developments in Electrostatic Loudspeakers," (*Journal of the Audio Engineering Society*, November 1980), is that the diaphragm's presence must be negligible with respect to the surrounding air. Imagine the polarizing voltage is applied to an ESL, but the stators are left open-circuited, and there is a sound wave from another source, so the ESL acts as a microphone. A requirement for Walker's analysis to be valid is that the diaphragm must be light and unconstrained enough not to disturb the sound wave. Other

Walker's paper, driving an electrostatic with constant voltage should result in an ESL response in the far-field that is not flat, but rises by 6 dB per octave. This is because the electrostatic looks a lot like a capacitor, and a capacitor has an impedance that is inversely proportional to frequency. If you drive a capacitor with constant voltage, the current will double in magnitude with each doubling of the frequency.

Even though I drive my ESLs with constant voltage, I do not observe this 6 dB/octave rising high end. To illustrate, I present an on-axis measurement of one of my ESLs with the microphone placed about 10' from the speaker. I used the maximum length sequence (MLS) technique to derive the impulse response of the speaker, then I used fast Fourier transform (FFT) analysis of the time waveform to determine both the magnitude and phase response in the frequency domain. I used a Brüel & Kjær 4006 studio microphone to perform the measurement. **Figure 3** shows the impulse response I measured for the speaker. For this measurement, there was no active crossover in the circuit. (However, I did include resistance in series with the step-up transformer output to correct for the transformer resonance effect. I will discuss this later in the article series.) The vertical line in **Figure 3** shows the truncation point I used. Everything to the right of this point was treated as zero in the FFT analysis in an effort to remove room reflections. Therefore, this is a "quasi-anechoic" measurement.

Taking the FFT of **Figure 3** yields the magnitude response versus frequency of the ESL shown in **Figure 4**. There are localized dips and peaks in the frequency response, but there is no evidence of a rising high end. I should mention that the roll-off starting around 20 kHz is at least partially due to my microphone's frequency response. According to the calibration curve, it is down by 1.5 dB at 20 kHz and about 7 dB at 30 kHz.

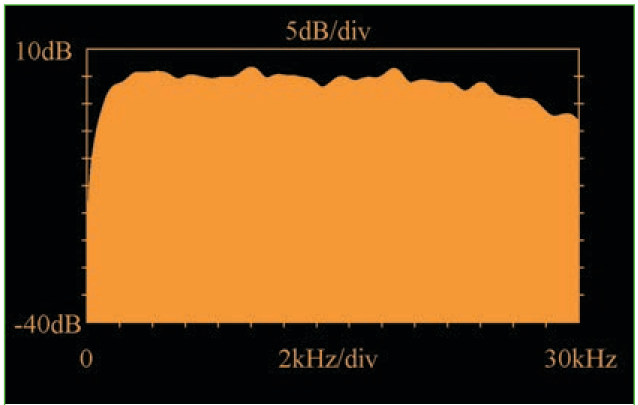

Figure 4: This shows the 10' on-axis frequency response of my ESL.

I believe Walker's analysis is correct, and furthermore, he has presented measurements that confirm it to be valid. So what is going on? I can only conclude that my ESLs must not fulfill all the requirements that were assumed in his derivation. One requirement that is easy to check is whether the mass of the diaphragm can be neglected with respect to the mass of the surrounding air. Assuming the velocity of sound is 345 m/s, then the wavelength of sound at 20 kHz is 17.2 mm. The density of air is about 1.18 kg/m^3. Therefore, the mass of air contained within one wavelength at 20 kHz is 0.0203 kg/m^2. The density of my film is 1.36 × 10^3 kg/m^3, and since I am using 12-μm thickness, its mass is 0.016 kg/m^2. Clearly, the mass of 12-μm film cannot be ignored with respect to the mass of surrounding air at high audio frequencies. Also, I suspect I am stretching the film sufficiently during heat shrinking so the restoring force of the diaphragm would interfere with a surrounding sound field.

As a result of these effects, I think the diaphragm response is falling off with a frequency at a rate that counteracts any rise that would be caused by constant voltage drive. I am speculating about this, since I have not tried a thinner diaphragm material and I can't say for certain that this is the case. But whatever the reasons, **Figure 4** demonstrates that it is possible to obtain flat response using a voltage drive with 12-μm film for the diaphragm. Since this thickness is easier to obtain, easier to use, and more durable, I recommend 12-μm film if you intend to drive your speakers with constant voltage. If you use thinner material, you may have to change to a different method of driving your speakers to obtain flat response.

OBTAINING THE FILM

But now on to the difficult part, how to obtain suitable film. If you wish to stretch the diaphragm by heat shrinking as I do, then the problem is even more difficult since you need material that will heat shrink the right amount. In the 1990s, Sanders was kind enough to supply quantities of Mylar Type S film to builders. This material will shrink up to about 3.5% if heated at 190°C, and I found that it worked beautifully for electrostatics. Recently, I have not been able to find any more of the Mylar Type S material. I talked with someone at DuPont Teijin Films who told me that Mylar Type S is no longer manufactured. But, he said there is a Melinex Type S material that is still made, which is identical to the old Mylar Type S. The only differences are that Melinex is made in the U.S., whereas Mylar is manufactured in Europe. The minimum thickness for

Melinex Type S is 48 gauge or 12 µm (4 gauge is 1 µm in plastic film parlance). As I explained earlier, I prefer 12-µm thickness for my ESLs, so this was not a problem.

But even though Melinex Type S is still available, the amount it typically wants you to buy is daunting. If you order directly from DuPont Teijin Films, it has a minimum order of 5,000 lbs! Fortunately, there is another option. There are several Mylar and Melinex film distributors that sell in smaller quantities, if the film you want is in stock. The distributor I had the best luck with was Plastic Suppliers. But even from this distributor, we are still talking about huge quantities. The smallest amount of Melinex Type S that I was able to buy was 20,000′, and that was only because they had a special deal on some rolls. Normally, I would have had to order 40,000′. However, the company was able to cut the Melinex film to just the width I needed, which was 18″. If this width is sufficient for the ESLs you want to make, I will be glad to provide moderate quantities of this film to builders. E-mail me at rkm@acoustic-instruments.com.

If you require a different film width for your project, or a film other than 12-µm Melinex Type S, I can suggest some possible strategies. You can find small quantities of Mylar or Melinex or Hostaphan film for sale online.

Be aware that not all films have the same heat shrinking properties as Melinex S. For example, I once purchased some Mylar Type C film from a seller on eBay; however, I was disappointed to discover that the diaphragms I made from this material did not adequately heat shrink. According to the datasheet, Mylar Type C 48 gauge will heat shrink to 2.1% in one direction, but it will only shrink to 1% in the orthogonal direction (i.e., it does not heat shrink uniformly). Melinex Type S will shrink uniformly up to 3.5%.

Another possibility is to get a group of ESL builders to buy a large quantity of film from one of the DuPont Teijin Films distributors and share the cost.

ELECTROSTATIC CELLS

In this article, I discussed some of the raw materials required for making ESLs, described how to obtain them, and showed preliminary cell fabrication steps.

In the second article in the series, I will provide a step-by-step description detailing how I make electrostatic cells, with particular emphasis on techniques that differ somewhat from the original instructions provided by Sanders. Then, in the third article in the series, I will discuss the electronics I use to drive the speakers. *aX*

RESOURCES

C. Mimbs, "Hybrid Electrostatic Loudspeakers Constructed with a Twist," *audioXpress*, April 2012.

R. Sanders, "A Compact Integrated Electrostatic/Transmission Line Part I," *Speaker Builder*, January 1990.

——, "A Compact Integrated Electrostatic/Transmission Line Part II," *Speaker Builder*, February 1990.

——, "A Compact Integrated Electrostatic/Transmission Line Part III," *Speaker Builder*, March 1990.

——, "An Electrostatic Speaker System Part I," *Speaker Builder*, February 1980.

——, "An Electrostatic Speaker System Part II," *Speaker Builder*, March 1980.

——, "An Electrostatic Speaker System: Part III," *Speaker Builder*, April 1980.

——, "The Sanders Electrostatic Speaker," *The Audio Amateur*, April 1975.

P. Walker, "New Developments In Electrostatic Loudspeakers," *Journal of the Audio Engineering Society*, November 1980.