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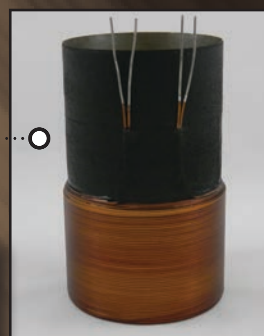

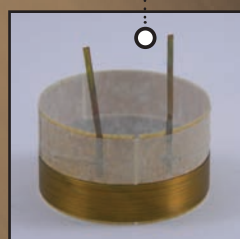

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From the Editor's Desk

Audio International

I am pleased to announce that we updated *audioXpress*'s layout. If you turn to pages 6–7, you'll see that we added a new imprint to the magazine. Our goal is to make each member of the *audioXpress* community aware that he or she is part of diverse international group of audiophiles, designers, and technology enthusiasts. Since I work on all the titles in the Elektor group's portfolio in one form or another, I have the privilege of interacting with some of the most talented and creative people on the planet. And what excites me the most is that many of you share a passion for innovation, technology, and knowledge. *audioXpress* readers are interested in microcontroller-based amp projects published in *Elektor*. I hear from speaker builders who want to learn more about the electronics applications in *Circuit Cellar*. And *Elektor* readers worldwide have a serious passion for DIY audio projects. In fact, several of my colleagues at *Elektor* in The Netherlands are musicians and audiophiles.

After you check out the new imprint, you'll find the rest of the issue packed with interesting content. We've got a little of everything.

On page 8, Mike Klasco and Steve Tatarunis review voice coil materials and how they work. Once you're up to speed on voice coil assembly, read Vance Dickason's review of Fountek's FR89EX driver (p. 19). Then, flip to page 14 for a Charlie Mimbs's DIY electrostatic loudspeaker (ESL) project.

Interested in amplifiers? Check out Richard Honeycutt's two articles. One is a review of a book on amp techniques (p. 26). The other covers the frequency response of hollow-state amps (p. 37).

Many of the same aforementioned technologies were discussed and presented at AES 2012. On page 30, Gary Galo describes his trip to the convention.

Finally, note the interview on page 29 with transformer designer and businessman Per Lundahl. His story is sure to inspire you to start an audio project—or even your own audio-related business!

Regards,

C. J. Abate
editor@audioxpress.com

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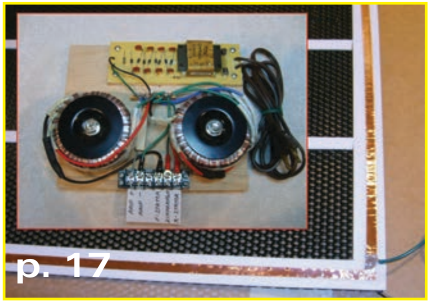

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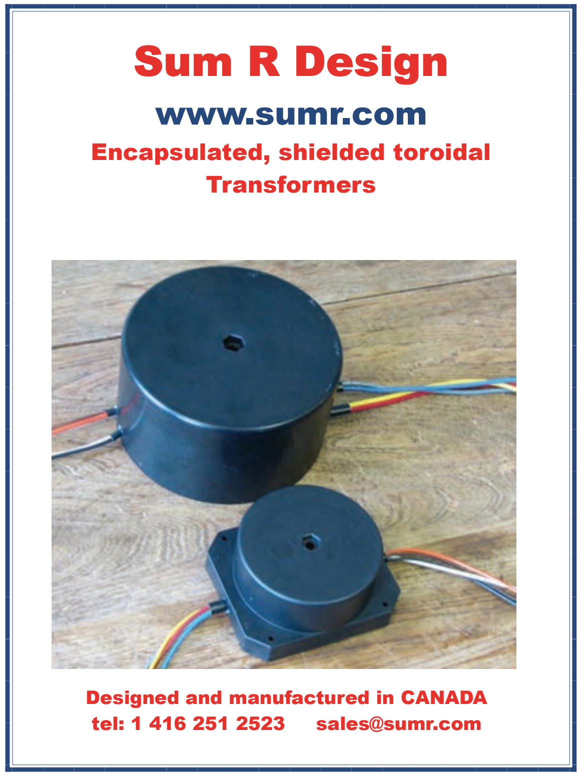
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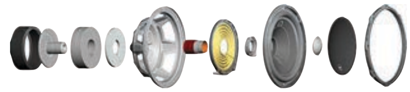
SEGMENT LLC
audioXpress | VOICE COIL
WORLD TUBE DIRECTORY | LIS LOUDSPEAKER INDUSTRY SOURCEBOOK

 By Mike Klasco and Steve Tatarunis (United States)

The Voice Coil (Part 2)



Materials and Quality

The selection and application of materials can affect sound quantity, quality, and power handling

The voice coil assembly is the heart and motor of the speaker. It consists of the bobbin, voice coil wire/winding, collar, and lead-out wires (see **Photo 1**). Essentially it is a coil of wire wound on a form (bobbin) with the beginning and ends of the coil leading out to the speaker terminals. Wire gauge, voice coil winding length, type of wire conductors (including materials and cross-sectional shape, fabrication techniques and many other parameters) are considerations for the engineer spec'ing the voice coil. Voice coil bobbin materials include polyimides (such as Kapton), aluminum and other metal foils, and composites using woven fabrics with resin coatings. Voice coil wire can have a round, square, or rectangular cross-section and the conductor can be copper, aluminum or copper-clad aluminum. Even the "dress" of how the voice coil wire lead-outs are run and the flex wire woven composition is a significant consideration. All these factors influence efficiency, power handling, distortion, and cost. We started our epic journey into voice coils last month, and in this installment we continue to explore materials and construction of voice coils, including their impact on sound quality.

SOUND QUALITY OF MATERIALS AND ADHESIVE BONDS

Loudspeaker voice coils are seldom considered critical elements that contribute to sound quality, and few technical papers have addressed this issue. When designing a voice coil, the selection and application of materials can have profound effects upon sound quantity, quality, and power handling. The mechanical energy from the winding stack moves by transconduction through the bobbin and collar before reaching the diaphragm, and any non-linearity in this path are superimposed upon the response of the speaker. Intrinsic characteristics of materials such as electrical induction of eddy currents and resulting IM distortion, tan delta, internal damping (or lack of it) and Young's modulus create specific sonic signatures, resonances due to topology, acoustic impedance damping such as vent holes. Figures of merit also include high-tensile strength, high-torsional modulus, and good damping and contribute to the residual distortion spectrum of the transducer. Even the glue joints, as well as the adhesives used to bond the magnetic structure, have an impact on sound quality—a topic which will be covered later in this series.

LEAD-OUT WIRES

This can be a whole topic in itself, but we will spare you. If you feel you would enjoy this kind of article, or a detailed article on collars (etc.) you should be reading our sister publication, *Voice Coil*. Lead-out wires are stranded and/or twisted. Damping fibers, such as cotton, Technora, and Nomex are interwoven. The "dress" of how the voice coil wire lead-outs are run from the ends of the coil to the speaker input terminals is a significant consideration (see **Photo 2**). The most expedient production solution is to have the two lead-out wires attach near each other on the bobbin, but on low-mass designs this can cause rocking. A more refined solution is to have the lead-out wires at 180°. Whipping of the lead-out wires at high excursion has been remedied by stapling at one point to the cone, or to the spider, or weaving the lead-outs into the spider itself (see **Photo 3**).

VOICE COILS FORMER MASS AND THICKNESS

In some applications, such as full-range, midrange and high-frequency drivers, a low-moving mass voice coil assembly is vital in achieving sufficient sensitivity and a smooth upper-frequency response. While a low mass is important, the wall strength and rigidity of the candidate former material must be considered carefully.

Former materials are available from 0.05 to 0.25-mm in thickness, although the vast majority of voice coil formers use 0.08, 0.1, or 0.13-mm stock, and the adhesive coating thickness is specified separately from the base material. Thinner material naturally weighs less and requires less width in the magnetic gap, preserving efficiency and magnetic damping. But, designers must

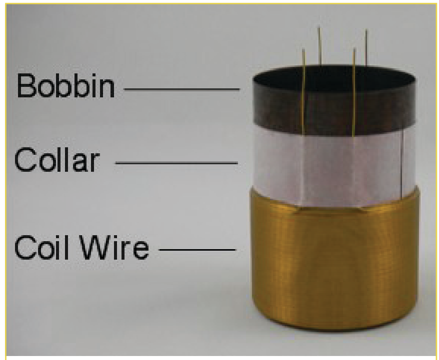


Photo 1: The construction of a voice coil (Source: Precision Econowind)

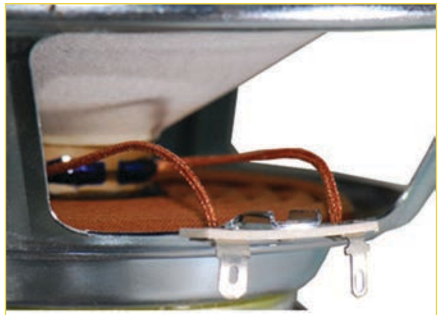


Photo 2: Lead wires attached at speaker terminals (Source: Taiwan Maiden)

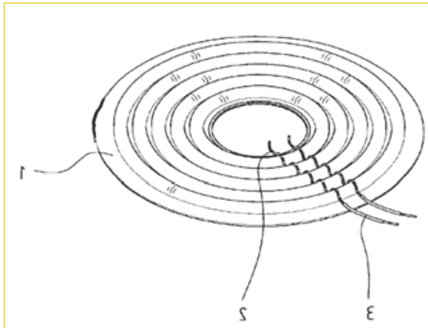


Photo 3: Lead wires woven into a spider (Source: U. S. Patent 6,810,988)

balance these benefits against their reduced rigidity.

REVIEW OF AVAILABLE MATERIALS

There was a time when paper was the dominant material in voice coils and it remains popular for low-performance formers and collars because it is very economical, moderate in mass, and available in useful gauges. Paper is highly hygroscopic, so its strength and weight are directly affected by humidity. Variations on the basic bond (rag) and Kraft (semi-chemical) pulps are common and include: "phenolic sealed," "moisture resistant," "thermally enhanced," and "fire-retardant." All of them raise the performance somewhat. Kraft, Bond, Cotton linter rag, or Press papers are used when the cost is the principal criterion. Paper is frequently used as a collar material on low-cost coils, particularly in combination with aluminum formers, but will become brittle after exposure to high temperature.

ALUMINUM, DURALMIN, AND OTHER METALS

Aluminum, in plain or black anodized finishes, together with titanium, beryllium-copper, phosphor bronze, brass and stainless steel have all been utilized as former materials (see **Photo 4**). Aluminum, first used for voice coils almost 50 years ago, is still popular, and it is a useful standard to compare performance against. With a specific gravity (SpGr) of 2.7, aluminum has almost twice the mass of polyimides or composites, while titanium (SpGr 4.5), phosphor bronze, and stainless steel (both SpGr 8.1) are much higher yet. Aluminum can be black anodized, to improve thermal emission and lower coil operating temperatures, but the largely theoretical "black body" radiating effect is barely measurable (perhaps 5%) at typical coil operating temperatures. Aluminum is economical but suffers from eddy currents. It is also heavy, yields lower Qms, lower sensitivity, suffers from self heating, and transmits heat from the coil winding into the three-way bond joint (coil, spider, diaphragm).

The principal path of heat conduction is through the thickness of metallic formers, rather than along them, and the effectiveness of their heat-sinking ability is closely

Photo 4: A voice coil with an aluminum bobbin (Source: Precision Econowind)

proportional to the proximity of the magnet's pole piece, just as it is in non-metallic formers. As the pole-to-coil clearance inside gap is narrowed, heat transfer is enhanced, and if the pole is also extended to produce a symmetrical gap, that additional height increases the heat-sinking area of the steel.

Phosphor bronze and brass have been used, presumably because they are readily available and have low electrical conductivity, but those advantages come with a cost, these materials are heavy. Stainless steel has also been used, such in a/d/s/ woofers and cone midrange drivers in the 1990s.

Sonically, aluminum requires special care in the design as intermodulation distortion (IM) results if measures such as aluminum shorting rings or copper cap on the pole piece is not implemented. Bobbin resonances are also an issue and a collar can help keep these under control. Venting the bobbin above the coil winding can provide cooling and damping of resonances—but again, venting is best used in conjunction with a collar to prevent metal fatigue and tearing between the vent holes. Other secondary acoustic effects of the vent hole is to equalize the cavity pressure at high excursions under the dust cap. But, the vent holes must be high enough above the windings so the vents are not submerged into the gap or modulation noises will result. Aluminum bobbins have been used in a number of classic woofers and tweeters, but competent design is important.

POLYIMIDES—KAPTON, APICAL, AND UPILEX

DuPont Kapton polyimide films have

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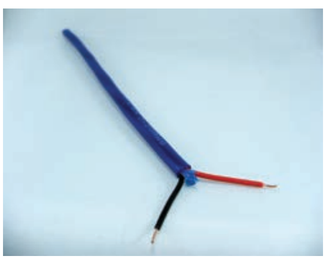

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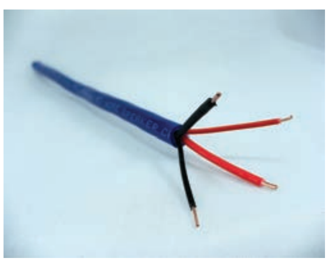

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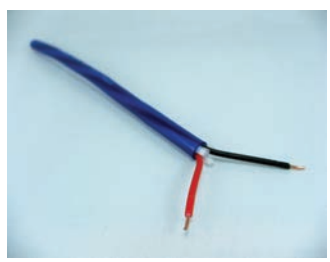

Jantzen Solid Core 6N OFC
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been used for formers for many years. Kapton was commercialized over 45 years ago, but not used on loudspeaker bobbins until years later (see **Photo 5**). Kapton HN, the classic brown Kapton film, is quite light (SpGr 1.42) giving it a good mass-to-strength ratio, and has a usefully high-tensile strength and specific modulus. Kapton is rated for use to 240°C, is an excellent thermal insulator, and is not electrically conductive. Adhesive compatibility has been a tremendous problem with Kapton. Adhesives used on Kapton must be selected and applied with great diligence, and once coated, Kapton must be stored and handled carefully to avoid adhesive flaking off. Surface treated Kapton HPP-ST provides improved adhesion, and the base film is pre-shrunk to be more dimensionally stable than HN, making it the most attractive material for formers.

DuPont expanded the Kapton line with black Kapton (MTB) which has three times the thermal conductivity of HN. The material spreads coil hot spots over a slightly larger area, and offers a better thermal path into the steel pole piece. Since MTB is black, it also improves thermal emissivity to lower winding temperatures. From a sonic perspective, one of the added "ingredients" of black Kapton also enhances damping so bobbin resonances are suppressed.

While there are quite a few low cost Chinese polyimides, most are similar to DuPont JP, which is more easily formable than the standard grades of "thermoset" polyimides—not ideal at all for voice coils. From the 1980s through the last decade, Kapton dominated the voice coil bobbin market. A combination of Kapton supply shortages accompanied by DuPont's rationing and support of markets other than the speaker industry, along with the transition of even high-power speaker manufacturing to China, has seen increased common use of Chinese polyimides. While they appear identical to Kapton at room temperature, high-power performance is mediocre as they become soft and lose their integrity below 200° C.

There are two alternatives to DuPont Kapton both based on similar polyimide chemistry. Apical AV film has been used for voice coil formers for about 10 years. It looks, feels, and functions like Kapton. Also Ube Industries manufactures Upilex, a slightly different polyimide film, with higher torsional modulus that offers good performance as a bobbin material. Kapton E is DuPont's highest performance polyimide with higher Young's Modulus and other enhanced characteristics.

NOMEX

DuPont Nomex is another popular DuPont material most often used for voice coil bobbin collars (see **Photo 6**). Nomex is an aramid, which is at the top-end of the nylon family, but the poor cousin of Kevlar. Nomex is valued for its outstanding sonic qualities due to both good Young's modulus and high internal damping. On the other hand from a manufacturing perspective, it is challenging to dry out and not dimensionally stable.

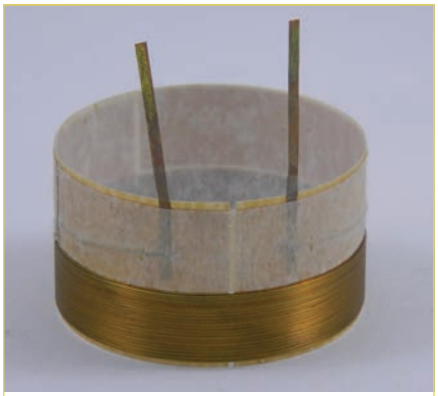

Photo 5: A voice coil with a Kapton bobbin (Source: Precision Econowind)

Photo 6: A voice coil with a Nomex bobbin (Source: Precision Econowind)

Photo 7: A voice coil with a compsite bobbin (Source: Precision Econowind)

Aside from collars, adhesive impregnated Nomex has been used for tweeters and compression drivers. Nomex has been hard to source for years as military applications absorb most of this material.

COMPOSITES

Composite former materials are typically used in woofers and subwoofers and are fabricated from a woven glass-fiber fabric, impregnated with an epoxy or polyimide resin (see **Photo 7**). Typically, they are fully cured and coated with adhesives, including TIL, a dark red, fiberglass and resin composite. TC sheet, like TIL, is a capable glass-fiber and resin composite, suitable for coating with almost any adhesive, but it is mostly commonly used with SV adhesive. Carbon fiber composite bobbin materials have been introduced as an alternative to glass composites. Toray, Hisco, and SKT offer these materials.

Alternatively, some composites are self-adhesive, using their polyimide resin to bond to the magnet wire, and to permanently thermoset the bobbin into a circular shape, unlike polyimide films that retain flat-sheet memory. This creates a coil that is exceptionally strong and eliminates coil shedding in high-output speakers. The glass fiber matrix material has a very high melting point, which ensures that the voice coil assembly exhibits high-thermal tolerance, and low loss of rigidity at elevated temperatures.

NEXT MONTH—HARD PARTS

The speaker components can be separated into the hard parts and the soft parts. (It's the soft parts that move.) Hard parts include the magnet, the magnet return structure, and the frame that all the speaker parts are assembled to. Frames can be stamped steel, cast aluminum, or plastic. Each of these can have an impact on distortion, power handling, and other factors. Recent pricing trends are forcing a move away from the compact high-energy neodymium magnets back to ferrite. The physical topologies of the various magnetic structures have an impact on sound quality including venting of air pressure. Magnetic structure design also has an impact on sound quality, from design issues such as saturation of the top plate, aluminum bobbin eddy currents creating intermodulation distortion, and remedial techniques such as a copper cap on the pole piece and shorting rings, etc.

A side bar discussion will cover both recent and past politics on speaker magnetic structure design. *aX*

 By Charlie Mimbs (United States)

Hybrid Electrostatic Loudspeakers Constructed with a Twist

Great sound with DIY ESLs

It still amazes me that an average Joe can build a speaker on the kitchen table—especially one that rivals the $20,000 high-end commercial offerings—with some perforated sheet metal, plastic film, and double-sided tape. And, building the actual transducer itself takes the cool factor to a whole new level. The "twist" in my speaker design is the unique beam-splitter bass cabinet. I am most happy to share with you my DIY electrostatic speaker project (see **Photo 1** and **Photo 2**).

My fascination with electrostatic loudspeakers (ESLs) began in 2006 when I auditioned a pair of Martin Logan Summits. Their phenomenal speed and clarity seduced me instantly, but I couldn't swing the cost to take them home. Of course, I was ruined thereafter—not even wanting to power up my old home system again. It was a dark time.

New hope came while browsing the DIY Audio forum, where I found guys rolling their own ESLs for cheap[1]. Encouraged by those brave souls, I got myself a copy of Roger Sanders' s *Electrostatic Loudspeaker Design Cookbook* and dived in.

OPERATING PRINCIPLE

An ESL is a push-pull motor driven by electrostatic attraction and repulsion. The heart of the motor is a charged plastic diaphragm suspended between two conductive screens called stators. A separate power supply puts a fixed DC charge on the diaphragm, while the music output from an audio amplifier, routed through a pair of step-up transformers, puts the driving AC charges on the stators. And the ultra-lightweight diaphragm responds with instant precision to reproduce the music with exquisite fidelity. An exploded view of my ESL panel is shown in **Figure 1**.

WARNING!

Builders beware that the charges on the stators are dangerous. If the insulating stator coatings are not essentially perfect, touching both stators while the panel is playing could stop your heart. Also, the DC power supplies should be enclosed inside the speaker cabinets. With children or pets in the home, protective speaker grills are a necessity.

THE BASICS

My speakers are 67.5" tall, 15" wide and 21" deep. A 10" woofer in a quarter-wave transmission-line cabinet provides the bass, and a 12 × 48" flat-panel ESL in an open frame provides the midrange and treble. The woofers and panels are controlled and powered separately, using an active crossover and dual stereo power amps. The frame/cabinet is red oak and 0.75" MDF, sheathed in 5-mm oak plywood and oak edge-trim. The finish is two-tone stain with a clear satin polyurethane topcoat. The woodwork was an insane amount of labor, but the results are pleasing (see **Photo 1** and **Photo 2**).

THE BEAM-SPLITTER DESIGN

The TL bass requires a huge woofer box. The challenge was cramming all that volume into a small footprint, and mating it to an ESL in the desired slimline package. The beam-splitter design allowed extending the box volume upward, behind the ESL panel. Normally, this would be a bad idea because the panel radiates sound from both sides, and placing a flat box behind it would bounce the rearward sound back to the diaphragm, a bad effect. But by forming the woofer box into a V-section beam-slitter, the rearward sound is deflected off the angled box surfaces and out the open sides of the speaker (see **Photo 2**).

The woofer is coupled to a 9', folded transmission-line of 4-cubic feet vol-

Photo 1: A view of the speaker, looking up (Source: Jason Bateham, Azelle Photography, Savannah, GA)

Photo 2: The "beam-splitter" cabinets are ready for the woofers and stat-panels. The angled box faces behind the panel opening to form the "beam-splitter."

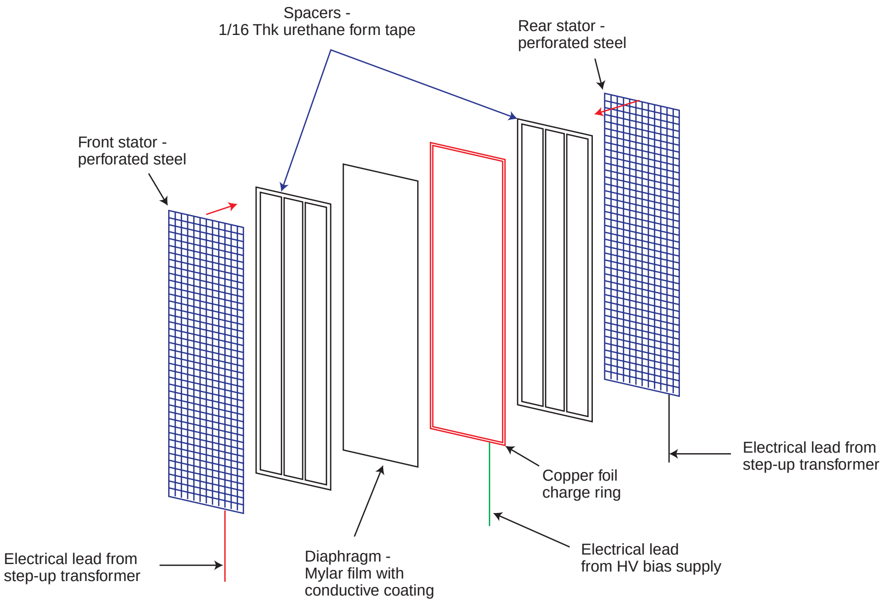

Figure 1: An exploded view of the ESL panel

ed transmission-line of 4-cubic feet volume. Behind the woofer, the line transitions to vertical in a gentle curve that prevents the back waves from rebounding to the cone, then guides the waves upward about 4', around the top bend, and then downward 4' to exit the port in phase—supporting the deep bass. Line sectional area is 125% of the woofer's cone area at the front end and tapers to 100% of cone area at the port end. The line is stuffed with polyfil (pillow stuffing) to a density of 0.5 pounds per cubic foot see **Photo 3**).

The research, design, parts searches, testing, and construction took me five months. The ESL panels and electronics were assembled from scratch in just three days, but the beam-splitter TL cabinets took over a month of evenings and weekends. My DeltaCad drawing is too extensive to show here, but is available by request through my website. The beam-splitter cabinets perform beautifully, but were a huge amount of work. Omitting the cosmetic oak sheathing and edge trim, and just painting the MDF, would save a good third of the labor.

Of course, the same ESL panel and electronics could be mated to a simpler-to-build, sealed or ported woofer box/frame design (similar to Martin Logan hybrids) and still retain that magical ESL sound and imaging.

FABRICATING PERF-METAL STATORS

I recommend 16 1to 20-gauge steel perf stators with 40 to 61% open area and hole diameters at least 2× the metal thickness, but not larger than 0.187". (Refer to OnlineMetals.com) Mine are 18 gauge/40% open with 0.125 holes, and painted with polyurethane. Powder-coated stators require at least 50% open perf, as powder coating closes up the holes more than painting. Punch-perforated metal has smoother hole-edges on one side and sharper hole-edges on the opposite side. As sharp edges are prone to coronal arcing, the side with the smoother hole-edges must face the diaphragm. Sharp points along the trimmed edges are likewise prone to arcing and should be ground smooth. Smoothing the stator edges takes about 0.125" off their initial width and length, so allow for it. Also, mating stators should be match-trimmed so that their holes align exactly.

Next, the electrical leads were attached using standard insulated 16-gauge stranded copper cable, soldered to an outside corner, using care that no bare wire extended beyond the stator edge to create a potential arc path to the mating stator. If powder coating the stators, braze on a metal tab-connector, rather than using solder.

The stators were solvent spray-cleaned with naphtha, and then spray-

Photo 3: The transmission-line cabinet, partially complete with fiber stuffing in place.

painted with 10 to 14 mils of automotive polyurethane (black base plus clear topcoats). Initial coats were misted on to build a foundation, allowing at least ten minutes between coats, followed by multiple additional coats to build thickness. It's better to spray too dry than too wet, as too wet paint will flow away from sharp edges, and that's where it's needed most. Final coating thickness was verified by caliper measurement. I used a two-component polyurethane applied with a paint gun, but practically any non-metallic paint (Krylon, e.g.)

any non-metallic paint (Krylon, e.g.) should work fine if applied to adequate thickness.

SPACERS, DIAPHRAGM-TO-STATOR SPACING, AND SUPPORT SPACING

Because electrical field-strength falls off quadratically with distance, minimal diaphragm-to-stator spacing (d/s) is essential for high output, but we still need enough room for the diaphragm to move and make sound without striking the stators (see **Figure 2**). For hybrid panels operating above 200 Hz, as little as 0.062" d/s can be used. My spacers are 3M 0.062 × 0.75" double-sided urethane foam tape. The tape bonds the diaphragm to the stator instantly with minimal fuss and greatly simplifies panel construction. The tape is applied to the stators' periphery edges as shown. My 12"-wide panels also have two 0.375"-wide center support-spacers, which divide the diaphragm area into three vertical sections. Sanders's Cookbook recommends spacing the supports at intervals equivalent to 70 to 100× the d/s. The relatively close spacing of my supports ensures that bass energy (including pressure waves off the woofer) can't drive the diaphragms into the stators at high volume.

TENSIONING THE DIAPHRAGMS

The diaphragm (see **Photo 4**) can be tensioned by shrinking the film with a heat gun (post install) or mechanically tensioned using a stretching jig (pre-install). Mine were mechanically tensioned using a bike-tube jig made from 0.75" MDF, sized 1" larger than the stator all around. The jig edges are 2" high with a 0.375" hole at one end for the tube's valve stem, and the top edges were radiused using a router and 0.75 round-over bit. Car wax or baby powder applied to the rounded edges helps the fragile Mylar film slide without snagging and tearing. The tube is a size 27 × 1.25 (700 × 32 metric) with a Schraeder valve, purchased from Wal-Mart. (Don't use a Presta valve tube. It will leak down.) Six µm diaphragms should be tensioned to 1% to 2% elongation. Mine were tensioned to 1.5% elongation as follows:

- Ready the stator for bonding and lay out a bath towel on the work surface.
- Layout the Mylar film on the towel and place the jig facedown onto the film.
- Wrap the film over the jig edges and secure it to the underside with double-backed tape.
- Attach a hand pump to the tube valve, and then turn the jig over (film side up).
- Use a fine-tip felt pen to place reference marks on the film, exactly 12" apart, in the width dimension.
- Slowly inflate the tube to stretch the film. Stop when the marks are 12 3/16" apart, for 1.5% elongation.
- Immediately place the stator on the film and press down along the tapelines to seal the bond.
- Release the tube pressure and trim off the excess film net to the tape edges.
- Over-wrap the outer 0.125" of the edges with 2-MIL polyester packing tape as shown in **Figure 2**. (It enhances the electrical insulation along the stator edges.)

COATING THE DIAPHRAGMS

We want the diaphragm conductive enough to accept a charge, yet resistive enough to slow charge migration and limit the potential current flow that would otherwise sustain any diaphragm-to-stator arcing that might occur.

Peter Baxandall showed that the ideal coating resistance is about 10^{10} Ω. The TechSpray Licron Crystal ESD coating has good resistance (10^7 to 10^9 Ω), adds very little mass, adheres well, and is unaffected by humidity.

Before spray coating the diaphragms, mask off the outermost 0.25" of their edges, because we don't want to conduct the polarizing charge out to the panel edges where it might find a leak path into a stator edge. Shake the can well, spray on one wet coat, and allow it to dry for at least three hours. It looks thick initially, but it's 98% solvents, so it flashes off and dries to a thin, clear coating, about 2-µm thick.

INSTALLING THE COPPER FOIL CHARGE RING

The charge ring conducts the polarizing voltage onto the diaphragm (see **Photo 5**). First, the lead from the DC power supply (20-gauge insulated/stranded wire) was soldered to the copper foil (off the stator). The exposed wire-end was flattened before soldering and the excess solder was carefully ground off to make the joint flat and thin, as we don't want a fat solder joint pressing into the foam tape and arcing to the underlying stator. The foil charge ring and wire-lead were then applied to the mating stator, centered on the foam

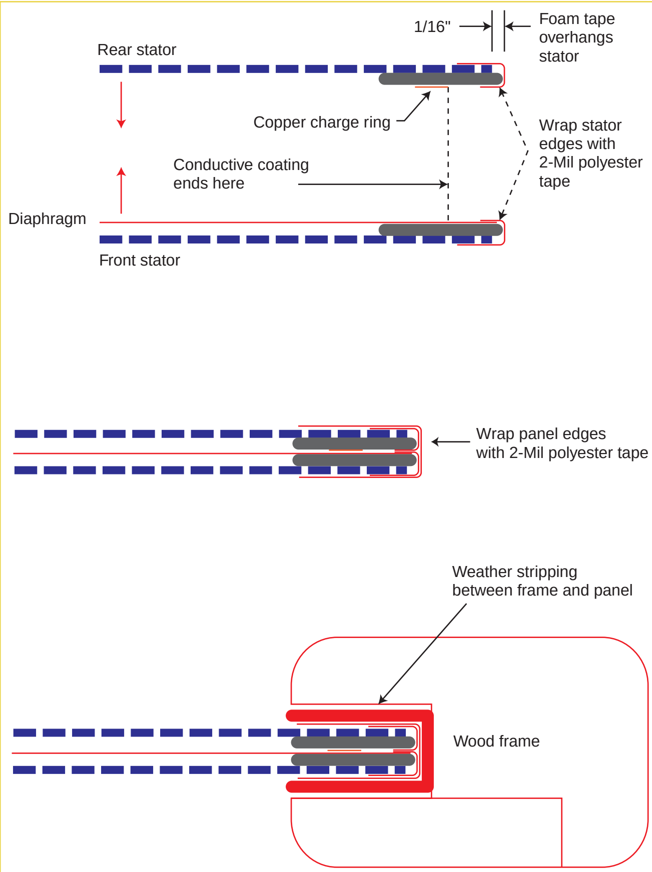

Figure 2: A section view of stator and panel edges

tape as shown.

ASSEMBLING THE ESL PANEL

Mate the stators together and press along the tapelines to secure the bond. The foam tape makes a permanent bond on contact, so the stators must be exactly aligned when contact is made. This is properly a two-person job but I managed to do it alone by standing the stators vertically, on-edge, on a worktable. After assembly, the panel edges were over-wrapped again with polyester packing tape as shown in **Figure 2**. Murphy's Law gives no quarter, but if all steps were done right, there is virtually no chance of arcing at the panel edges, which is the number one cause of failure in DIY perf-metal panels.

THE ELECTRONICS PACKAGE

The DC power supplies plug into a 115 VAC wall outlet and use a 115-V/230-V transformer in front of a diode/cap cascade; outputting about 2.7 kV (see **Figure 3**). The circuit resistors limit the potential output current to safe levels and help stabilize the charge on the diaphragm (see **Photo 5**).

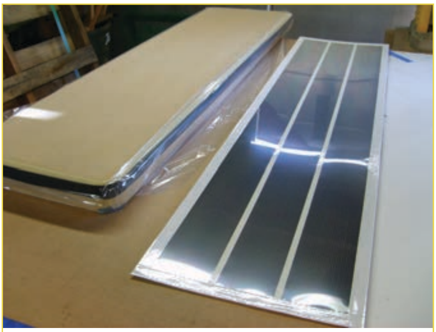

Photo 4: The bike-tube stretching jig and front stator, with a tensioned diaphragm attached

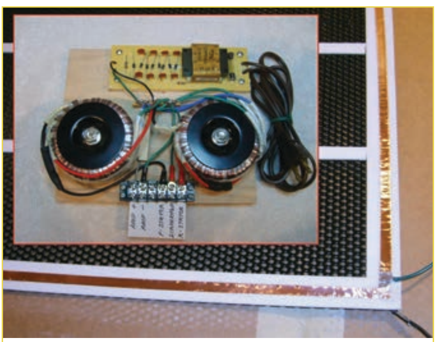

Photo 5: In the back is rear stator with a copper charge ring. In the front is the DC power supply and step-up transformers.

The generic PC board shown in the parts list was cut in half to yield boards for both supplies. To step up the voltage from the power amps, each speaker uses a tandem of 50 VA, 2 × 115 V/2 × 6 V toroidal transformers, wired for a 76:1 step-up ratio (with losses, about 68:1 in praxis).

THE CROSSOVER AND THE DIAPHRAGM RESONANCE

My speakers are vertically bi-amp'd, using a Behringer DCX2496 digital crossover feeding a pair of Carver TFM-25 power amps. ESL diaphragms have a nasty-loud resonance that typically falls between 50 to 110 Hz, depending on the tension and span between supports. My speakers' diaphragms resonate at around 80 Hz, and in order to avoid exciting this resonance, my rule of thumb is to set the crossover frequency at least one octave above the resonance with a 48 db slope, or at least two octaves above resonance with a 24-db slope. I currently use a 200 Hz/48-db crossover.

I didn't use passive crossovers with a single amp for two reasons: First, an ESL panel is a big capacitor with impedance that varies widely with frequency, which makes passive crossovers troublesome and impractical. Second, an active adjustable crossover is superior in every way and provides for easy, real-time adjustments to precisely balance the woofer and panel outputs. Believe me; it's better to spring for the extra amp and go active.

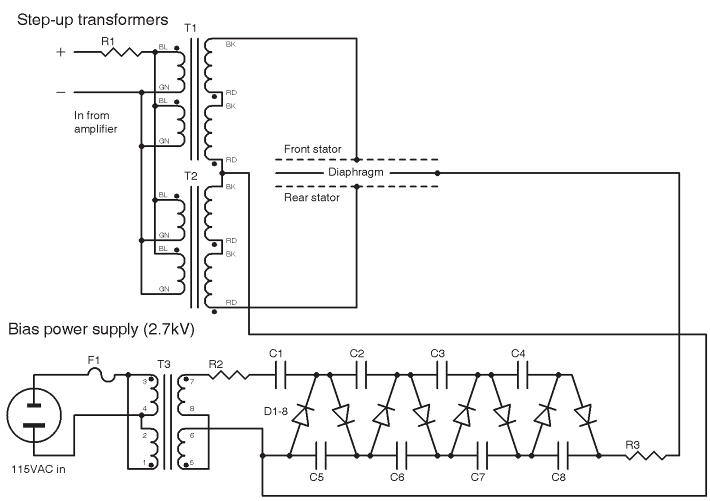

Figure 3: Step-up transformers and the bias power supply. Woofer connections not show: connect amp (+) to woofer (+) for matched phasing

Qty	Component	Source/PN
4	T1, T2:TFMR, 50 VA, 2 × 115 V/2 × 6 V	Antek #AN-0506
2	T3: TFMR 115 V/230 V	Mouser #553-FP230-25
2	R1: 1Ω, 10 W, Wire wound	RadioShack
2	R2: 3.9 kΩ	Mouser #660-CF1/2CT52R392J
2	R3: 18 MΩ (3500 V Max)	DigiKey #PPCHHJ18MCT-ND
2	F1: Fuse, 0.25 A 250 V 5 × 20 mm	RadioShack
4	Fuse clip, 5 mm	Mouser #576-01110501N
1	PCB 4.5" × 6.5" (cut for 2 pcs)	RadioShack
16	D1-8: Rectifier diode, 1 kV	Mouser #625-1N4007GP-E3/54
16	C1-8: Ceramic cap, 5 kpF/1 kV	Mouser #75-562R5GAD50
2	Power cord 2-BL Pol	Home Depot

Table 1: The electronics parts list for (2) speakers using Antek transformers

COMPENSATING THE DIPOLE PHASE-CANCELLATION

The dipole ESL panel emits sound waves from the front and back, which are exactly opposite in phase. Accordingly, any waves long enough to reach around the panel (the lower frequencies), meet at the panel edges and cancel, resulting in a null off-axis and reduced energy on-axis for wavelengths longer than about four times the panel's width (including the frame). Below this threshold, the frequency response rolls off at about 6 db/octave. For a full floor-to-celing-height, line-source panel, the dipole roll-off would be 3 db/octave near-field, transitioning to 6 db/octave far-field. This roll-off can compensated using either a 6 db/octave shelving circuit to pull down the frequencies above the threshold, or an equalizer to boost the frequencies below the threshold. I use a Behringer DEQ2496 EQ to compensate the roll off. Any decent-quality equalizer would suffice. If using a DSP crossover with parametric EQ, its EQ could be used instead. The off-axis cancellation still occurs, of course, but the now-boosted lower frequencies are projected forward enough, within the on-axis lobe, to reach and envelop the listener with balanced sound.

HOW DO THEY SOUND?

I fired these babies up on July 4, 2008 with my goddess Diana Krall singing Nat King Cole's "You're Looking at Me." The illusion of Diana performing live in my living room was so scary-real, I could almost smell her perfume. Finally, I have speakers worthy of Diana, who now sounds as good as she looks.

The electrostats give jaw-dropping clarity and realism, and the transmission-line bass is full and tight, with gorgeous tone. After listening to these speakers for three years now, I'm still amazed. No other speakers I've ever heard can match their imaging. They are, however, ultra-directional, with a tightly focused sweet spot only about 18" wide—so, not a good choice for party speakers. Moving out of the sweet spot, the highs fall off a cliff and the sound becomes bass-heavy. Within the sweet spot there is perfect frequency balance, surreal imaging, exquisite detail, and stunning speed. My friend Jason described them as "remote headphones." A woman's voice through these speakers just takes your breath away.

BUILD YOUR OWN

With earnest study, average skill, and encouragement from the great community of DIYers on the web, it really is possible to build your own ultra-high fidelity ESLs right in your own garage, without breaking the bank. And the satisfaction of building magnificent speakers with your own hands is priceless. Even more, for me, is the huge kick I get whenever someone sees my project web page and is inspired to dive in and build their own ESLs— because I know what the effort and the results will do for them. *aX*

Editor's note: Charlie Mimbs provided a complete parts and components list that is available at www.audioamateur.com.

RESOURCES

Forum: DIY Audio www.diyaudio.com/forums/planars-excotics/

Jazzman's DIY Electrostatic Loudspeaker page www.jazzman-esl-page.blogspot.com/

M. Rehorst, www.mark.rehorst.com/ESLs/index.html

P. Baxandall, *Loudspeaker and Headphone Handbook*, edited by John Borwick, Focal Press, 2001.

R. Sanders and B. McClune, *The Electrostatic Loudspeaker Design Cookbook*, Audio Amateur Press Publishers, New Hampshire, 1993.

S. Stokes's ESL 1.0 project,
www.quades1.com/diy_es11.shtml

SOURCES

TechSpray Licron Crystal ESD
Allied Electronics | www.alliedelec.com

16-to-20-Gauge steel perf stators
McNichols Perforating |
www.mcnichols.com

Fountek's 3" Full Range

A small high-end home audio driver

The sample I'll cover in this month's Test Bench review is a 3" full-range from Fountek, the FR89EX. Fountek Electronics is a relatively new high-end audio manufacturer out of China. Founded April 2003 in Shanghai, China, the company released its first ribbon transducer in June of that year. Besides OEM transducers like the FR89EX that is the subject of this review, Fountek also produces finished home audio speakers, a line of solid state and tube amplifiers, plus a line of powered studio monitors.

The Fountek FR89EX is 3"-diameter aluminum cone neodymium motor full-range transducer built on a four-spoke, cast aluminum frame. For cooling, the proprietary frame has four 2 mm × 35 mm vents below the spider-mounting shelf, located between the frame spokes. The cone assembly consists of a silver (natural finish) coated aluminum cone with a 1"-diameter coated aluminum dust cap. Suspending the cone and dust cap are a low-loss NBR surround and a treated cloth spider that has been vented with four 4-mm diameter holes. The FR89EX has a 25-mm diameter, single-layer voice coil wound with flat (ribbon) copper wire on aluminum former and terminated to a pair of gold-plated terminals. Powering this structure is an under-hung neodymium motor with a black emissive coating on the motor return cup for better heat transfer to the surrounding air.

I commenced testing the Fountek FR89EX full-range using the LinearX LMS analyzer and VIBox to create both voltage and admittance (current) curves with the driver clamped to a rigid test fixture in free-air at 0.3 V, 1 V, 3 V, 6 V, and 10 V. The 10-V curve was too non-linear to get an appropriate curve and was discarded. Next, the eight 550-point stepped sine wave sweeps (four current and four voltage sweeps) for each of the FR89EX samples were post-processed. The voltage curves were divided by the current curves (admittance curves) to produce the impedance curves, phase generated by the LMS calculation method, and along with the accompanying voltage curves, imported to the LEAP 5 Enclosure Shop software. Since most Thiele-Small data provided by OEM manufacturers is being produced using either a standard transducer model or the LEAP 4 TSL model, I additionally created a LEAP 4 TSL model using the 1-V free-air curves. The complete data set, the multiple voltage impedance curves for the LTD model (see **Figure 1** for the 1-V free-air impedance curve) and the 1-V impedance curve for the TSL model were selected in the transducer derivation menu in LEAP 5 and the parameters created for the computer box simulations. **Table 2** compares the LEAP 5 LTD, TSL data, and Fountek factory parameters for both of the FR89EX samples.

LEAP parameter calculation results for the FR89EX were sufficiently close to the factory data, so I proceeded to set up computer enclosure simulations using the LEAP LTD parameters for Sample 1. Two computer box simulations were programmed into LEAP, a 30-cubic-inch sealed box alignment (50% fill material) and a 42-cubic-inch vented alignment tuned to 103 Hz (15% fill material). **Figure 14** displays the results for the Fountek full-range driver in the two enclosure simulations at 2.83 V and at a voltage level sufficiently high enough to increase cone excursion to Xmax + 15% (5.8 mm for the FR89EX). This produced a F3 frequency of 165 Hz with a box/driver Qtc of

a)

b)

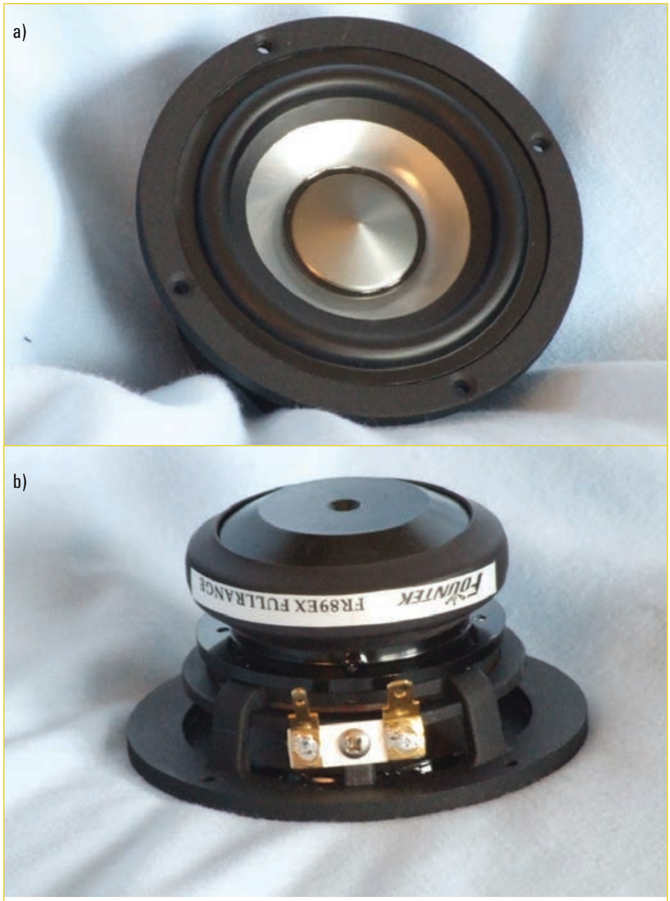

Photo 1a: The top of the Fountek FR89EX. b: The bottom of the FR89EX

	TSL model		LTD model		Factory
	sample 1	sample 2	sample 1	sample 2	
Fs	101 Hz	101 Hz	101 Hz	103 Hz	96 Hz
Revc	3.20	3.21	3.20	3.21	3.40
Sd	0.0027	0.0027	0.0027	0.0027	0.0027
Qms	1.16	1.08	1.26	1.21	1.47
Qes	0.55	0.51	0.57	0.59	0.65
Qts	0.37	0.35	0.40	0.40	0.45
Vas	1.01 ltr	1.02 ltr	1.07 ltr	1.03 ltr	1.11 ltr
SPL 2.83 V	84.7 dB	85.0 dB	84.4 dB	84.4 dB	83.7 dB
Xmax	5.0 mm	5.0 mm	5.0 mm	5.0 mm	5.0 mm

Table 1: Fountek FR89EX full-range

0.71 for the 30-cubic-inch sealed enclosure and –3 dB = 103 Hz 42-cubic-inch vented box simulation. Increasing the voltage input to the simulations until the maximum linear cone excursion was reached resulted in 104 dB at 21 V for the sealed enclosure simulation and 104.5 dB with a 21-V input level for the larger vented enclosure (see **Figure 3** and **Figure 4** for the 2.83-V group delay curves and the 21-V excursion curves).

Klippel analysis for the Fountek 3" full-range (our analyzer is provided courtesy of Klippel GmbH), and testing performed by Pat Turnmire, Red Rock Acoustics (author of the SpeaD and RevSpeaD software) produced the Bl(X), Kms(X) and Bl and Kms Symmetry Range plots given in **Figures 5** to **8**. The Bl(X)

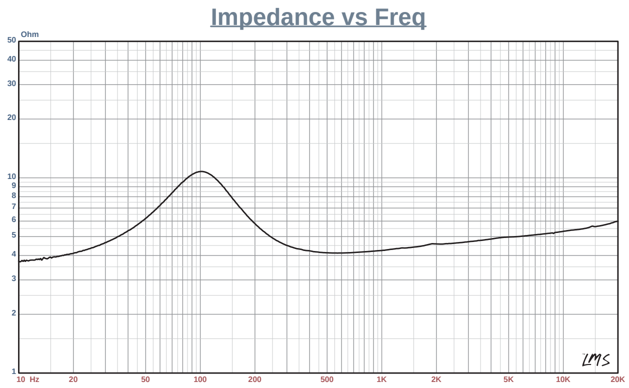

Figure 1: The Fountek FR89EX free-air impedance plot

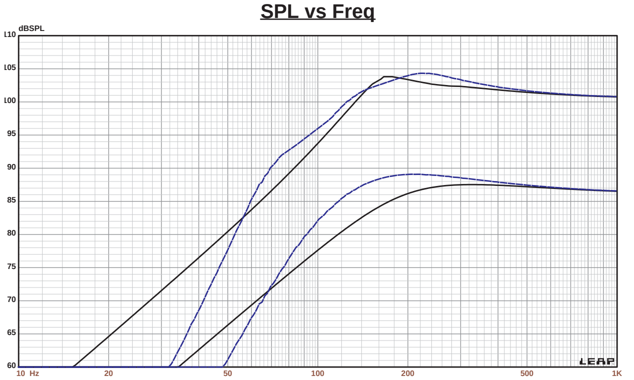

Figure 2: The Fountek FR89EX computer box simulations (black solid = sealed @ 2.83 V; blue dash = vented @ 2.83 V; black solid = sealed at 21 V; blue dash = vented at 21 V)

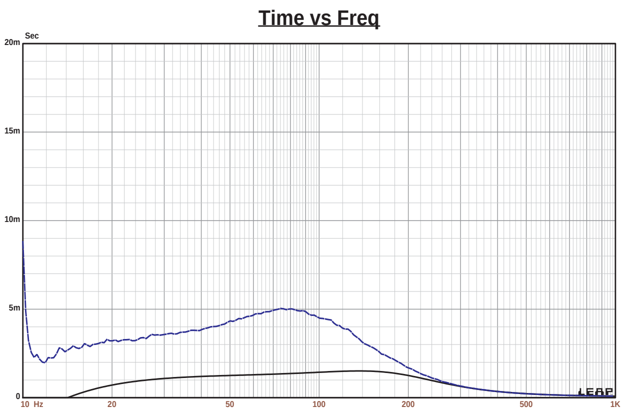

Figure 3: Group delay curves for the 2.83-V curves in Figure 2

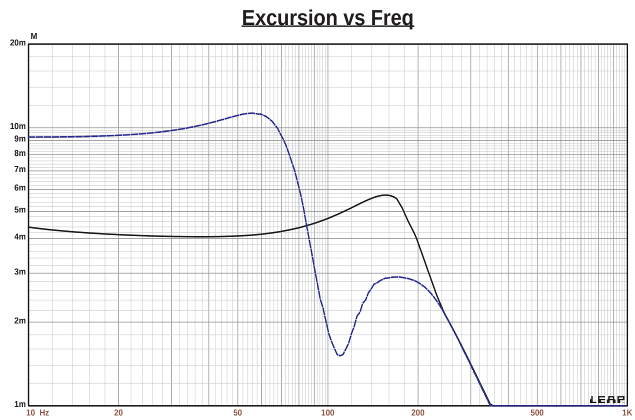

Figure 4: Cone excursion curves for the 21 -V curves in Figure 2

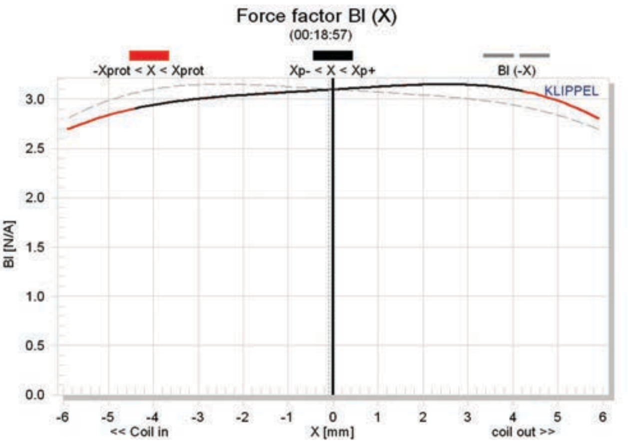

Figure 5: Klippel Analyzer Bl (X) curve for the Fountek FR89EX

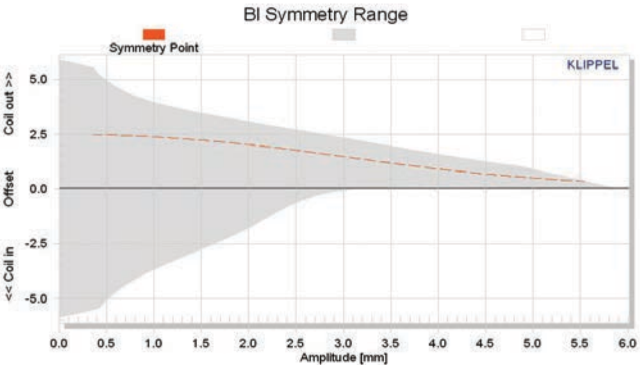

Figure 6: Klippel Analyzer Bl symmetry range curve for the Fountek FR89EX

curve for the FR89EX (see **Figure 5**) is relatively broad and with a degree of asymmetry. Looking at the Bl Symmetry plot (see **Figure 6**), this curve shows a 2.5-mm coil-out offset at the rest position that decreases to 0.50 mm at the physical 5.0 mm Xmax of the driver.

Figures 7 and **8** show the Kms(X) and Kms Symmetry Range curves for the Fountek full-range driver. The Kms(X) curve is also somewhat asymmetrical, and has a very minor forward (coil-out) offset of about 0.3 mm at the rest position, decreasing to zero at 2.5 mm, and ending up at 5-mm excursion with about 0.6-mm coil-in offset. Displacement limiting numbers calculated by the Klippel analyzer for the FR89EX were XBl @ 82% Bl is 4.2 mm and for XC @ 75% Cms minimum was 2.9 mm, which means that for this Fountek woofer, the compliance is the most limiting factor for prescribed distortion level of 10%.

Figure 9 gives the inductance curve Le(X) for the FR89EX. Inductance will typically increase in the rear direction from the zero rest position as the voice coil covers more pole area; however,

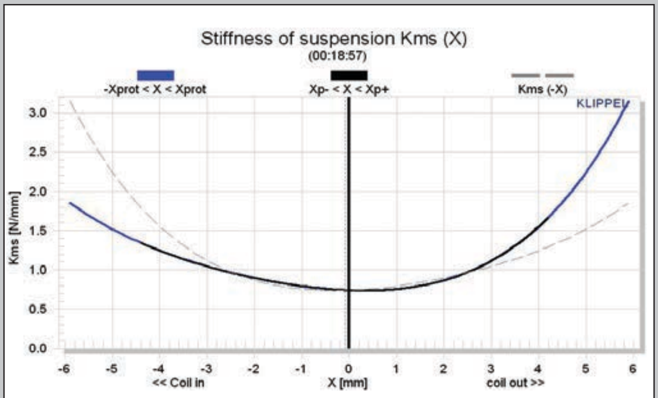

Figure 7: Klippel Analyzer mechanical stiffness of suspension Kms (X) curve for the Fountek FR89EX

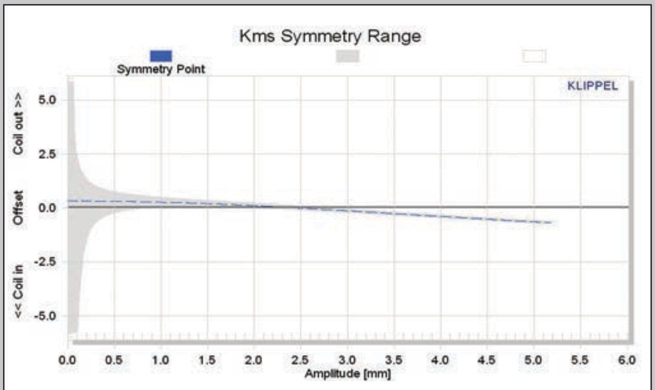

Figure 8: Klippel Analyzer symmetry range curve for the Fountek FR89EX

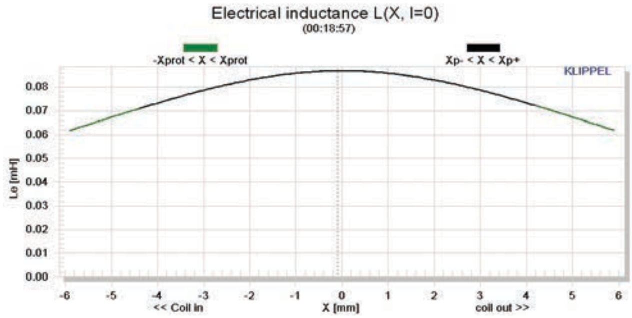

Figure 9: Klippel Analyzer Le (X) curve for the Fountek FR89EX

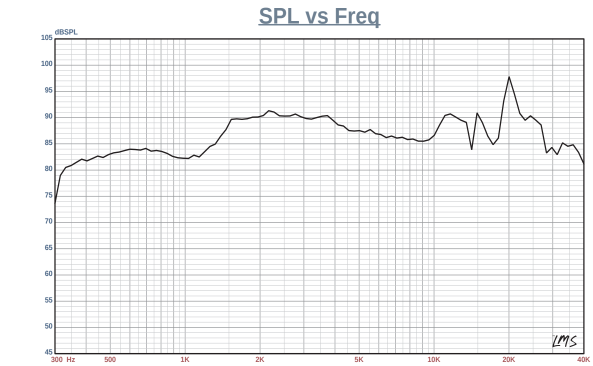

Figure 10: Fountek FR89EX on-axis frequency response

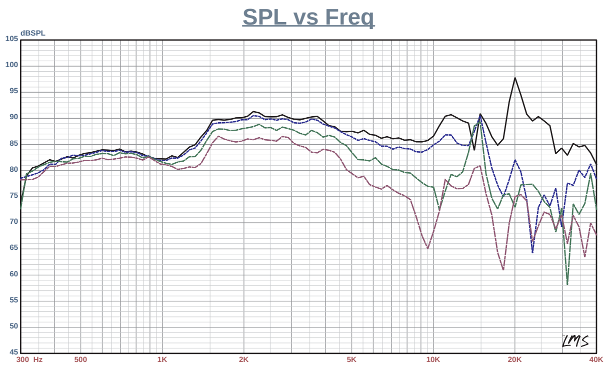

Figure 11: Fountek FR89EX on- and off-axis frquency response at 0° (black-solid), 15° (blue-dot), 30° (green-dash), and 45° (purple-dot/dash)

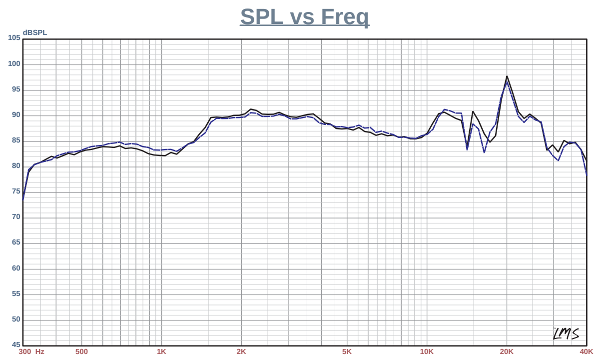

Figure 12: Fountek FR89EX two-sample comparison

the FR89EX inductance variance due to the neodymium motor structure is rather small. The inductance variation is only 0.09 mH from the in and out Xmax positions, which is very good.

Next, I mounted the FR89EX full-range in an enclosure which had a 9"× 4" baffle, filled with damping material (foam), and then measured the transducer on- and off-axis from 300 Hz to 40-kHz frequency response at 2.83 V/1 m using the LinearX LMS analyzer set to a 100-point gated sine wave sweep. **Figure 10** gives the FR89EX's on-axis response indicating a smoothly rising response to about 10 kHz with a 6-dB step between 1 kHz and 1.5 kHz, and some break-up peaking at 20 kHz prior the high-pass roll-off. However, it definitely has a response on-axis out to 35 kHz.

Figure 11 displays the on- and off-axis frequency response at 0°, 15°, 30°, and 45°. As with most full-range transducers, the 30° off-axis is about the same as any 3" piston driver. The last SPL measurement is given in **Figure 12**, and illustrates the two-sample SPL comparison for the 3" driver, showing a close match to within less than 1 dB throughout the operating range.

For the remaining series of tests, I employed the Listen, Inc. SoundCheck analyzer with the Listen, Inc. 0.25" SCM microphone and power supply (courtesy of Listen, Inc.) to measure distortion and generate time frequency plots. For the distortion measurement, the Fountek fullrange was mounted rigidly in free-air, and the SPL set to 94 dB at 1 m (6.4 V) using a noise stimulus, and then the distortion measured at with the microphone placed 10 cm from the dust cap. This produced the distortion curves shown in **Figure 13**. I then used SoundCheck to get a 2.83 V/1 m impulse response for the driver and imported the data into Listen, Inc.'s SoundMap Time/Frequency software. The resulting CSD waterfall plot is given in **Figure 14** and the Wigner-Ville (for its better low-frequency performance) plot in **Figure 15**.

The Fountek FR89EX is a well-designed 3" full range and possible applications would include use as a driver in a high-end iPod speaker, as a midrange in a three-way speaker, are in high-end line-source speaker.

For more information, visit the Fountek website at www.fountek.net. *aX*

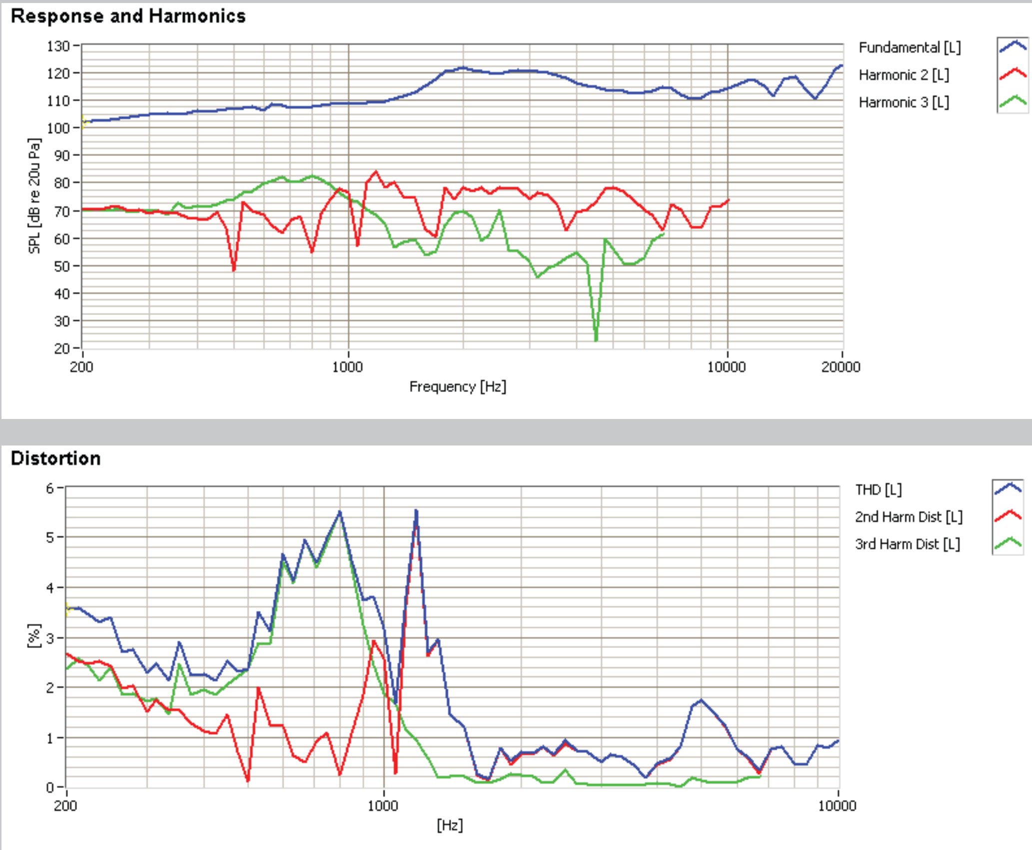

Figure 13: Fountek FR89EX SoundCheck distortion plots

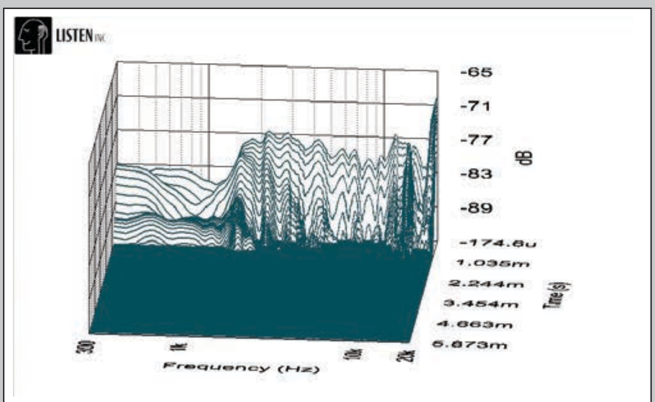

Figure 14: Fountek FR89EX SoundCheck CSD waterfall plot

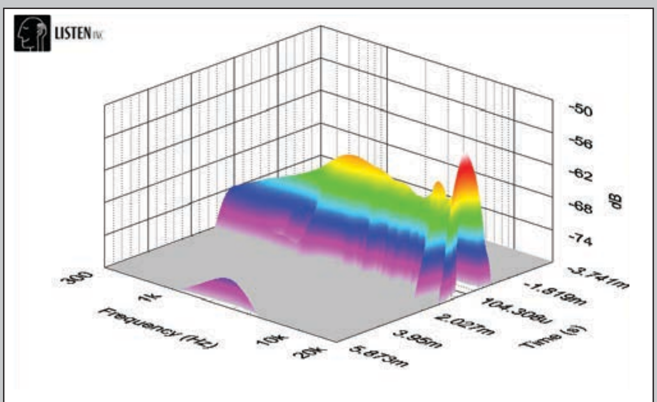

Figure 15: Fountek FR89EX SoundCheck Wigner-Ville plot


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book review By Richard Honeycutt (United States)

Fundamental Amplifier Techniques with Electron Tubes

A 21st-century perspective on hollow-state design

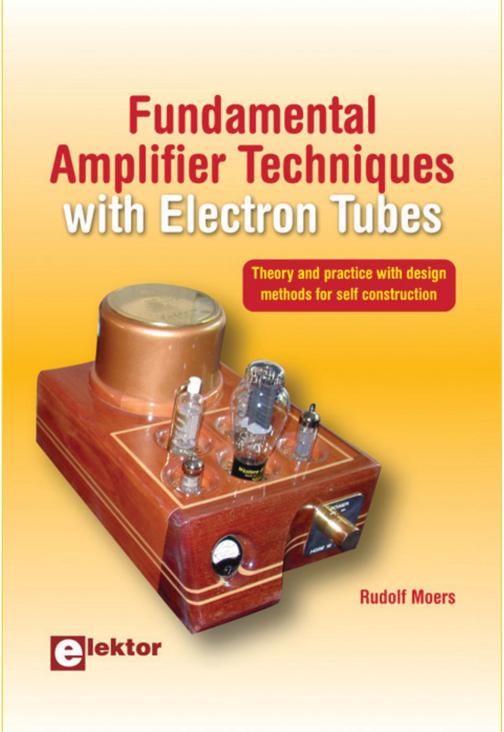


Photo 1: A reference book that focuses on the science of hollow-state design as applied to amplifiers and power supplies.

Back in the 1950s and 1960s, if you wanted to learn about vacuum tube amplifiers, you could read the *Radiotron Designer's Handbook*, a 1,500-page behemoth that covered all kinds of vacuum tube circuits that were known at the time, and also included abundant information on passive components as well. Or you could use the introductory material and example schematics in the *RCA Receiving Tube Manual*—much shorter and less expensive, and also far less comprehensive. Of course, it did include data on most tubes then being manufactured by RCA. If you just wanted to build your own amplifiers, but were not interested in designing, there was the *Mullard Circuits for Audio Amplifiers*. For a more scholarly approach, you could check out an electrical engineering textbook such as *Analysis and Design of Electronic Circuits* by Paul Chirlian.

Now, however, things are different. Although some of these references can be found on the Internet, they are no longer up-to-date. Happily, however, Elektor recently published *Fundamental Amplifier Techniques with Electron Tubes* by Rudolf Moers, which presents a 21st-century perspective on the science of hollow-state design as applied to amplifiers and power supplies. Beginning with the principles of electron emission, the book progresses through standard vacuum tube varieties: diodes, triodes, tetrodes, and pentodes, after which it covers such general principles as frequency dependent behavior, nonlinear distortion, noise, and negative feedback. The book concludes with a chapter on the construction of electron tube amplifiers. Unlike many of the earlier authors of books on electron tubes, Moers is not constrained by a need to cover such specialized tubes as pentagrid converters, or circuits specifically used in radio and TV receivers. Instead, he uses his 800 pages to discuss the physics underlying electron tube operation far more comprehensively than did any of his predecessors. He does this in a way that maximizes presentation of principles while minimizing unnecessary mathematics. In many cases, the physical explanations can be skipped over by those whose only interest is design methods. For the reader who does take advantage of the physical explanations, Moers's inclusion of an eight-page listing/definition of mathematical symbols makes the explanations easy to follow.

The focus is by no means primarily on physics, however. None of the classic texts provides anything like so comprehensive coverage of the design and operation of half- and full-wave rectifier/filter circuits, or vacuum tube phase shifters, to mention a couple of examples.

Moers's book assumes that the reader is familiar with basic DC and AC circuit theory, and therefore does not undertake the task of educating those who lack this understanding. The book is written from a scientific perspective in that, while mentioning the disconnect between measured and perceptual performance of an amplifier, the author makes no dogmatic claims about the relationship between the two, other than to opine that most of the "tube sound" results from harmonic distortion components that some people find pleasing to the ear. (Having followed this discussion for about four decades, your reviewer partially concurs, but believes that there are other elements involved as well.) The author lightheartedly introduces the quantity "cm² of gooseskin/watt" as an example of a measurement of perceptual phenomena.

A consequence of Moers' scientific approach is that specific catch phrases found in many amateur-oriented publications on tube technology are conspicuously absent. For example, it is difficult to read much about tube power amplifiers without noticing mention of the "Williamson amplifier." This circuit was developed by D. T. N. Williamson and described in articles in *Wireless World* in April and May, 1947. It was unique in that it applied negative feedback around the entire amplifier, including the output transformer, thus reducing nonlinear distortion. Doing this required very careful design to ensure stability, including the elimination of interstage transformers such as the phase splitter transformer used in many prior designs.

Moers does not mention the Williamson amplifier by name, but the vacuum tube phase splitter design Williamson used is discussed in detail in the book, as is the method of designing a negative feedback loop encompassing the entire amplifier. Moers also gives a unique explanation of another pivotal power amplifier circuit: the ultralinear circuit invented in 1951 by Hafler and Keroes. It's a case of content versus jargon.

In his otherwise excellent discussion of damping factor, Moers unfortunately makes the all-too-common error of ignoring the effects of voice coil and lead wire resistance. He gives the common equation for damping factor: DF = (loudspeaker impedance)/(amplifier output impedance). Since the amplifier (modeled as an AC generator or Thevenin source), voice coil resistance, lead wire resistance, voice coil inductance, and reflected mechanical impedance form a series circuit whose actual damping is influenced by all elements, the lead wire resistance and voice coil resistance cannot be ignored. In fact, they can easily swamp the effects of the amplifier output impedance, at least for a pentode stage using negative feedback. However, Moers does not make the further error of insisting that the damping factor be a minimum of 100 as have some earlier authors. Using an 8-Ω speaker having about 6-Ω DC resistance, the effect of a combined output impedance and lead wire resistance less than 0.5 Ω is negligible.

Two shortcomings of *Fundamental Amplifier Techniques with Electron Tubes* are more or less linguistic. English may well be the only Germanic language in which the verb in a sentence is not at the end of the sentence required to come. Thus syntactical intrusions from the author's native language sometimes make the text difficult for native English speakers. Also, Moers has chosen to use terminology that is probably not standard in English (at least American English) books on electronics. For example, he uses the term " anode static steepness" to denote "transconductance" (also commonly called "mutual conductance.") A common-cathode (or "grounded-cathode") amplifier stage is called a "basic cathode" stage in Moers' book.

These three small complaints pale in the face of the outstanding job the author has done in bringing together the theory, design, and practice of vacuum tube amplifiers in a single volume. Anyone who wants to go beyond the Heathkit level of tube amplifier understanding owes it to him/herself to buy and study this excellent volume. aX

Innovation in Transformer Design

Shannon Becker discusses the family business with a second-generation audio transformer designer

SHANNON: Where do you live?

PER: I live in the small town of Norrtälje, 45 miles north of Stockholm, Sweden.

SHANNON: You are currently the CEO of Lundahl Transformers. This is your family business. Can you tell us a little about the company's history?

PER: The company was started by my parents in their garage around 1958. The first decade or so, most transformers were manufactured for use in Swedish military equipment, which was a relatively large volume application at that time. In the late 1960s, we started making audio transformers for the state-owned Swedish Radio and a few years later also for some UK audio companies.

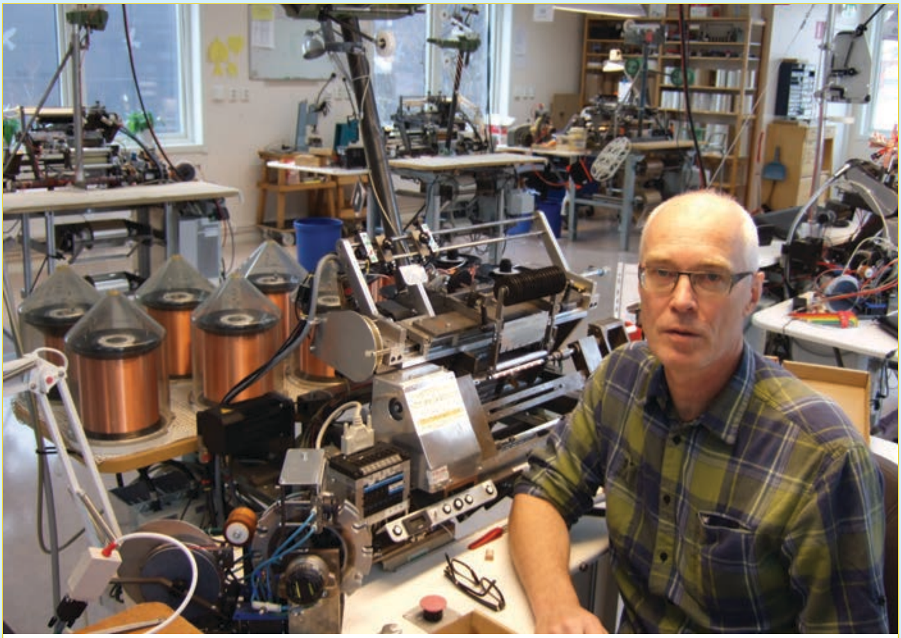


Photo 1: Per Lundahl at work in front of a winding machine

SHANNON: Are your parents still involved in the day-to-day operation of the company?

PER: My parents live in an apartment in the same building as the factory. My father is 82 years old and spends a couple of hours every day in the factory. However, he does not contribute much to the operation anymore because of his Parkinson's disease. My mother still participates in our weekly production planning meetings.

SHANNON: Can you tell us about the company's location? How did it end up in Norrtalje, a popular seaside resort?

PER: When the company was started, we lived in Lidingö, a small town very close to Stockholm. But when my parents started looking for a free lot to build a factory, the Lidingö community was uninterested, as they did not want to have any industry. After scouting a number of towns in the Stockholm area, we ended up in Norrtälje

SHANNON: Lundahl Transformers is known to be one of the world's leading manufacturers of high-performance audio transformers. What was the company's first audio transformer success story?

PER: When stereo was introduced in the late 1950s, it was suggested that the TV set could be used for one channel, and the radio set could be used for the other channel. But as a TV set is full of high-voltage circuits, an isolation transformer was required on the audio input of the TV. Our first audio transformer was such an audio line input transformer for AGA television sets. From an audio point of view it was not a very good transformer, and I think a very small percentage of these transformers were ever used as intended.

The transformers developed for Swedish Radio about a decade later were much better audio transformers, even by today's standards, and they remained in production until 1995.

SHANNON: You have a master's degree in applied science. How did you come to join the family business and how do you think your prior work experiences have prepared you for this chapter in life?

PER: When graduating from the School of Physics at the Royal Institute of Technology (KTH) in Stockholm, I had no intention of working in my parents' company. I worked as a computer consultant at Ericsson and at a spin-off company from KTH. But after my two sons were born, I and my wife, Gunilla, decided that we did not want to raise our children in Stockholm. So, we decided to move away from the city. I asked my parents if I could come work for them in Norrtälje, and they invited me to join them.

For me, it was important to have a career of my own before joining the family business. I brought, among other things, knowledge on how to use computers in industrial applications. This knowledge has been very useful when modernizing our production system.

SHANNON: Do you do any of the design work at the company?

PER: I design all our new transformers.

SHANNON: What is your most popular audio product?

PER: Our audio market is divided into two segments, Pro Audio and Audiophile. The Pro Audio segment includes transformers for microphones, mic pre-amps, splitters, distribution amplifiers, and other studio equipment. The Audiophile segment is transformers for MC phono cartridge step-up and for tube and solid state amplifiers.

Our biggest selling products are two types of transformers for microphone preamplifier inputs. In the Audiophile domain, our tube amplifier interstage and line output transformers are popular.

SHANNON: Do you have any new projects on tap?

PER: We constantly develop new transformers based on the requests of our customers. Presently we are developing an auto-transformer for a Chinese company and an interstage/line output transformer for some European customers. The latter will probably be added to our range of standard transformers, available to everyone.

For the very fastidious audiophile, we are also introducing silver wire in some of our transformer types. Initially, the wire will mainly be in our high-end MC transformers, but depending on the response, it is possible that we will extend the silver wire product range.

SHANNON: To what do you attribute your company's continuing success?

PER: I think few other transformer companies have developed the manufacturing process as much as we have. Based on our strong ideas how a good audio transformer should be built, with internal structure, internal insulation, winding technique, etc., we have built our own machines for winding, shielding, housings and cores. We have done this not because we particularly enjoy building manufacturing machines, but because no one else builds these machines.

Photo 2: A cart full of ready-to-ship tube amp transformers

I think that most other transformer manufacturers accept the limits imposed by what is commercially available in terms of parts, material, and machines. This is also why transformers from other manufacturers look similar, and why our transformers stand out. Our manufacturing process not only contributes to the excellent sonic results that our transformers have a reputation for, but to an unusual degree of flexibility, by readily enabling us to make all winding ends available for the user. This gives most of the audio transformers in our product line the ability to be used for multiple applications.

SHANNON: Where do you see the audio industry being in 10 years?

PER: If we are lucky, today's young people, who consume an enormous amount of music through MP3 players and streaming radio, will develop a more sophisticated listening preference and request both well recorded and well-reproduced sound.

SHANNON: What do you see ahead for Lundahl Transformers?

PER: Good transformers are expensive components and will continue to be used only by the high end part of the market, be it Pro Audio or Audiophile. I hope that there will be a market for these kinds of products in the future, and I expect we will continue to service that market.

SHANNON: Do you have any advice for *audioXpress* readers who are considering building their own amplifiers?

PER: If it is your first, start with something uncomplicated. Ignore cosmetics initially and build on top of a wooden board. Once the amplifier is working, you can consider cosmetics, such as a nice enclosure, or you can start tweaking it.

SHANNON: There has been a long-running debate between tube and solid state amplifier aficionados? Do you have a personal preference, and if so, why?

PER: I don't pretend to be an audiophile. I design transformers from an engineering approach, and leave it to my customers to do the listening.

When I visit audio exhibitions I hear a lot of what I consider nice sound in many rooms, some with tube amplifier equipment and some with solid state equipment. But considering the price tags of most of these systems, what else can you expect?

SHANNON: What other projects are you currently working on?

PER: Apart from the new transformer designs I mentioned above, we are presently working on an improved process for slitting amorphous foil. We are also building a new machine for Faraday shield insertion, working on a new website, and preparing for the Musik Messe in Frankfurt.

And, we are considering advertising again for the first time in many years.

SHANNON: If you could design/build your personal dream audio system, what would it include and why?

PER: As I am not an audiophile, I think I would like to pass on this question. But as I am also a transformer manufacturer, my dream system of course contains lots of chokes and lots of transformers. aX

By Gary Galo (United States)

AES 2011

Details of a Full Technical Program

A review of audio technology, briefs, and presentations

In *audioXpress* February 2012, David J. Weinberg gave an insightful overview of the 131st AES 2011 convention that took place last October in New York City (see **Photo 1**). He carefully outlined the presentations, events, and exhibits. In this article, I'll detail my personal experiences at the convention and describe some of the audio-related conversations I had with attendees. I'll also wrap up the information about a few specific presentations that *audioXpress* readers will find intriguing.

INDUSTRY EXPERTS

In 2010, Jan Didden, longtime author for *audioXpress*, *Audio Amateur,* and *Audio Electronics*, and currently a technical editor for *audioXpress*, launched a new periodical titled, *Linear Audio*, and he was on hand to introduce his new publication to attendees. *Linear Audio* (www.linearaudio.net) is devoted to technical articles on all aspects of audio equipment design, authored by some of the most respected authorities in the field. Three issues have been published so far. I authored a guest editorial for Volume 2 on the evolution and current state of digital audio, and I am, indeed, humbled to be in such distinguished company.

Sharing a booth (see **Photo 2**) with *Linear Audio* magazine was Cordell Audio (www.cordellaudio.com), a company run by an engineer well-known to long-time attendees at AES conventions, Bob Cordell. Bob has spent his career in the telecommunications industry, specializing in integrated circuit design and fiber optic communications systems. But, he's had a lifelong passion for audio since he began building vacuum tube amplifiers as a teenager, and that interest recently culminated with the publication of his book *Designing Audio Power Amplifiers* (McGraw Hill, 2010).

Bob spent about 2.5 hours with me discussing some of the issues he's been exploring in the area of moving magnet phono pre-amp design. While most designers would agree that accurate RIAA equalization is crucial in any phono pre-amp, Bob has begun exploring issues beyond the phono equalization circuits themselves. Cartridge loading has an important impact on the final frequency response of any phono cartridge/phono pre-amp combination, but calculation of the optimum loading capacitance and resistance for a given cartridge is normally based strictly on the cartridge's electrical specifications. Mechanical issues, particularly cantilever resonances, are often not part of the equation when determining optimum cartridge loading. One idea Bob shared with me involves using the cartridge resistive loading with the cartridge inductance to provide the 75 µs time constant (Lipshitz's F5, the treble transition point between the constant-velocity and constant-amplitude portions of the RIAA recording characteristic). This eliminates the role of the electrical resonance created by the cartridge and its loading. This would require customizing the resistive loading for each phono cartridge, which he concedes may be impractical. And, while the electrical calculations are fairly manageable, bringing the cartridge's mechanical parameters into the equation complicates matters considerably. Ever since Stanley Lipshitz's 1979 paper, "On RIAA Equalization," was published in the *AES Journal*, the math for designing accurate RIAA equalization circuits has been widely available. But the Lipshitz math only ensures that the phono pre-amp's RIAA equalization by itself is correct. It does not take into account the characteristics of the cartridge (or the tonearm or turntable, for that matter). Since the Lipshitz paper was published, there's been a general consensus in audio circles that RIAA accuracy of ±0.1 dB from 20 Hz to 20 kHz will eliminate, or at least minimize, frequency response errors as a cause of audible differences between phono pre-amps. Though designing a conventional phono pre-amp with this level of accuracy is not difficult, Bob would like to achieve such performance using real-world phono cartridges and a test record, which is an enormously difficult task.

Photo 1: AES attendees await the opening of exhibits. Although the number of exhibitors was the smallest in recent memory, there were several interesting products on display. (Source: www.aes.org)

Photo 2: Jan Didden, editor and publisher of *Linear Audio* magazine, and Bob Cordell, proprietor of Cordell Audio, offer a rich assortment of books, including Jan's first reprint of classic audio papers, "Baxendall and Self on Audio Power," and Bob's "Designing Audio Power Amplifiers."

DECCA TREE REVISITED

Classical record collectors have long admired the excellent sound on the stereo recordings made by The Decca Record Company, Ltd. (known in the United States as London Records until the late 1990s). When Decca began making stereophonic recordings in the mid-1950s, they used a microphone system developed in house, known as the "Decca Tree." The Decca Tree consisted of a pair of omnidirectional microphones spaced 6' apart for left and right pickup, and a center

Photo 3: Telefunken Elektroakustic's ELA M260 Tri-Mono recording system, recreated the classic Decca Tree introduced for stereophonic orchestra recording in the mid-1950s. In the lower left, the same microphones are shown on a small bracket suitable for coincident stereo recording or a near-coincident ORFT configuration.

microphone placed 3' in front of the spaced pair. (The original microphones were Neumann M 50.) The arrangement produced a very strong and stable stereo image with a wide and deep soundstage. The Decca Tree served as the main stereo pickup for much of Decca's classical output during the LP era, including such landmark recordings as Wagner's "Der Ring des Nibelungen" with the Vienna Philharmonic Orchestra conducted by George Solti, recorded in Vienna's Sofiensaal, and numerous recordings of "L'Orchestre de la Suisse Romande," conducted by Ernest Ansermet, recorded in Geneva's Victoria Hall. Decca always used multi-miking on their classical recordings, but spot miking was always applied judiciously so as to not distort the natural stereo perspective captured by the three-microphone tree.

Until recently, recording engineers, who wanted to imitate the Decca Tree technique, were forced to "roll their own" since the necessary hardware was not available from any single manufacturer, until now. Telefunken Elektroakustic (www.telefunken-elektroakustik.com), a company specializing in re-creating vintage vacuum tube microphones, including such classics as the Neumann U47 and AKG C12, was exhibiting their ELA M 260 Tri-Mono Recording Set (see **Photo 3**). The ELA M 260 is Telefunken's recreation of the Decca Tree, and their system includes three matched ELA M 260 small-diaphragm vacuum tube condenser microphones, a custom M 963 Tri-Mono three-channel power supply, and a special stand bracket to hold the three microphones in the tree configuration. Each ELA M 260 microphone is supplied with three capsules: omnidirectional, cardioid, and hyper-cardioid, for maximum flexibility. Telefunken also includes three pair of adaptor spheres, both 30 and 50 mm, designed to work with the omnidirectional capsules to emulate the spherical-omni characteristics of the M 50 microphones originally used in the Decca Tree.

HIGH-RESOLUTION DIGITAL

Bricasti Design (www.bricasti.com) demonstrated their M1 digital-to-analog converter, a true dual-mono design with two completely isolated channels, each powered by its own linear power supply (see **Photo 4**). (An internal photo can be seen on their website). At the heart of each channel is an Analog Devices AD1955, a stereo D/A converter chip used in a mono configuration for improved linearity. The AD1955 is Analog Device's flagship converter, featuring Sigma-Delta architecture, an 8× oversampling digital filter, and support for all standard sampling rates and word lengths up to 192 kHz/24 bit. For backwards compatibility, the AD1955 supports the Red Book 50 μs/15 μs de-emphasis standard.

CHIPS FROM TEXAS INSTRUMENTS

Texas Instruments was featuring several new high-performance audio chips at their AES booth, including the OPA16*xx* series of high-performance audio ICs, which includes both bipolar and FET-input devices. The OPA1641 single, along with the 1642 dual and 1644 quad versions, are FET-input op-amps with a slew rate of 20 V/μs, a gain-bandwidth product of 11 MHz, very low input bias current of 15 pA, and ultra-low THD+N of 0.00005%. They also feature much lower noise than most FET-input op-amps: 5 nV/√Hz at 1 kHz, though Texas Instruments's "ultra-low noise" description is a bit exaggerated for this number. For designers in need of true ultra-low noise performance, Texas Instruments offers the OPA1611 single and OPA1612 dual bi-polar input devices, which boast a noise spec of 1.1 nV/√Hz at 1 kHz, THD+N of 0.000015%, a slew rate of 27 V/μs and a gain-bandwidth product of 40 MHz. Input bias current of 60 nA is significantly higher than that of the 1641-series devices, which is expected with bi-polar input op-amps. The OPA61*xx* series of op-amps are available only in surface-mount packages, but the adapters made by Brown Dog Electronics (www.cimarrontechnology.com) should make them attractive to DIY audiophiles, including modifiers of equipment with DIP devices.

Texas Instruments was also promoting their PCM5100 and PCM5102 stereo D/A converter chips (see **Figure 1**). These chips are manufactured in a small 20-pin TSSOP, and are designed to operate on 3.3-V supply rails. Both are 32-bit chips operating at sampling rates of up to 384 kHz, with standard left-justified I2S digital audio interfacing, an internal PLL, and unbalanced 2.1 VRMS audio line drivers. The signal-to-noise ratio is 106 dB for the PCM5100 and 112 dB for the PCM1502.

PAPERS, BRIEFS, & PRESENTATIONS

The decline in exhibits has had no effect on the rest of the convention program. The AES offered a very full slate of papers, workshops and other presentations, with several events usually taking place simultaneously. Since it was impossible for one reporter to cover them all, I'll highlight a few that I found especially interesting.

AES has introduced a new category of presentations called "Engineering Briefs," two of which were especially interesting. "Consumer Attitudes Toward Digital Audio Quality" was given by Ainslie Harris, a doctoral candidate pursuing a DBA degree at the Aberdeen Business School, Robert Gordon University in Aberdeen, Scotland. Harris noted that most of the literature covering digital audio quality has focused on technical aspects—bit rate, sampling frequency, methods of compression, and so forth. Little has been written on the consumer attitudes that cause them to choose various digital audio formats. The choices they make often have little to do with their ability to perceive audi-

Photo 4: Bricasti Design's President Brian Zolner explains the performance features of their M1 Dual Mono Digital-to-Analog Converter, shown on the right. According to Bricasti, extremely low levels of jitter and the complete absence of sample rate converters and digital volume controls contribute to the unit's excellent sonics. The design of the finished product was finalized based on extensive listening with a wide variety of music.

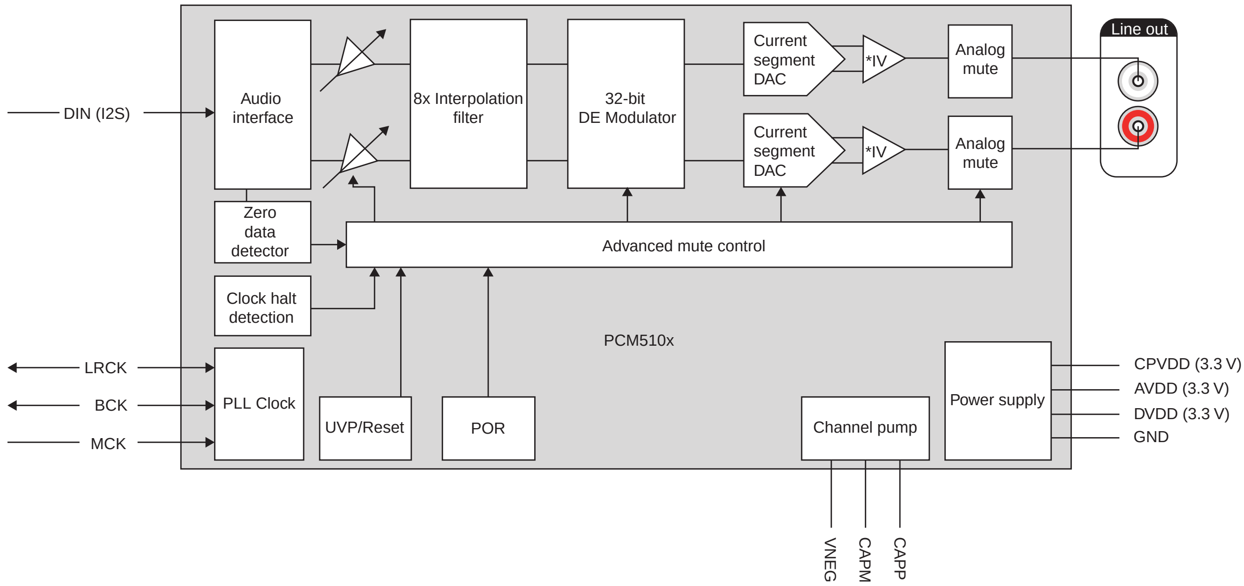


Figure 1: Texas Instruments's PCM510x family of stereo D/A converter chips are manufactured in a small 20-pin TSSOP and are designed to operate on 3.3-V supply rails. Both 32-bit chips operate at sampling rates of up to 384 kHz. (Source: Texas Instruments)

ble differences between various available formats. The research she undertook while preparing her previous AES paper—"Do Young People Actually Care About the Quality of Their MP3s?" (130th AES Convention, London, England, May 13-16, 2011)—led her to group consumers of digital audio into three categories. The first category consists of high school students, aged 15 to 17, with little technical knowledge about digital audio, who are happy as long as the sound is free of clicking noises or fuzzy artifacts. Their only requirement is that their download should play from beginning to end without obvious glitches. These "consumers" generally do not pay for their music. Second, there are college and university students, aged 19 to 24, who are technically more savvy, can perceive more subtle audible differences and, therefore, tend to prefer higher quality audio files. For reasons mostly practical, they tend not to download the highest quality files available. This group tends to divide their music consumption between legitimate purchases and illegal downloads. In the third category are people with full-time employment, aged 25 and older, who have the highest ability to perceive subtle audible differences, have the highest level of technical understanding, and prefer higher quality audio formats. Because of their economic situation, they are able to buy higher quality formats, and pay for their music more often than younger consumers. Harris's research has shown that even when consumers can hear differences between various digital audio formats, they often fail to choose the highest quality formats for purely practical reasons, even when they can afford to do so. Citing research by M. Mulligan, she notes that CDs are the format of choice for the third group, while the second prefer downloads, and the first prefer streamed music.

For Harris, the broad research questions do not include whether or not listeners can hear differences between various digital formats and coding schemes. The purpose of this research is to determine the other reasons that affect their choices including, in particular, why they often choose lesser quality formats even though they can hear those differences. Once recording companies understand why consumers are using and buying certain formats, they will be able to allocate their resources more efficiently. A detailed methodology for this research has not yet been determined, but Harris envisions a qualitative first stage with interviews and focus groups in order to gain a broader contextual understanding of the research questions and consumer behavior. The second stage will be quantitative, and probably conducted online in order to broaden the available sample. Harris originally did not envision listening tests as a part of the methodology, since this is a study on attitudes and not listening ability. But, she has more recently come to believe that listening tests may be valuable in order to determine if the downloading preferences of experienced listeners differ from those of "average" listeners. (She realizes the importance of determining what constitutes an "average" listener.) Harris's research probes a most interesting and worthwhile topic, one of great concern to those of us who lament the current lack of interest in high-resolution digital audio.

John "Beetle" Bailey, proprietor of The Drive Shed, a recording studio based in Toronto, gave a most interesting e-Brief titled "Playback Disappointment in Linear PCM Recording Systems." Bailey noted that after spending many hours doing a mix, when he finally saves his mix as a WAV file on his workstation, he feels a sense of disappointment when that file is played back. The recording seems lacking in depth, reverb tails are truncated, transients are dulled, and the mix has an overall grainy sound. Bailey's paper described the differences he was able to verify by conducting direct-stream digital (DSD) null tests, in order to prove that the sound had somehow changed when the file was saved.

Bailey identified three areas where degradation can be heard in playback: for every individual track on the multi-track recording; when printing the final mix (i.e., saving it as a WAV file); and in the mastering process. To conduct the null tests, Bailey borrowed a Korg MR2000-S DSD digital recorder (www.korg.com/mr2000S) and an Antelope Trinity Clock (www.antelopeaudio.com). The test involves taking the analog outputs from his workstation's main D/A converters and feeding that output to the DSD recorder. At the same time, a PCM recording of the mix is being recorded internally at

44.1 kHz/24 bit. In order to perform a null test, the DSD recorder and workstation must be tied to the same 44.1-kHz clock. (44.1 kHz was selected for the test even though his normal workflow is at 88.2 or 96 kHz; he says the degradation is even worse at the higher sampling rates.) In playback, Bailey says that the DSD recording sounds are essentially identical to what he hears in the live mix, and while recording the mix. But, the 44.1 kHz/24-bit mix, stored as a WAV file, exhibits decreasing depth, less detail in the reverb tails, and grainy high frequencies when played back. He notes that these are classic symptoms of time domain problems.

The null test involves recording three passes of a mix and comparing them by inverting the polarity on one. The first pass is the live mix, the second is the playback of the saved WAV file, and the third is another playback of the same WAV file. The null tests, performed on a Pyramix workstation in the Digital eXtended Definition (DXD) mode at 352.8 kHz/32-bit floating gave very interesting results. Passes 2 and 3 were identical—the null left virtually nothing except the residual noise floor of the two files. But, Passes 1 and 2 showed music-related differences between the live mix and the WAV file playback. (The "live" mix is simply the playback of the multi-track original, and can be called up at any time.) Bailey then conducted a test to see if an external digital recorder could faithfully capture and play back the mix, so he set the Korg recorder to capture the mix using its 96 kHz/24-bit mode. Toggling between the workstation D/A output and the Korg output, he found that they were very close. But, playing back the stored WAV file from the Korg recorder proved just as disappointing as the playback of the stored file on his workstation. Bailey's conclusion is that current methods for capturing a live PCM digital stream are flawed. He notes that, from a perceptual standpoint, the DSD capture of a live steam followed by conversion to the original PCM sampling rate retains much of what direct capture of the linear PCM data stream lacks. He concludes that current Linear PCM technology is capable of storing the audio in a satisfactory manner. It's the capture method that needs improvement. To avoid this problem, Bailey now captures his mixes in DSD at 5.6 MHz, twice the normal DSD sampling frequency. If the mastering engineer wants linear PCM files, he converts the DSD to 96 kHz/24-bit PCM. The resulting WAV files are much closer to the live mix than a direct PCM capture.

DISC PLAYBACK

The playback of disc records may not generate as much interest at AES as is does in audiophile circles, but two poster sessions on that subject caught my attention. Baozhong Tian (see **Photo 5**) presented "Practical Digital Playback of Gramophone Records using Flat-Bed Scanner Images." Tian is a faculty member in the Department of Computer Science at the West Virginia University Institute of Technology. He co-authored the presentation with Samuel Sambasivam and John L. Barron, from the Computer Science departments at Azusa Pacific University and The University of Western Ontario, respectively. The authors note that there have been a number of previous efforts to digitally play disc records from scanned images. Optical Audio Reconstruction offers great potential for restoring fragile or broken records without any physical contact. In their survey of previously-used methods, they note that various

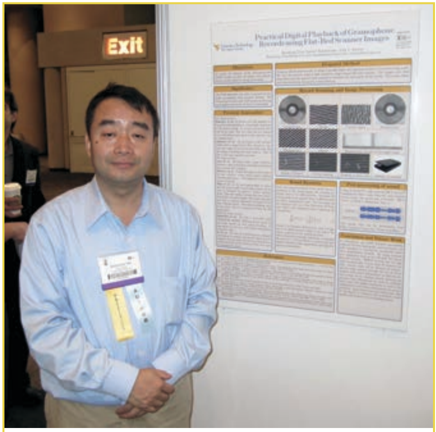


Photo 5: Baozhong Tian gave a poster session titled "Practical Digital Playback of Gramophone Records using Flat-Bed Scanner Images." Dr. Tian's paper described an affordable method for scanning and digitizing 78-rpm records.

two- and three-dimensional, and hybrid approaches, have been both expensive and time intensive. This paper describes an affordable method using a normal flat-bed scanner costing about $230, with a resolution of 2400 dpi, A3 format (11.69 × 16.54"—sufficient width for the actual playing surface of a 12" disc record). Since the stylus follows the groove in a spiral, it may seem intuitive to track the scanned groove using a polar coordinate method. But, it would be very time consuming to convert the image space—which is in Cartesian coordinates—into polar coordinates every time a location on the scan is accessed. So, the scanned record is transformed from its original circular shape into a rectangular shape. The flat-disc record has constant angular velocity, so the authors have factored the changes in linear velocity from the outside to the inside of the disc when digitally tracking the record groove.

The authors used a 78-rpm disc record to develop and demonstrate the method. The record was a common 12" Victor Red Seal disc, featuring Arthur Fiedler and the Boston Pops Orchestra performing highlights from Rodgers and Hammerstein's "Oklahoma!" (catalog number 11-8742A, recorded in 1944). Once the scanned image has been converted into a digital sound file, digital click removal, plus broadband noise reduction, is performed on the file. Then, the appropriate disc playback equalization—the complement of the recording characteristic—must be applied. The authors used the free digital audio editor, Audacity, for both (www.audacity.sourceforge.net). Audacity has an equalization module that applies an RIAA playback curve. (Some people, who transfer disc records to digital format, prefer to play a record "flat" and apply the playback equalization in the digital domain.)

I raised two questions with Dr. Tian regarding their method for processing the audio once the digital scan has been converted into analog. First, the RIAA curve provided by Audicity does not provide the correct playback equalization for 78-rpm records. For a mid-1940s Victor 78, a bass turnover of 500 Hz and a treble roll-off of –8 or –10 dB at 10 kHz should be reasonable. While the RIAA bass turnover of 500 Hz is fine, the treble roll-off of –13.75 dB at 10 kHz is excessive and will make the record sound dull. Second, the RIAA equalizer in Audicity was designed using finite impulse response (FIR) filters, which do not mimic the phase response of analog filters and, therefore, do not provide a phase response that is the inverse of the recording characteristic. Audacity verified this to me when I was preparing a conference paper and article on this subject. I suggested an affordable program that provides correct phase equalization—the Virtual Phono Pre-amp equalization module supplied with the DC 8 audio restoration software (www.tracertek.com). The filters in the Virtual Phono Pre-amp were designed using infinite impulse response (IIR) filters, which mimic the phase characteristics of analog filters. This module is supplied with over 50 predefined playback curves for a variety of records, including RIAA, early LPs and 78-rpm discs. DC 8 also has click-and-crackle removal and broadband noise reduction modules that are more flexible and effective than those supplies with Audicity. I also recommended Sound Forge Pro 10, an outstanding digital editor, which has excellent noise reduction modules for click-and-crackle removal and broadband noise reduction. These are actually easier to use than those supplied with DC 8 and give excellent results.

Photo 6: Jason Hockman's poster session, "Discrimination between Phonograph Playback Systems," described listening tests designed to assist institutions and sound archives in the selection of equipment for digitizing disc records.

A poster session titled "Discrimination between Phonograph Playback Systems" was presented by Jason A. Hockman (see **Photo 6**) and co-authored by David M. Weigl, Catherine Guastavino and Ichiro Fujinaga (all from the Center for Interdisciplinary Research in Music Media and Technology at McGill University). This paper is also concerned with the digitization of phonograph records, though using conventional playback equipment. The study was to determine the extent to which expert listeners can hear differences between phono playback systems in different price ranges. There are currently no standards for librarians and archivists to use when selecting a phono cartridge, turntable, phono pre-amplifier, analog-to-digital converter and interconnect cables. The results are intended to serve as the basis for guidelines for purchasing phono playback systems for the digitization of record collections. The double-blind A-B tests conducted by the authors compared two phono playback systems—a high-end system costing approximately $31,250 and a mid-range system priced at $8,066. For this paper, results are presented for two listening conditions. The first compared excerpts from phonograph records played on the two systems. The second compared recordings on the expensive system, substituting only the phono pre-amplifier for the A-B comparisons.

The listening panel for Condition 1 consisted of 11 "expert" listeners (nine males and two females). And 14 participated in Condition 2 (10 males and four females). Eight of the same people were able to participate in both tests. Five recordings representing various types of music were tested: classical, jazz, rock, Latin/fusion, and rock/fusion. Records were cleaned on a VPI HW-17 machine and digitized at 96 kHz/24 bit using a PrismSound ADA-8XR A/D converter.

The subject of the audibility of phono playback systems has aroused considerable controversy since the 1970s. Longtime *Audio Amateur* readers will recall the ongoing debates in the letters section between Stanley Lipshitz and Walt Jung in the late 1970s. The former insisted that audible difference between phono pre-amplifiers could be accounted for only by frequency response differences caused by errors in RIAA equalization, and the latter believed that a variety of other pre-amp design factors influence perceived sound quality. Although that debate remains unresolved, by the early 1980s both the "scientific" and the "golden ear" camps had generally agreed that if RIAA playback equalization was accurate from 20 Hz to 20 kHz, ±0.1-dB, frequency response could probably be ruled out as a cause of audible differences between phono pre-amplifiers.

Listening comparisons was the subject of another AES paper—"Distortions in Audio Op-Amps and Their Effect on Listener Perception of Character and Quality"—presented by Robert-Eric Gaskell and co-authored by Peter E. Gaskell and George Massenburg. All three authors are on the faculty at McGill University. The authors note that the excellent specifications for today's audio op-amps—low distortion and noise, high slew rate and extremely high common-mode rejection—would seem to make audible differences between them unlikely. Nonetheless, there is ongoing discussion between design engineers and audio engineers regarding the sonic characteristics various op-amps. The authors set out to quantify distortion behavior of op-amps, and determine if the distortion characteristics impact sonic perception. Four distortion mechanisms were identified for voltage-feedback op-amps: slew rate, transfer nonlinearity, input nonlinearity, and output nonlinearity. The authors designed a test circuit which was used for both the measurements and the listening tests. The test circuit uses a distortion-selective amplifier configuration in order to maximize the listener's ability to hear a change in sound quality for the various op-amps. With this test configuration, the op-amps' inherent distortion is amplified by 40 dB. The frequency response of the various devices under test was kept to within 0.6 dB at 20 kHz and 2 dB at 48 kHz using compensation capacitors in the feedback networks of the test circuit.

Twenty op-amps from seven different manufacturers were tested—seven discrete designs and 13 integrated-circuit types. THD measurements were made using an Audio Precision System Two. The source material for the listening test all came from 96 kHz, un-mastered mixes of music from different genres—two jazz, two folk-rock, two rock, and two classical—all 10 to 20 seconds long. The authors note that "only two of the detected comparisons produced statistically-significant preference data…As a result, very little can be said with confidence about listener preference for the subtle differences in distortion characteristics. Surprisingly, higher levels of distortion did not generally reduce preference for the amplifiers. In the two cases that showed significance, the op-amp that produced higher levels of distortion was preferred." The authors also note that even in the results that were not statistically significant, "there appears to be a trend toward preference of the higher-distortion op-amps." Another outcome was the inability of listeners to distinguish between op-amps with low and high levels of high-order THD. Two op-amps—whose spectral distortion characteristics were nearly identical through the 5th harmonic, but were very different (one low, one high) above this point—were not discriminated by listeners. The preference for higher levels of distortion and the inability to distinguish between an op-amp that produces the type of distortion thought to be audibly nasty (high order) and one that does not, is indeed puzzling, and makes the test results inconclusive, in my view.

HISTORICAL PRESENTATIONS

Two particularly noteworthy presentations were featured on the AES Historical Program this year. Robert Auld, a well-known audio engineer in the New York City area, presented "Leopold Stokowski and the History of Analog Recording." Stokowski, one of the most important 20th-century conductors, made his first recordings in 1917 and remained a prolific recording artist until the year before his death in 1977. As such, his recording career spanned nearly every analog recording technology, including acoustically-recorded 78-rpm discs, electrically recorded 78s, long-playing records, and magnetic tape, the latter including single-track mono, two-track stereo and, finally, multi-track. The only consumer format he missed was the cylinder. Its popularity was fading rapidly by the time he made his first records. Auld (see **Photo 7**) began his presentation with Stokowski being interviewed by Glenn Gould, in which Stokowski recalled his rejection of the first offer he received to make records with the Philadelphia Orchestra, from The Victor Talking Machine Company. After thinking it over, he realized how foolish he had been. He contacted Victor, reversing his previous decision, deciding to work with them to try to improve the technology. From that point, Stokowski was at the forefront of virtually every important development in the evolution of sound recording.

Auld has a special interest in playback of historical recordings, and his own transfers and restorations of disc sources—both 78 rpm and LP—were very well done. His website has some interesting and useful information on this subject and related audio issues,

Photo 7: Robert Auld makes a point during the question-and-answer portion of his excellent historical presentation, "Leopold Stokowski and the History of Analog Recording."

Photo 8: Tom Fine's historical presentation, "American Classical Recording from One Microphone to Many Tracks," surveyed the work of the great American recording engineers of the 1950s and 1960s, including C. Robert Fine, Lewis Layton, and Carson Taylor.

plus a number of other interesting articles on various aspects of sound recording (www.auldworks.com).

Tom Fine presented "American Classical Recording from One Microphone to Many Tracks." Tom (see **Photo 8**) discussed classical recording aesthetics, and the concept that the sound of a recording is often intentionally different from that of a live event. Tom played many recorded examples to illustrate the differences between the various "purist" approaches to classical recording (using as few microphones as possible, often one or two), and contrasted those with the multi-microphone approaches that became more and more popular for classical recording as the stereo era progressed. Not all purist recordings sound the same, by any means. Some engineers, like Robert Fine, preferred a fairly close perspective on the orchestra whereas his counterpart at RCA Victor, Lewis Layton, tended to move a bit further back. Tom began by comparing two monaural recordings of Richard Strauss's "Ein Heldenleben," the first a 1952 Mercury with Antal Dorati and the Minneapolis Symphony, and the second being the legendary 1954 Fritz Reiner/Chicago Symphony recording for RCA Victor. Both examples offered remarkably natural sound and illustrated the differences between the conductor's perspective on the Mercury and a "down front" audience perspective on the RCA Victor. He then did a mono/stereo comparison of the two versions of the RCA recording that were made at those sessions. The experimental stereo version, engineered by Leslie Chase, had two spaced microphones and one accent microphone on the percussion. The early RCA Victor stereo tapes hold up amazingly well even today.

To illustrate the increasing number of microphones used in classical recording in the 1960s and 1970s, Tom compared four recordings of the "Interlude and Dance" that begins Part 2 of Ravel's *Daphnis et Chloé*. From the 1950s we heard Antal Dorati's mono Mercury from 1954, made with a single Telefunken U-47 condenser microphone, followed by Charles Munch's classic version with the Boston Symphony Orchestra made in 1955 with just a handful of microphones —two for main stereo pickup and two spot mics. By the time Munch did his remake in 1961, RCA Victor had increased the number of microphones to 11, with three-spaced omnis for main stereo pickup, plus spot mics. By the early 1970s, Columbia, under direction of their main classical producer Andrew Kazdin, and engineer Bud Graham, had moved heavily into multi-microphone recording. Their view was that a classical recording was a production unto itself, having nothing to do with the sound of an orchestra in a concert hall. Their 1974/75 recording of Daphnis with Pierre Boulez and the New York Philharmonic was made with 26 microphones on eight separate tracks. The engineers and producers of most major recording companies were moving toward more complex multi-microphone recording throughout the 1960s, but for most of them a main stereo pickup—whether two or three spaced omnidirectional microphones or a co-incident pair—still served as the focal point for the stereo image and perspective. The spot microphones were strictly to reinforce the main pickup. The Graham/Kazdin approach, by contrast, produced a completely manufactured stereo perspective.

Tom concluded with two examples from the American Capitol/Angel/EMI catalog, both recorded by veteran Capitol engineer Carson Taylor, but with two rather different perspectives on the orchestra. Taylor favored coincident microphones for mono compatibility, using Neumann SM-69 and AKG C-24 stereo microphones. Accent microphones included Neumann KM-86s and U-87s, plus AKG C-414s, all fed to a 1", 3M DynaTrack recorder. Tom explained that Taylor didn't like Dolby-A noise reduction, and instead used 3M's unusual method for minimizing tape hiss, which involved a duplicate recorded track. One track would be recorded with normal NAB equalization. The other had a shelving pre-emphasis beginning at 400 Hz, with a maximum boost of 15 dB between 15 and 20 kHz. A complementary de-emphasis was applied in playback. In playback, the recorder would automatically switch between the two, depending on overall playback level.

Tom's presentation was an extremely interesting survey of the various classical recording philosophies in practice during the first two decades of the high-fidelity age. AES's historical presentations have had a loyal following at the New York conventions and one hopes that they will continue to feature these in the future. The AES deserves kudos for providing a fine playback system for this presentation. The PMC loudspeakers (www.pmc-speakers.com) do a remarkably good job of filling a large space with reasonably refined and accurate sound, which is not an easy task. aX

RESOURCES

G. Galo, "New York AES 2009: The Audio Industry feels the Economic Pinch," *audioXpress*, June 2010 (on-line edition only).

S. Lipshitz, "On RIAA Equalization Networks," *Journal of the Audio Engineering Society*, June 1979.

G. Galo, "Phase Equalization and its Importance in the Reproduction of Disc Records," *ARSC Journal*, Spring 2010.

By Richard Honeycutt (United States)

The Frequency Response of Hollow-State Amplifiers

Varying frequency responses explored

Early electronic devices were intended mainly for speech amplification and reproduction. By the 1930s, however, musical program material gained importance, and an extended frequency response became a commercial necessity. This emphasis grew until, in the 1950s and 1960s, the Harmon Kardon Citation audio amplifier claimed frequency response from 1 to 100,000 Hz flat within a decibel or better. Although today, other performance metrics have surpassed frequency response in advertising emphasis—in part because wide, flat frequency response is now easier to obtain with modern circuitry—frequency response remains a very important parameter.

HUMAN HEARING

Humans with pristine hearing (mainly the very young) are often described as being able to hear sounds having frequencies between 20 and 20,000 Hz. In fact, the limits of hearing are difficult to determine, for two reasons. At the lowest frequencies, the ear is relatively insensitive, and the level of a 20 Hz sound has to be so high that it approaches the threshold of feeling, so that a person can perceive acoustic vibrations all the way down to about 1 Hz. Just where hearing merges into feeling is a matter of some debate. At the upper end of the hearing range, facial cilia are able to sense vibration up to about 100 kHz. Again, whether this constitutes hearing is open to debate.

So the range of frequencies that an audio amplifier should cover is by no means well-defined. However, some speaker manufacturers do make super-tweeters designed to reproduce frequencies above 20,000 Hz. And if the amplifier is to be used for an electric musical instrument, a more restricted range is justified, since the instrument's range may not be greater than about 80 to 5,000 Hz.

Just which factors determine the low- and high-frequency limitations of vacuum tube amplifiers? In order to examine these factors, we need to discuss a bit of electric circuit theory. If a voltage source —AC or DC, it doesn't matter—is connected to a resistance, the resulting current is given by Ohm's Law: I = V/R. If the voltage source is of the AC variety, and the resistor is replaced by a capacitor or inductor, the current is given by: I = V/X where X is the reactance of the capacitor or inductor. Reactance limits current flow by means of temporary energy storage: capacitive reactance X_C does so via the electric field, and inductive reactance X_L stores energy in the magnetic field. **Figure 1** shows the values of reactance provided by a 0.1 µF capacitor and a 254 mH inductor, for a frequency range of 10 to 30,000 Hz. Notice that capacitive reactance decreases with frequency; whereas, inductive reactance increases as frequency increases.

The equations for calculating reactance are:

$$X_C = \frac{1}{2\pi fC}$$

f is frequency in Hz. C is capacitance in farads.

$$X_L = 2\pi fL$$

L is inductance in henries. If a capacitor and a resistor are connected in parallel, and X_C is small compared to the resistance of the resistor, the AC voltage drop across the circuit will be less than if the capacitor were not present. Thus a capacitance appearing between a signal conductor and ground will tend to reduce the signal voltage on that conductor. Thus, the higher the frequency, the lower the signal voltage will be in this arrangement. The opposite is true for an inductor in parallel with a resistor: the lower the frequency is, the lower the signal voltage.

In a series RC circuit, a high capacitive reactance (low frequency) decreases AC signal flow in the resistor. The same is true for a series RL circuit, except that high inductive reactance implies a high frequency.

So how do these facts apply to vacuum tube amplifiers?

LOW-FREQUENCY RESPONSE

Figure 2 shows a triode amplifier stage having three capacitors: one for input coupling (to block DC voltages from the control grid), one for output coupling (to block the DC plate voltage from the following stage), and one to bypass the cathode resistor (to prevent AC voltage from causing negative feedback to the cathode, thereby reducing the gain). All three of these capacitors affect the low frequency response. The coupling capacitors do so by dropping part of the signal voltage at low frequencies so that less of it appears at the control grid of the present stage, or the next stage. The cathode bypass capacitor af-

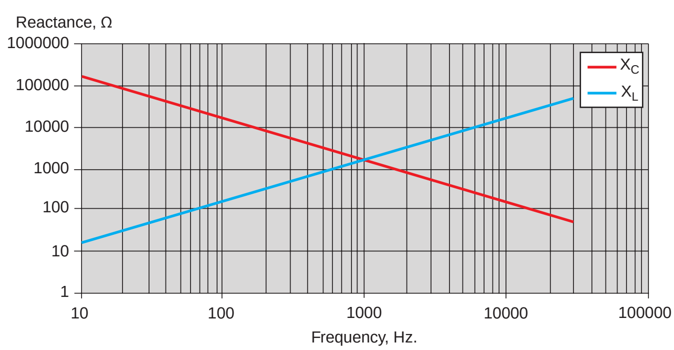


Figure 1: Capacitive and inductive reactance

fects the low-frequency response because at low frequencies it no longer prevents AC voltage from appearing at the cathode, so that negative feedback reduces the low-frequency gain.

The equation for calculating the value of input coupling capacitor to provide a gain that drops by 3 dB at low frequency f_L is:

$$C = \frac{1}{2\pi f\ (R_S + R_G)}$$

R_S is the resistance of the signal source and R_G is the resistance from grid to ground (the 1-M Ω resistor in **Figure 2**.)

If the resistance of the signal source is not known, or is much smaller than R_G, then R_S can be omitted from the equation. The calculated value for the capacitor will then be larger than necessary, giving a slightly lower low cutoff frequency.

The same equation can be used for the output coupling capacitor, except that in this case, R_S is the parallel combination of r_P (the plate resistance of the tube) and R_P, the resistor between the plate and B^+:

$$R_S = \frac{r_P R_P}{r_P + R_P}$$

For the low frequency at which stage gain drops by 2 dB, the capacitor value should be multiplied by 1.316. For 1 dB, multiply by 1.96.

The equation for the cathode bypass capacitor is much more complex:

$$C_K = \sqrt{\frac{0.25\left\{\left[1 + \frac{(\mu + 1)R_k}{R' + r_P}\right]^2 - 1\right\}}{0.75\ (2\,\pi\, fR_k)^2}}$$

where R' is the parallel combination of R_P and R_{G2}, the grid resistor of the following stage (or the load resistance, if there are no more stages in the amplifier):

$$R' = \frac{R_P R_{G2}}{R_P + R_{G2}}$$

The above equation for C_k gives the value of a cathode bypass resistor in farads to achieve a 3-dB drop in stage gain at the frequency f. For 2-dB or 1-dB drops, the correction factors given above can be applied.

For example, a triode having a μ of 20, with a 2700-Ω cathode resistor, a 100-kΩ plate resistor, and the following stage having a 1-MΩ grid resistor, would require a 0.63-μF cathode bypass capacitor for response to be flat within 3 dB down to 100 Hz.

The effects of the various capacitors on low-frequency response combine: If the input coupling capacitor causes a 3-dB drop in response at a certain frequency, and the cathode bypass capacitor causes a 3-dB drop at the same frequency, the amplifier's overall response will be down 6 dB at that frequency. But this combining can easily be taken care of in the design stage. Since the capacitor values are scalable, if you have a multistage amplifier, you just count up the capacitors and multiply the calculated value for each one by the number of capacitors. For example, a three-stage amplifier will probably have one input capacitor, two interstage capacitors, one output capacitor, and three cathode bypass capacitors (i.e., seven capacitors). Thus, you would calculate the necessary capacitor values and multiply each by seven.

Figure 3 shows a triode amplifier with a transformer-coupled output. Although transformer coupling is mainly used in power amplifiers, the same frequency-response issues pertain to small-signal amplifiers. Transformers are used in amplifiers principally to change impedances. If the number of turns in the primary winding is designated as N_P and the number in the secondary as N_S, then the voltage, current, and impedance transformation ratios are given by:

$$\frac{V_S}{V_P} = \frac{N_S}{N_P} = \frac{I_P}{I_S}$$

$$\frac{Z_P}{Z_S} = \left(\frac{N_P}{N_S}\right)^2$$

If the secondary winding is unterminated, the impedance of the primary winding is composed of the winding's resistance and inductive reactance. For good efficiency, the resistance is usually quite low. However, in order for the transformer to function well, the inductive reactance must be quite high. This is true because when

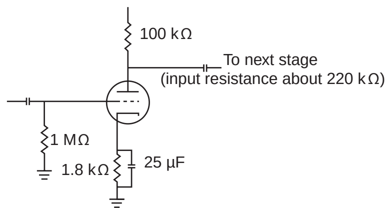


Figure 2: Triode amplifier stage

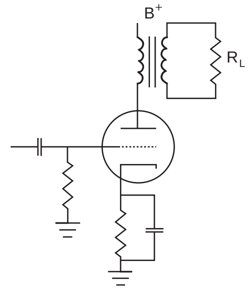


Figure 3: Transformer-coupled triode amplifer

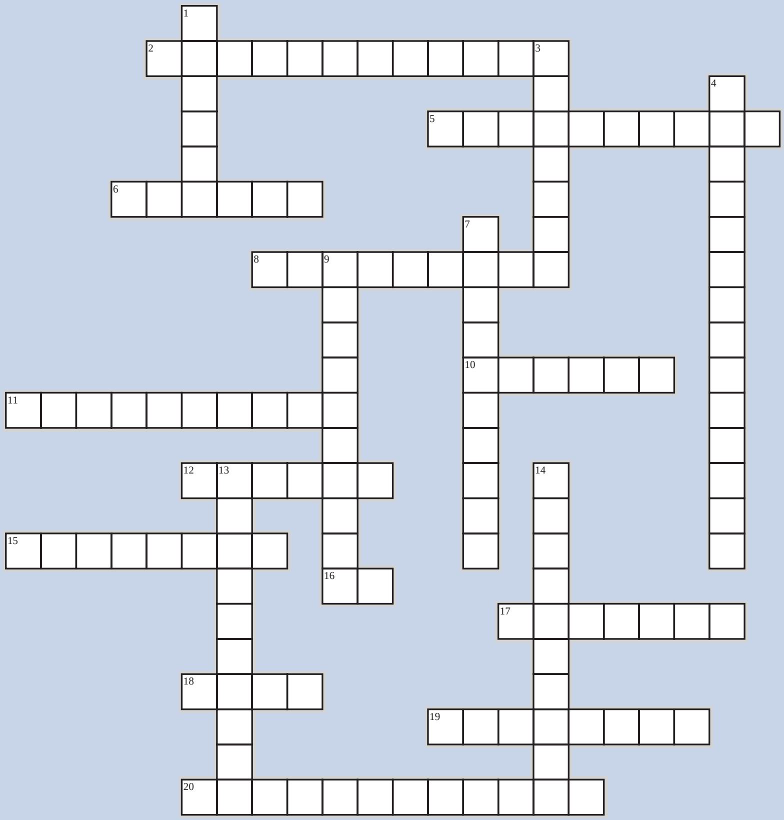
the secondary winding is terminated, the impedance reflected back to the primary circuit appears in parallel with the primary's unterminated impedance. If that unterminated impedance is high (due to a high inductive reactance), the unterminated impedance will have very little effect on the impedance appearing across the primary winding, so that the primary impedance will be as given by the equation above. Otherwise, the impedances and voltages will not be as expected. Since inductive reactance is directly proportional to frequency, the bottom line is that a transformer must have a high inductive reactance in order to provide good low-frequency response. Transformer designers take this into account, along with current-handling capability, in determining wire gauge and number of turns for transformers. Thus, high-power output transformers having good low-frequency response must be large, since they require many turns of fairly large-diameter wire. Input transformers handle much less current, so they can use smaller wire, and are not as large. The reason input, output, and interstage transformers are specified in terms of impedance is that the windings' inductances must be coordinated with the expected termination impedances in order to achieve good frequency response. Thus, a speaker line transformer having a primary:secondary impedance ratio of 49 Ω:8 Ω would not perform well as a line-level output transformer intended to provide a 490-kΩ load to a triode to feed an 80-kΩ device, even though the impedance ratios are the same.

HIGH FREQUENCIES

The high-frequency response limitation of a capacitor-coupled vacuum tube amplifier is mainly caused by the "parasitic" capacitances between the tube's electrodes. Any time there are two conductors separated by an insulator, you have a capacitor. Thus, in a triode, there is a capacitance between grid and cathode, and between grid and plate. The cathode-to-plate capacitance is generally unimportant. The grid-to-plate capacitance is the important one, because it provides a path for AC signals to travel from the plate back to the grid. Since the signal at the plate is out of phase with the signal at the grid, the result is negative feedback, which reduces the gain. And since the reactance of the grid-to-plate capacitance decreases as frequency increases, the gain is decreased more at high frequencies.

The grid-to-plate capacitance of a 12AX7 is about 1.7 pF, providing a reactance of about 4.7 MΩ at 20 kHz. This is a large enough reactance that it does not cause significant gain reduction. In general, audio amplifiers are not very susceptible to high-frequency response issues from interelectrode capacitances, although this can become an issue at radio frequencies.

Transformers are the other source of high-frequency limitation in tube amplifiers. There is a small capacitance between each pair of adjacent turns in the windings of any transformer, and unless a conductive electrostatic shield is used between the primary and secondary windings, there is a capacitance between the primary and secondary windings as well. These capacitances essentially shunt signal current so that it does not flow as intended in the transformer windings, and thus reduce the gain at high frequencies, where the capacitive reactance is low. Thus, a transformer-coupled tube amplifier (or a transformer-coupled solid state amplifier, for that matter) requires a high-quality transformer in order to have good high-frequency response.



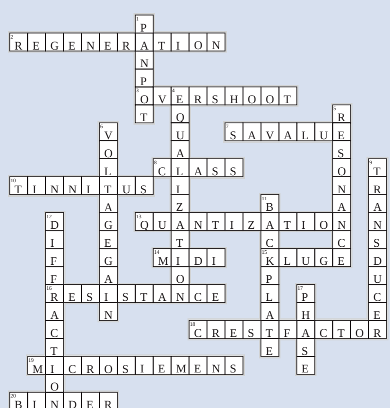
Editor's Note: Answers to this puzzle will appear in the next issue.

ACROSS

2. A measurement of inductance
5. A technique of introducing a small amount of line-frequency signal so as to cancel any hum introduced
6. A type of magnetic tape coating that uses iron oxide
8. A standing wave set up between two parallel room surfaces
10. Any time more than one speaker device covers the same part of the frequency range there will be some unevenness in the output.
11. A term used to describe the serial electrical connection between devices or modules (two words)
12. The process carried out by an A/D converter where the instantaneous amplitude of a signal is measured many times per second
15. The difference between positive and negative
16. A driver's rated RMS power-handling capability
17. A four-electrode tube (valve) based on the triode
18. 1000 ohms
19. A description of the synchronization systems available to MIDI users—MIDI Clock and MIDI Time Code (two words)
20. A 48-V DC supply for capacitor microphones that is transmitted along the signal core of a balanced mic cable (two words)

DOWN

1. A low-level random noise added to a digital signal to mask highly audible forms of digital distortion
3. Hidden data within the CD and DAT format that includes such information as the absolute time location, number of tracks, total running time, etc.
4. A baffle that completely isolates the back wave of a driver from the front without a standard enclosure (two words)
7. A measure of the springiness in a component
9. A wiring problem where multiple ground connections are causing audible mains hum to be picked up (two words)
13. A means of generating a control signal based on how much pressure is applied to the keys of a MIDI keyboard
14. A random signal with an energy distribution that produces the same amount of noise power per hertz (two words)



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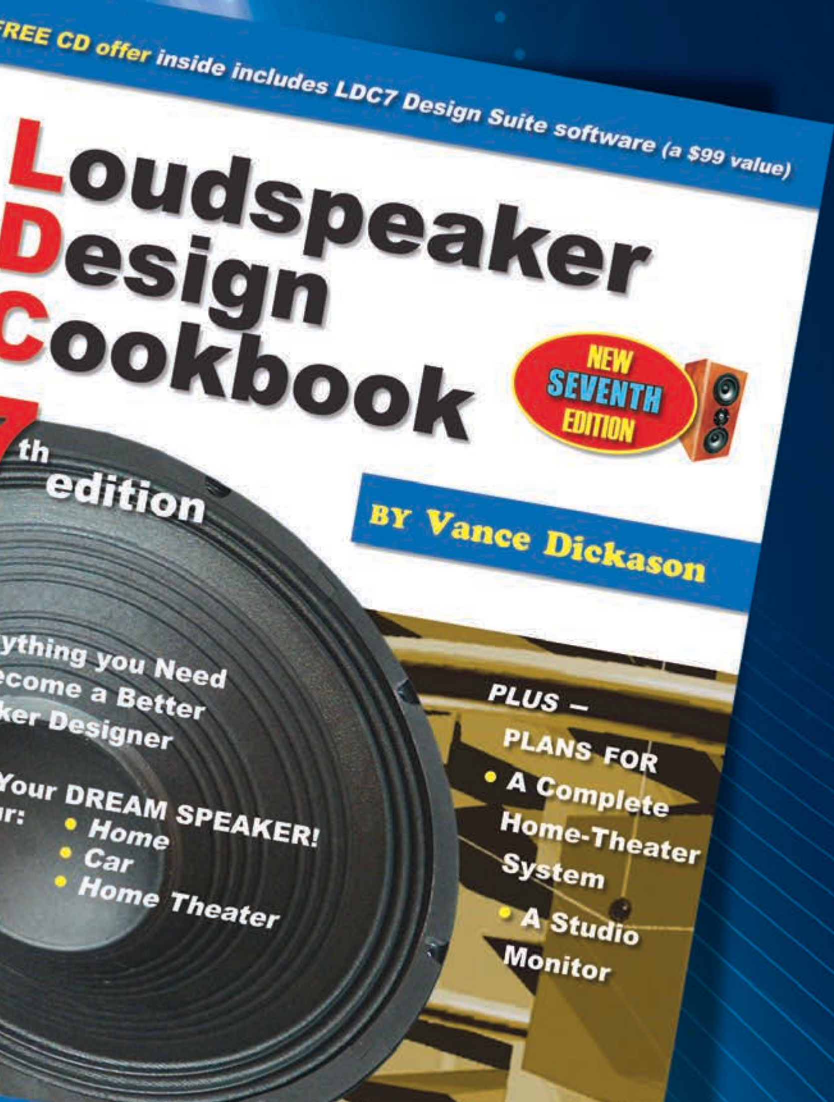

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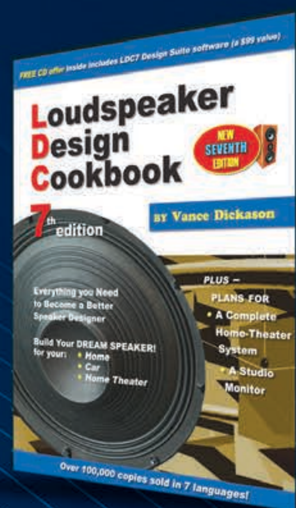

AudioControl's new concert AVR-4 HD home theater

AUDIOCONTROL LAUNCHES THE NEW CONCERT AVR-4 HD HOME THEATER RECEIVER

AudioControl—a United States-based designer and manufacturer of award-winning products that transform the audio and video performance of home theater and media systems—recently launched the Concert AVR-4 HD Home Theater Receiver. The Concert AVR-4 HD fills a gap in the market for an audiophile-quality receiver, at a great price, that does not compromise on the quality of its components, and delivers a stunning movie experience rich in top-tier audiophile sound quality and video performance.

The exclusive, up-market design of the Concert AVR-4 HD hides behind its subtle styling—a huge seven-channel, 120-W-per-channel amplifier with the advanced audio and video processing for which AudioControl is renowned. The Concert AVR-4 HD has been designed to deliver a stunning advance in AV quality without the gimmicks that compromise the core quality of the audio and video platform.

The Concert AVR-4 HD's features include:

- 7.1 Home Theater HD advanced surround sound receiver
- Upgraded 1080p HDMI switching up to five sources
- HDMI1.4, 3-D video pass through
- Powerful, dynamic 120-W-per-channel amplification into all channels driven
- Seven channels of "Cool Running" Class H amplification, reducing the need for rack fans
- AudioControl's discrete input AV processing
- Dolby volume, Dolby true HD and DTS-HD master audio lossless HD decoding formats
- Compatibility and integration with leading control systems, including RS-232 ports
- Sirius Satellite performance functionality with optional tuner module
- A versatile array of inputs and outputs to meet the needs of the connected home, including dual assignable HDMI outputs with priority switching
- AudioControl's unique and discrete styling for both rack and non-rack mounted needs
- AudioControl's 30-year focus on audiophile quality, made in the United States, and accompanied by a five-year warranty program

For more information, visit www.AudioControl.com.

EMINENCE ADDS THREE ADVANCED HIGH-POWER LOUDSPEAKERS

Eminence Speaker LLC has introduced three advanced high-power additions to the Professional Series: the Impero 12A, 15A, and 18A. Italian for "empire," Impero sets the tone for this elite offering of high quality, hand-built loudspeakers. Using high motor strength with balanced Vas and Mms, the Impero series plays extremely loud and low in compact vented designs. Advancements include a 0.5" × 7.5" machined top plate, one piece machined T-yoke, and a 4" deep wound fiberglass voice coil. These features combined with an ultra-linear long-excursion suspension and bumped motor assembly allow the Impero series to move serious air, generate serious SPL, while handling tremendous amounts of power.

At 1,100 W continuous/2,200-W program power, the 12" Impero 12A has a usable frequency range of 56 Hz to 3 kHz, making it perfect for two-way top boxes, full-range two-way boxes, bass guitar boxes, and small subwoofers.

The 15" Impero 15A is suited for two-way top boxes, full-range two-way and three-way boxes, bass guitar boxes, and small subwoofers, and is rated at 1,200 W continuous/2,400-W program.

With a frequency range of 46 Hz to 2 kHz, this 15" has the extension at both ends to go low and high, perfect for small vented boxes with conventional sized horns. This model is also available in a 4-Ω version, the Impero 15C. The 18" Impero 18A is also rated at 1,200 W continuous/2,400-W program power, has a frequency range of 39 Hz to 820 Hz, and is suited for full-range three-way boxes, bass guitar boxes, and small subwoofers. This model is also available in a 4-Ω version, the Impero 18C.

Like all Genuine Eminence-branded professional audio and musical instrument loudspeakers, the Impero 12A, 15A, and 18A are built by hand in the United States and are backed by seven-year warranties. The company has manufacturing operations in Eminence, KY, and Dongguan, China.

Most recently, Eminence added an enclosed manufacturing operation at its Dongguan location to provide its OEM customers with turnkey professional audio systems, musical instrument amplifiers, and related products. More information about the company is available at www.eminence.com. aX

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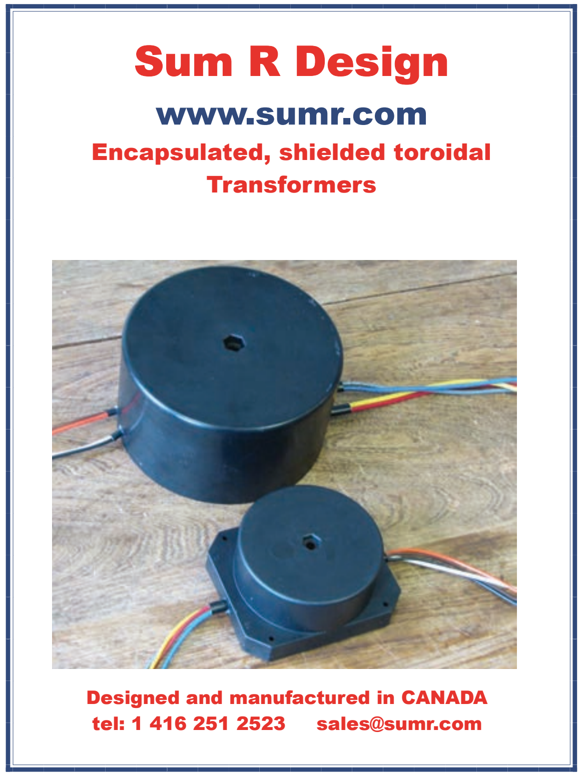

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MEMBER PROFILE

Member Name: George Danavaras

Location: Agia Paraskevi, Athens, Greece

Education: A diploma in Electrical Engineering from the National Technical University of Athens, 1986

Occupation: Product development manager working for a defense telecommunications electronics company

Member Status: George subscribes to *audioXpress*. He subscribed to *Audio Amateur* and *Speaker Builder* in the past.

Affiliations: George is a member of the Technical Chamber of Greece (since 1986), a member of the Federation of Telecommunications Engineers of the European Community (FITCE), and a member of the Audio Engineering Society

Audio Interests: George is interested in the design of pre-amplifiers, active crossovers, power amplifiers, and loudspeakers. He said other projects related to audio (like DIY measurement microphones and accelerometer amplifiers) are always on his agenda. He also enjoy listening to classical music.

Most Recent Purchase: Last year, he purchased a Rotel RSP-1570 pre-amplifier. He is using this pre-amplifier in a multi-channel system.

Current Audio Projects: "I am working on the design of a high-quality headphone amplifier and the design of an on-wall loudspeaker to replace my current surround loudspeakers," George said.

Dream System: "Most of my system comes from DIY projects so it is constantly changing, especially the loudspeakers," George said. His current audio and video setup comprises the following:

- Philips BDP 7500 MK II 3-D Blue Ray player
- Walker CJ-58 turntable with a NAD PP2 phono pre-amplifier
- Rotel RSP-1570 pre-amplifier
- Behringer DCX-2496 used as a crossover for a DIY Horizon loudspeaker
- The Horizon loudspeaker is a three-way open baffle design. Six power amplifiers (each one based on four LM3886 Chips) drive the two Horizons.
- Two subwoofers based on 12" Peerless 830500 woofers. Another Behringer DCX-2496 and a power amplifier are used to drive them.
- Two surround loudspeakers driven by another two power amplifiers.
- LG 48" LCD television for viewing movies and concerts.

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CONTRIBUTORS

Vance Dickason ("Fountek's 3" Full Range," p. 19) has been working as a professional in the loudspeaker industry since 1974. He is the author of the *Loudspeaker Design Cookbook*—which is now in its 7th edition and published in English, French, German, Dutch, Italian, Spanish, and Portuguese—and *The Loudspeaker Recipes*. Vance is currently the editor of *Voice Coil: The Periodical for the Loudspeaker Industry*, which is published monthly.

Gary Galo ("AES 2011: Details of a Full Technical Program," p. 30) recently retired after 34 years as an audio engineer at The Crane School of Music, SUNY at Potsdam, NY. He now works half-time in that capacity. Gary is the author of over 260 articles and reviews on both technical and musical subjects. He is a regular contributor to *audioXpress* magazine, and has written for the various magazines of Audio Amateur Publications since 1982. An active member of the Association for Recorded Sound Collections (ARSC), Gary has been a reviewer for the *ARSC Journal* since 1989, Sound Recording Review editor of the *ARSC Journal* since 1995, was co-chair of the ARSC Technical Committee from 1996-2004, and is a frequent presenter at ARSC conferences. He periodically reviews books for *Notes: Quarterly Journal of the Music Library Association*, has written for the *Newsletter of the Wilhelm Furtwängler Society of America*, *Toccata: Journal of the Leopold Stokowski Society*, and is the author of the "Loudspeaker" entry in *The Encyclopedia of Recorded Sound in the United States*, 1st Edition.

Richard Honeycutt ("Fundamental Amplifier Techniques with Electron Tubes," p. 26 and "The Frequency Response of Hollow-State Amplifiers," p. 37) has been an audio enthusiast, musician, recording engineer, broadcast radio engineer (FCC First Class Commercial licensee since 1969), radio announcer, electronics repairman, tube guitar amplifier designer, speaker system designer, pro-sound/video designer, and college electronics teacher. He is now a consultant in acoustics and electroacoustics. He possesses a PhD in Electroacoustics from the Union Institute, and lives in Lexington, NC, with his wife Betty Jane, near his daughter Alyson, her husband, and two of Richard's three grandchildren.

Mike Klasco ("The Voice Coil (Part 2): Materials and Quality," p. 8) is the president of Menlo Scientific Ltd., a consulting firm for the loudspeaker industry, located in Richmond, CA. He is the organizer of the Loudspeaker University seminars for speaker engineers. Mike contributes monthly to *audioXpress*. He specializes in materials and fabrication techniques to enhance speaker performance.

Charlie Mimbs ("Hybrid Electrostatic Loudspeakers with a Twist," p. 14) is a 58-year-old Georgia native and father of two currently living in Savannah. He works as a manufacturing engineer with Gulfstream Aerospace Corp., specializing in advanced composite materials. His interests include jazz music, tai chi, cycling, woodworking, and astronomy. He is a member of the CarverSite, DIY Audio, and Audio Circle web forums, and he collects vintage Carver audio gear. To date, he has built four pairs of electrostatic loudspeakers for himself and others, and he continues to gather and share information with ESL builders on the Internet. He also maintains a webpage dedicated to building DIY electrostatic loudspeakers.

Steve Tatarunis ("The Voice Coil (Part 2): Materials and Quality," p. 8) is a senior engineer at Menlo Scientific and a 30-year veteran of the loudspeaker industry. He runs Menlo's Boston-area test lab where he tests speakers, microphones, headsets, and other audio devices for Menlo's clients.

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