Exploring the Linkwitz Equalizer, p. 22

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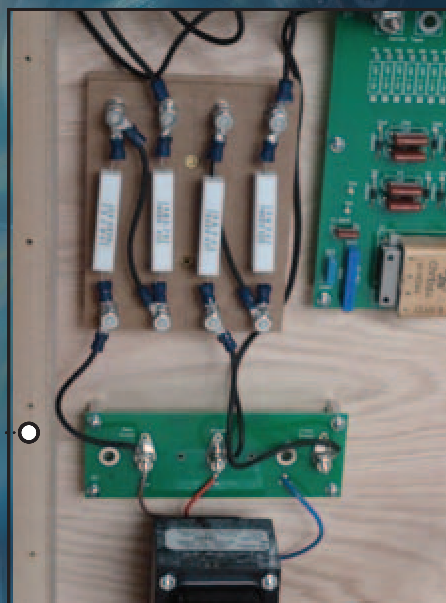
Two-Way Active Speaker Design

High-Performance 35-W Bullet Tweeter



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From the Editor's Desk

What's Inside Matters

The first issue of *Circuit Cellar* magazine—a sister publication of *audioXpress* that's focused on embedded computer engineering—was printed in 1988 with a now-famous quote on the cover: *inside the box still counts*. Since then, engineers and innovators have used the line to educate their nontechnical peers about the amazing technologies inside the devices they use every day. The line was on my mind as I chose articles for this issue. When it comes to audio systems like speakers and tube amps, the quality and design of the technologies inside the cabinets and enclosures are essential.

Turn to page 12 to learn about the technology Ton Giesberts implemented to improve his PC's sound. It's the first article in a series about an active loudspeaker system.

On page 22, Giovanni Bianchi elucidates some theory, math, and science behind loudspeaker sound. He details the Linkwitz equalizer and its limitations, and then proposes a possible solution.

The *inside the box* theme continues with Mike Klasco and Steve Tatarunis's article on ribbon and planar magnetic louspeakers (p. 8). They provide thought-provoking details about topics ranging from "pure" ribbon topology to ribbon transducer manufacturing.

As usual, Vance Dickason puts a speaker to the test. This month he presents what he learned about Faital Pro's FD371 bullet-type, high-SPL tweeter (p. 18).

Richard Honeycutt wraps up the issue with "The Differences in Amp Sound" (p. 31). What goes on inside tube amps and solid-state amps matters. Richard weighs in on the debate that will likely keep audiophiles arguing for decades to come.

Regards,

C. J. Abate
editor@audioxpress.com

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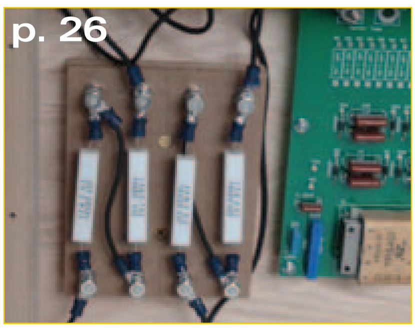

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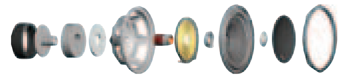
By Mike Klasco and Steve Tatarunis (United States)

Ribbon and Planar Magnetic Loudspeakers (Part 1)



Speakers offer smooth, transparent, and crisp sound quality

The vast majority of speakers used for speech and music consist of a tubular voice coil, a magnetic structure, and a cone. Yet for more than a half century, a number of ribbon speakers have been commercially available. This month, we will explore the reasons why the ribbon speaker continues to deserve its place in the sun.

PURE RIBBONS

The ribbon speaker's original implementation consists of a thin metal foil ribbon suspended in a magnetic field with the upper and lower points from which the metal foil is suspended also serving as the signal's positive and negative terminals. Today, the majority of ribbon planars use a derivative approach—essentially a flex circuit board with a thin polymer film with a conductor pattern that functions as both the diaphragm and the voice coil (see **Figure 1**).

The first conception of the "pure" ribbon topology was a ribbon microphone invented in 1924 by Walter H. Schottky, a noted European physicist, and Erwin Gerlach. (Note: Don't confuse Walter Schottky, creator of the Schottky diode, with another famous semiconductor scientist, William Shockley, co-inventor of the transistor.) They also invented a ribbon loudspeaker simply by reversing a microphone's physical effects.

Schottky's ribbon comprised an extremely thin corrugated (zig-zag cross section) ribbon of aluminum placed between a permanent magnet's poles. The electrical signal is applied to the ribbon, which moves with it to create the sound (see **Photo 1**).

A ribbon driver's advantage is that its diaphragm has little mass and it can accelerate quickly, yielding extended high-frequency responses. Compared to a dome tweeter, the ribbon driver's voice coil generates the sound energy, which then passes through the glue joint to drive the dome. This construction results in propagation delays, resonances, and other spurious acoustical by-products. On a pure ribbon, the voice coil is the aluminum foil diaphragm itself, eliminating some of the issues that plague voice coil driven cones and domes. For the more contemporary planar polymer film speakers, the diaphragm is driven in-phase throughout the diaphragm as the voice coil conductor is evenly distributed over the substrate. Early ribbon mics and loudspeakers were fragile. Some could be torn by a strong gust of air or, in the case of mics, a trumpet's toot. Most ribbon tweeters radiate sound in a dipole pattern. However, some have back chambers that block the dipole radiation pattern and prevent the diaphragm from being overdriven and modulated by the woofer when mounted in the same enclosure.

Pure ribbons just using the aluminum foil as the diaphragm have a low resistance that most amplifiers cannot directly drive. As a result, a step-down transformer is typically used to increase the current through the ribbon. The amplifier "sees" a load that is actually the ribbon's resistance times the transformer's turns ratio squared. As with mic applications, the transformer must be carefully designed so its frequency response and parasitic losses do not degrade the sound, further increasing costs and complications relative to conventional designs.

The British were among the first to produce the ribbon loudspeaker, introduced by Quad (see **Photo 2**).

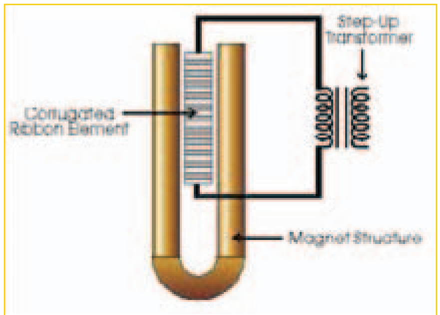


Figure 1: This is the front view of a ribbon microphone. (Photo courtesy of Shure Corp.)

The 1950 Quad Corner Ribbon preceded its introduction of the electrostatic loudspeaker (ESL) in 1954. Stanley Kelly's "Kelly 30" ribbon horn tweeter soon followed, generating interest from DECCA, which manufactured the Decca Kelly 30 (DK30) and the larger Decca London Ribbon. Many audiophile brands continue to offer pure ribbon speakers including Fountek, Orca, Raven, Aurum Cantus, LCY, and RAAL.

PLANAR POLYMER RIBBONS

Today, the most common planar ribbon topology is for a non-electrically

Photo 1: The RCA D40A (circa 1932) is an early example of a ribbon mic. (Photo courtesy of Wikipedia)

Photo 2: The Quad Corner Ribbon loudspeaker is one of the earliest examples of a ribbon speaker. (Photo courtesy of Audiosmile.com)

conductive film substrate to which an aluminum or copper conductor pattern is laminated. For most designs, the film is tensioned, although non-tensioned "self-supporting" techniques have been successfully commercialized for smaller tweeters. Traditionally, a thin polyester film called mylar was used. For high-power applications, polyimides (e.g., Kapton) was used. For medium power, Teonex polyester was used. The polymer/conductor composite diaphragm has the same construction as the flexible circuit boards commonly used in laptops, cell

Photo 3: Jim Winey was working on an application that involved laminating tape to flexible magnets when the idea for a large-format ribbon planar diaphragm speaker occurred to him. He is pictured with the Magnepan Speaker. (Photo courtesy of Magnepan)

phones, and tablets. The impedance of this conductor pattern/diaphragm can be readily designed by the conductor length, width, and thickness the amplifier can drive—not just a reasonable impedance, but also essentially a pure resistive load. Planar magnetic speakers (i.e., speakers with printed or embedded conductors on a flat diaphragm) are also considered ribbons, although purists might be insulted. The term "planar" is generally reserved for speakers with rectangular, flat surfaces that radiate in a bipolar (i.e., front and back) manner.

DESIGN HISTORY

Engineer Jim Winey began the initial conception and development of the planar in 1966 at the 3M Company in Minnesota. Winey was working on an application that involved laminating tape to flexible magnets when the idea came to him. Development of something workable followed during the next few years and the first full-range, large-format ribbon planar diaphragm speaker was built in early 1971 (see **Photo 3**). Background information is available at *Stereophile* magazine (www.stereophile.com/interviews/103winey/index.htmllink).

Magnepan's reproducing mechanism comprises thin conductive wires and foil strips attached to a thin sheet of mylar, residing in a magnetic field created by a vertical array of permanent strip magnets. When an amplified signal is applied to the conductors, the resultant electrical forces react with the magnetic field to excite the polymer film sheet, which projects sound as a dipole.

The "leaf tweeter" flex circuit board diaphragm variant of Winey's invention was introduced by Mashitusita's Panasonic Acoustic Research Lab in Osaka, Japan, in the mid-1970s. It was offered as its Technics brand "leaf" tweeter (for the conductor pattern on the substrate). These planar magnetic speakers consisted of a flexible membrane with a voice coil printed or attached to it. This approach is commonly used today. The current flowing through the coil interacts with the magnetic field of carefully placed magnets on either side of the diaphragm.

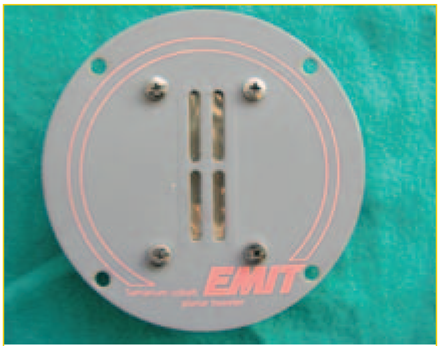


Photo 4: Infinity's EMIT tweeter used powerful Samarium cobalt magnets covered by a thin mylar polyester diaphragm.

The driving force covers a large percentage of the membrane surface and reduces resonance problems inherent in coil-driven flat diaphragms. A Japanese competitor, Fostex (a premium group at Foster), followed with its own leaf tweeter models and coined the phrase "uniform phase" due to the even driving force of the distributed voice coil conductor pattern. Following these flagship products, Panasonic affiliate JVC and Foster supplied mid-priced ribbon leaf tweeters and midranges on an OEM/ODM basis. Infinity has featured its Electro Magnetic Induction Tweeter (EMIT) and Electro Magnetic Induction Midrange (EMIM) ribbons since the late 1970s. Offshore ODM vendors for these EMIT round "isodynamic" ribbon tweeters included Sawafuji in Japan and later Meiloon. These days, there are quite a few others. The Infinity Quantum Series were the first to introduce the famed EMIT tweeters into the U.S. market. EMITs use powerful (for that pre-neodymium time) Samarium cobalt magnets covered by a thin mylar polyester diaphragm. Eventually, these EMITs evolved into ribbon midranges and even upper bass drivers (see **Photo 4**).

Ribbon transducer manufacturing has traditionally been a specialized craft, even when it is part of larger speaker manufacturing operations (e.g., Foster and Matsushita). There have been several popular ribbon tweeters used in many speaker systems, but full-range ribbon speaker systems had been limited to small-batch, high-end audiophile production. A few years ago, Sonigistix of Canada introduced its Monsoon ribbon multimedia system. Using ribbon "full-range" satellites and a conventional cone subwoofer, the Sonigistix Monsoon MM-700 was the first multimedia mainstream ribbon product to achieve mass-market success. Tymphany now owns the Sonigistix technology and its engineering staff includes members of the orginal Sonigstix design team.

HIGH-QUALITY SOUND

For professional and commercial sound installations, market acceptance is limited because ribbon speakers are perceived as expensive, having temperamental ribbon diaphragm assemblies, limited dynamic range, and requiring a larger surface radiating area for adequate low-end response. But, their smooth, transparent, and crisp sound quality and their inherent low distortion have made the development of wider dynamic range ribbons a seductive goal. Can the ribbon speaker achieve the wide dynamic range with low distortion for high output audiophile and professional applications (e.g., studio monitors and even concert sound)? Tune in next month for Part 2. *aX*

 By Ton Giesberts (The Netherlands)

Active Loudspeaker System (Part 1)

Multimedia Applications

Loudspeakers designed to boost your PC's sound system

The two-way active loudspeakers described here are primarily intended for use with a PC, but they could, in principle, be used in any other "medium-fi" application (see **Photo 1**).

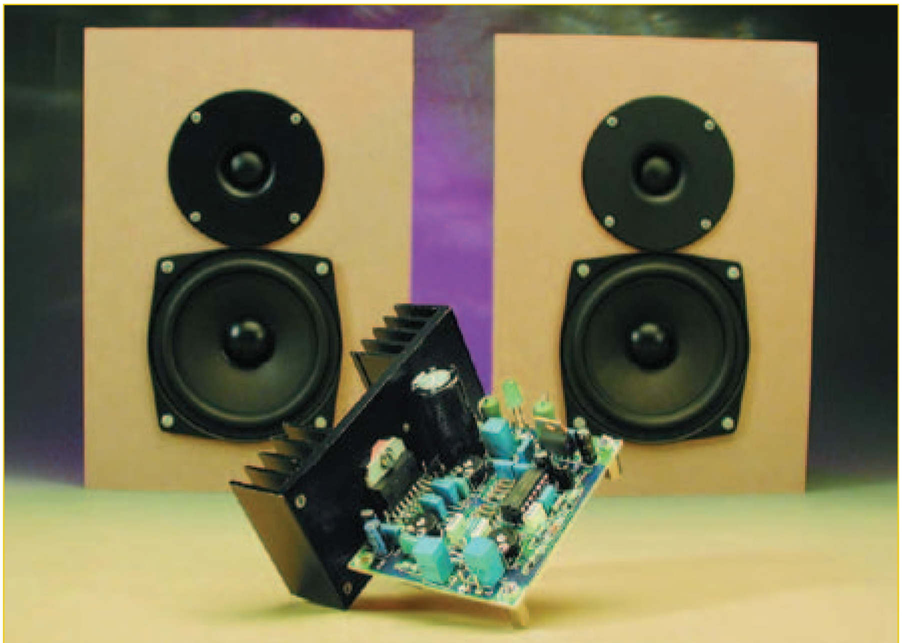


Photo 1: These two-way active speakers are well suited for PC use.

Since the speakers' universal crossover filter can be adjusted to your liking, you're not tied to using the speakers and enclosure suggested in this article. Other woofer/tweeter combinations may also be used.

This design's impetus was my irritation with the majority of PC loudspeakers' average-to-mediocre quality. I thought the speakers could be improved without expending too much effort or money. Of course, you don't need the ultimate Hi-Fi quality for a PC, but it would be nice to have a set of decent reproduction-quality loudspeakers.

I sought a compact woofer/tweeter combination that offered a good performance at a reasonable cost. Another requirement was that they should be magnetically shielded, because the loudspeakers will likely be placed close to a monitor or TV, which would be affected by stray magnetic fields. For the prototype, I used a Visaton 25-mm dome tweeter (SC10N) and a 13-cm bass/midrange unit (SC13). Vifa and Monacor (Monarch) also manufacture viable magnetically shielded speakers. Other speaker combinations with associated enclosures (recommended by the manufacturer) can also be used in this design.

I kept the electronics required to make the loudspeaker active as simple and adaptable as possible. A two-way crossover filter was designed around a quad op-amp, with a choice of slopes, characteristics, and crossover frequencies.

For use with the recommended speakers, the filter was set up as a third-order Butterworth type with a 4-kHz crossover frequency. For the power amplifiers, I used an integrated dual-bridge amplifier, which requires few external components. At a 16-V supply voltage, it delivers 2 × 19 W into 4 Ω or 2 × 12 W into 8 Ω. Compared to usual Hi-Fi standards this may seem a bit skimpy, but combined with average-efficiency speakers, a 100-dB sound pressure level can be attained (and that is really loud!).

One of the advantages of using an "active" design is that it enabled me to overcome a disadvantage of many small loudspeaker enclosures. Most enclosures with a volume of a few liters tend to have a poor response at the lower end of the frequency range. For this reason, I have included a corrective filter that can be activated with a jumper and boosts the frequencies between 100 Hz and 1,000 Hz up to a maximum of 6 dB at 100 Hz. My prototype seemed to benefit from this addition and this correction may also have a positive effect with other speaker combinations with similar bass/midrange units.

ELECTRONICS DESIGN

Figure 1 shows the circuit diagram, which consists of three distinct sections: the input buffer and supply, the filter, and the power amplifier.

Looking at the circuit diagram from left to right, you can see a terminating resistor, an isolating capacitor, and trim pot P1. This feeds the signal into an input buffer built around IC1a, which forms part of quad rail-to-rail op-amp TS924IN. This type is distinguished by its relatively large output current of up to 80 mA.

R2 and C2 form a low-pass filter that suppresses any high-frequency interference. The network C3/R3/R4 provides the previously mentioned correction at low frequencies. Closing JP1

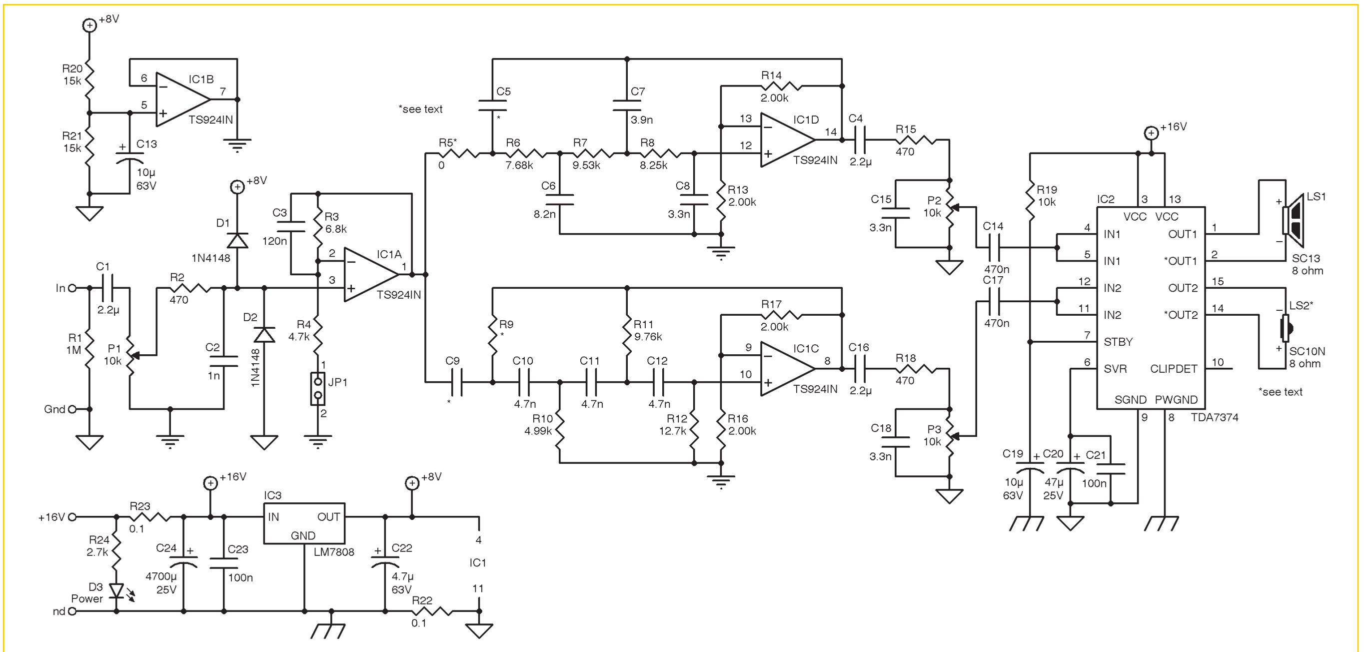


Figure 1: The circuit includes an input buffer, a crossover filter, and an integrated dual-power amplifier.

enables this filter. To obtain the optimum signal processing by the buffer and filters without relying on a symmetrical power supply, I used IC1b to create a stable virtual ground. C13 decouples the output of potential divider R20/R21, providing a virtual ground at exactly half the supply voltage. The large output current capability of the TS924IN is obviously a great advantage in the circuit round IC1b.

Op-amp IC1 has been equipped with its own voltage regulator (IC3). This keeps supply-borne interference from affecting the input buffer and filters. For proper operation of this 8-V regulator, the supply voltage to the active system should be at least 11 V. Resistor R22 has been added to separate the signal ground and supply ground for cases where a single power supply is used for two or more channels. When each channel has its own power supply, R22 can be replaced with a wire link.

Next comes the crossover filter. As shown in **Figure 1**, the input buffer's output is fed to two filter sections. These are built around the two remaining IC1 op-amps. The low-pass filter is built around IC1d and the high-pass filter is found around IC1c. For the filter design, I started with a fourth-order configuration, so the same PCB could be used with simpler filters by leaving out some components.

I calculated the values for several variants. **Table 1** shows the component values for a third-order Butterworth filter and a fourth-order Linkwitz-Riley filter with crossover frequencies at 1, 2.5, and 4 kHz.

With the Visaton loudspeakers used

	Third-Order Butterworth			Fourth-Order Linkwitz-Riley		
	1 kHz	2.5 kHz	4 kHz	1 kHz	2.5 kHz	4 kHz
R5	link	link	link	6k34	6k34	6k98
R6	8k06	6k65	7k68	13k3	9k31	10k7
R7	8k25	8k45	9k53	6k49	7k68	8k25
R8	6k81	8k06	8k25	9k31	7k50	8k45
C5	open	open	open	22 n	10 n	5n6
C6	33 n	15 n	8n2	39 n	18 n	10 n
C7	18 n	6n8	3n9	18 n	6n8	3n9
C8	15 n	5n6	3n3	8n2	3n9	2n2
R9	open	open	open	7k50	6k49	7k15
R10	5k23	4k53	4k99	3k83	3k32	3k65
R11	10k2	8k87	9k76	11k0	9k53	10k5
R12	13k0	11k5	12k7	19k6	16k9	18k7
C9	link	link	link	18 n	8n2	4n7
C10	18 n	8n2	4n7	18 n	8n2	4n7
C11	18 n	8n2	4n7	18 n	8n2	4n7
C12	18 n	8n2	4n7	18 n	8n2	4n7

Table 1: The component values for the filters are shown at different frequencies. For the third-order Butterworth filter C5 and R9 aren't mounted and R5 and C9 are replaced by wire links.

in my prototype, I found that a third-order Butterworth filter with a crossover frequency of 4 kHz gave the best result. The component values given in the circuit diagram are for this filer type. When we used a fourth-order Linkwitz-Riley filter at 4 kHz, the measurements showed there were problems with the radiation pattern due to the large phase shifts this filter produces. The third-order Butterworth variant suffered noticeably less from this. The Linkwitz-Riley values are mainly included for experimentation purposes.

Take care connecting the speakers with the Linkwitz-Riley filter because the connections to the tweeter are reversed. On the PCB, the indicated polarity is for use with third-order Butterworth filters. (In this case, the tweeter is "out of phase" compared to the woofer!) As a final point of interest, the Butterworth filter's crossover point is at −3 dB. For Linkwitz-Riley filters, it is at −6 dB.

The filters' output signals are fed to the power amplifiers via presets P2 and P3. The potentiometers compensate for the different efficiencies of the woofer and tweeter. Many dome tweeters are about 3 dB louder than small bass/midrange speakers at the same input level. However, the loudspeakers chosen in my design have similar efficiencies, so both P2 and P3 can be turned to their maximum levels.

For the power amplifier, I used a TDA7374B double-integrated amplifier. These are primarily intended for automotive use, but they are also suitable in these applications. This IC requires surprisingly few external components (no Boucherot network or output capacitors) and contains adequate internal

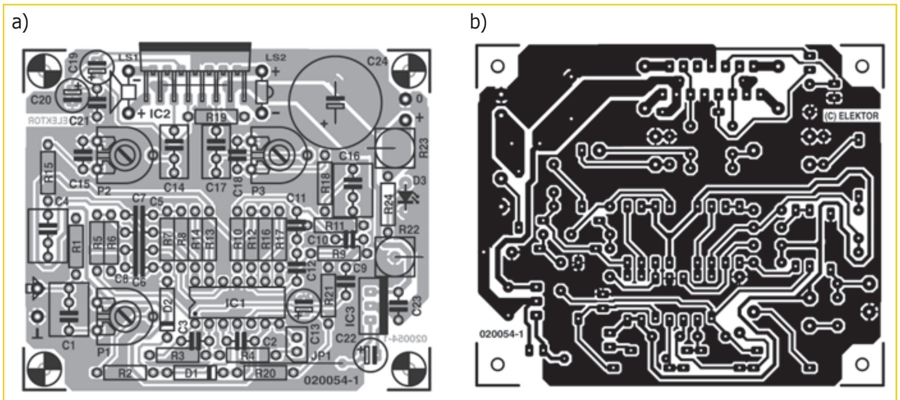


Figure 2: This compact PCB contains a crossover filter as well as a 2 × 20-W power amplifier.

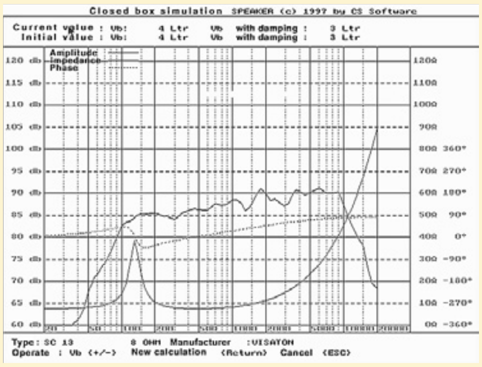
protection circuits to aid against overheating and short circuits. With these features, my dual-power amplifier is a shining example of compactness. This is clearly seen in the circuit diagram.

I already mentioned the output power. With a load of 8-Ω loudspeakers, cool the IC2 using a small heatsink with a thermal resistance of 3 K/W. R19 and C19 ensure that virtually no plops are heard when the amplifier is switched on. (There is always a small offset voltage at the amplifier's output.) RC networks R15/C15 and R18/C18 restrict

SPECIFICATIONS (using a 16-V supply voltage)

Input impedance	10 kΩ
Sensitivity (12 W/8 Ω, JP1 open, P1, P2, P3 max.)	270 mV
Distortion + noise (1 W/8 Ω, 1 kHz)	0.013% (B = 80 kHz)
Bandwidth woofer amplifier (P2 max., JP1 open)	32 Hz to 4 kHz
Bandwidth woofer amplifier (P2 half, JP1 open)	25 Hz to 4 kHz
Bandwidth tweeter amplifer	4 kHz to 45 kHz
Output power per amplifier (THD + N = 0.5%)	12 W (8 Ω)
	19 W (4 Ω)
Quiescent current (no load)	0.17 A
Bandwidth amplifer + box (−3 dB)	100 Hz to 18 kHz

a)



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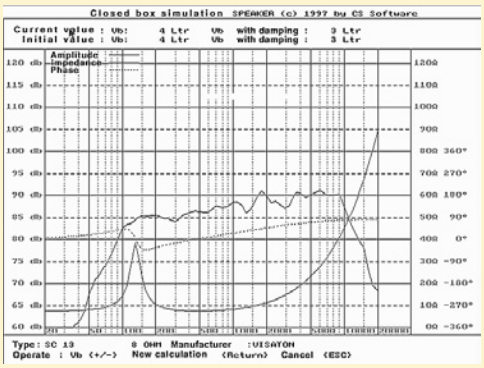

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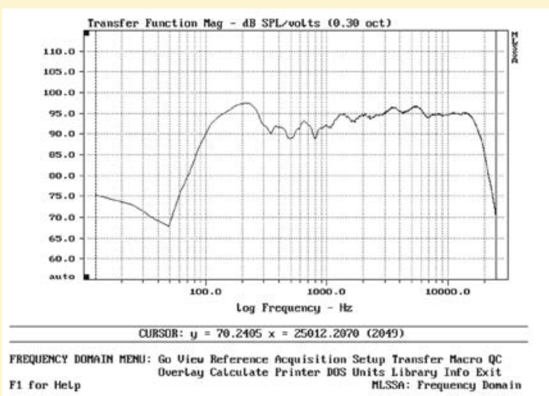

The first graph (a) is a frequency-response simulation for the SC13 woofer. The absence of prominent peaks or troughs shows a steep drop at the lower end of the frequency range. The second graph (b) shows the low-frequency booster and the filters' measured response. The loudspeakers' measured frequency response when driven by their amplifiers is shown the third graph (c). The low-frequency boost should have been set a bit higher. When the loudspeaker is placed on a desk or near a wall, the lower frequencies will be boosted higher, making the curve somewhat flatter.

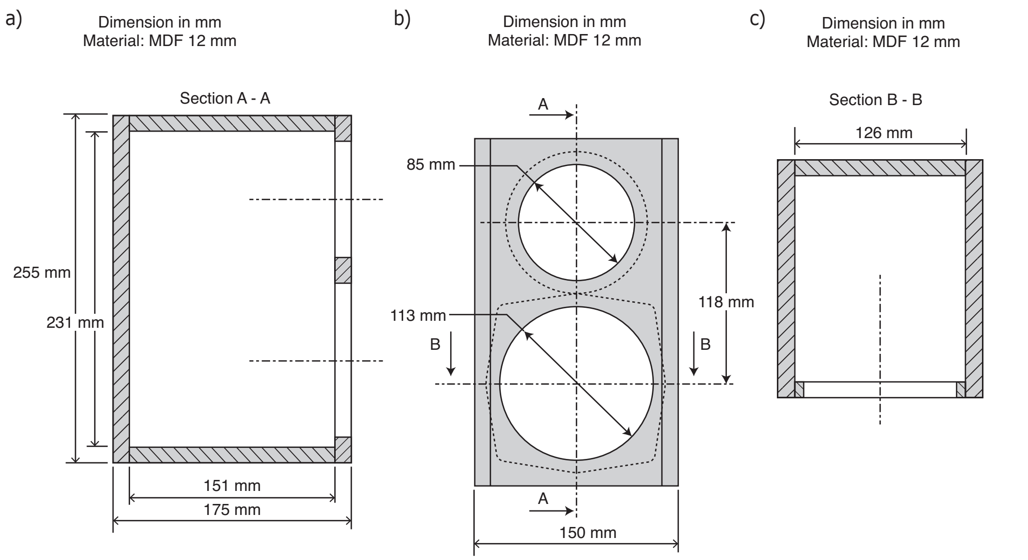

Figure 3: These design drawings for the enclosure are complete with all the measurements, assuming that 12-mm thick MDF board is used.

the power amplifier's bandwidth to minimize any HF interference effect. It would have been better to place these networks after the potentiometers, but then the bandwidth will vary noticeably. C20 decouples the internal potential divider, which supplies several stages with half the supply voltage and is also responsible for supply ripple suppression, about 50 dB at 100 Hz.

THE PCB

Figure 2 shows the PCB designed to hold the active two-way loudspeaker's electronics.

There is not much to be said about the PCB. The layout is fairly well organized and the various connections are clearly marked. At the bottom left are the input pins, diagonally opposite (top right) are the connections for the supply voltage and just below that is the power LED D3. The connectors for the woofer and tweeter (LS1 and LS2) are on either side of IC2.

There are two wire links on the board worth mentioning: one just next to R20 and another underneath (!) the IC2 pins. The latter could also be soldered to the PCB's underside, but in either case, an isolated wire should be used for this link. IC2 has been placed near the PCB's edge, making it easy to mount a small heatsink (3 K/W) to it. Remember to use an isolating washer between the IC and heatsink!

Once the PCB has been populated and tested, there are several possibilities for completing the construction. It can be mounted inside the loudspeaker enclosure, it can be mounted in a separate box (e.g., a two-channel version), or it can be combined with a subwoofer. A separate box isn't a bad idea since I may add a tone control to this system. But, the choice is yours.

The supply can be provided with a transformer, a bridge rectifier, and a smoothing capacitor. Each channel requires a 12-V/15-VA transformer and a 4,700-μF/25-V smoothing capacitor. For a stereo version, these values should be doubled. When a stabilized supply is used, the voltage may be increased from 16 V to a maximum of 18 V, which slightly increases the output power.

WOODWORK

The loudspeaker enclosure's size and construction depends primarily on the size of the woofer used. The Visaton SC13 used here is mounted in a closed box with a volume of about 4 liters. This makes the construction a fairly straightforward job, since a closed box is little more than six panels

PARTS & COMPONENTS

Capacitors
- C1, C4, C16 = 2-μF2 MKT, lead pit 5 or 7.5 mm
- C2 = 1 nF, lead pitch 5 mm
- C3 = 120 nF, lead pitch 5 mm
- C5 = open*
- C6 = 8 nF2, lead pitch 5 mm
- C7 = 3 nF9, lead pitch 5 mm
- C8, C15, C18 = 3 nF3, lead pitch 5 mm
- C9 = wire link*
- C10, C11, C12 = 4 nF7, lead pitch 5 mm
- C13, C19 = 10 μF, 63-V radial
- C14, C17 = 470 nF
- C20 = 47 μF, 25-V radial
- C21, C23 = 100 nF, lead pitch 5 mm
- C22 = 4 μF7 63-v radial
- C24 = 4,700 μF, 25-V radial, lead pitch 7.5 mm, 17-mm diameter max.

Miscellaneous
- Heatsink for IC2: 3 kW
- JP1 = two-way pinheader with jumper
- LS1 = SC13 8-Ω Visaton (Conrad Electronics)
- LS2 = SC10N 8-Ω Visaton (Conrad Electronics)
- PCB
- Wadding material (BAF)
- Wood: 12-mm MDF (see Figure 3)

Resistors
- R1 = 1 MΩ
- R2, R15, R18 = 470 Ω
- R3 = 6 kΩ8
- R4 = 4 kΩ7
- R5 = 0 Ω
- R6 = 7 kΩ68
- R7 = 9 kΩ53
- R8 = 8 kΩ25
- R9 = open*
- R10 = 4 kΩ99
- R11 = 9kΩ76
- R12 = 12 kΩ7
- R13, R14, R16, R17 = 2 kΩ00
- R19 = 10 kΩ
- R20, R21 = 15 kΩ
- R22, R23 = 0Ω1, 5 W
- R24 = 2kΩ7
- P1, P2, P3 = 10-kΩ preset

Semiconductors
- D1, D2 = 1N4148
- D3 = LED, green, high efficiency
- 1C1 = TS9241n (ST from Farnell)
- 1C2 = TDA7374B (ST from C-1 Electronics, www.dil.nl)
- 1C3 = 7808

*see text and Table 1

glued together. This may seem a bit difficult to the inexperienced carpenter, but once the wood has been cut to size, the rest of the job should be simple if you use some clamps. In any case, no special baffles or ports are required. The only part that is a bit tricky is cutting out holes for the speakers.

Due to the box's small dimensions, it is unnecessary to use thick wood (although you could if you wanted). I have based the drawing on MDF board with a 12-mm thickness (see **Figure 3**). This drawing shows all the details. The enclosures required acoustic damping, which is done by filling it loosely with polyester wadding. The connectors can be mounted on the back panel. When the electronics are mounted inside the enclosure, it is important that the heatsink is on the outside.

It is unnecessary to strictly adhere to the design shown in **Figure 3**. A different style of enclosure is acceptable, as long as its volume is near the recommended 4 liters. **Photo 1** shows that I've also deviated from the standard "shoebox" design for the prototypes. I tried some triangular designs for my enclosures, mainly for originality. An advantage is that there won't be any standing waves, but this comes at the expense of added complexity, since three of the five panels now have edges with "difficult" 30° and 73.9° angles! The front and base panels are the easiest to make since they still have 90° edges. I used an equilateral triangle for the base panel and the top panel slopes down from the front at a 30° angle.

There are several options for finishing off the enclosures. An "old fashioned" veneer is a possibility, but the enclosures could also be covered with self-adhesive vinyl or another material. Spraying them with paint is another popular option. For the perfect finish, you could have it professionally done at a garage. aX

Faital Pro's New Bullet Tweeter

A pro sound tweeter with a ferrite motor structure

Photo 1: Faital Pro's new FD371 bullet-type tweeter

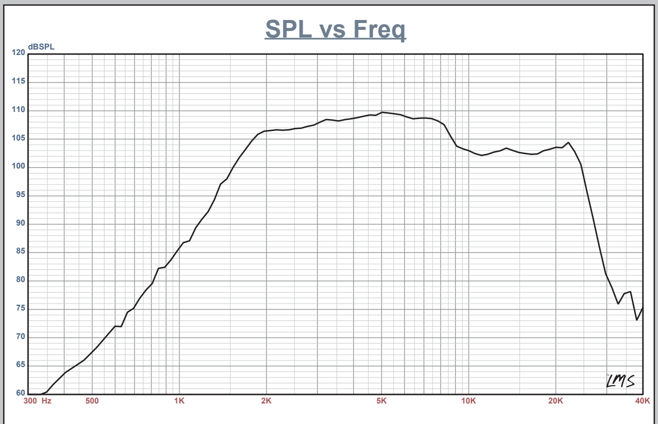

Figure 2: Faital Pro FD371's on-axis response

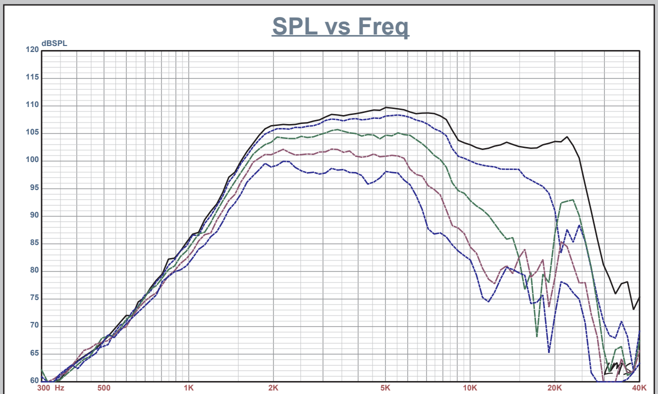

Figure 3: Faital Pro FD371's on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot; 60° = dash)

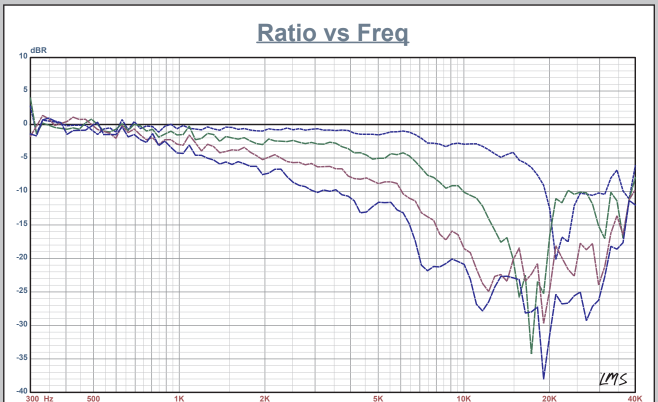

Figure 4: Faital Pro FD371's normalized on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot)

The Faital Pro FD371 is a bullet-type, high-SPL tweeter (see **Photo 1**). Faital Pro's FD371 is built around a CNC'd aluminum horn and phase plug. Other features include a Ketone polymer annular-shaped diaphragm, a 37-mm (1.4") diameter voice coil using a Kapton former wound with aluminum wire, and a ferrite motor structure with polished front and back plates.

Testing commenced using the LinearX LMS analyzer to produce the 300-point sine wave impedance plot (see **Figure 1**). With a 5.6-Ω DCR, the FD371's minimum impedance above resonance was 7 Ω at 7.79 kHz. The FD371's primary resonance occurs at 1.79 kHz with an 11.2-Ω magnitude.

Next, I mounted Faital Pro's FD371 in an enclosure with a 15" × 6" baffle and measured the on- and off-axis frequency response at 2.83 V/1 m with a 100-point gated sine wave sweep. **Figure 2** shows the bullet

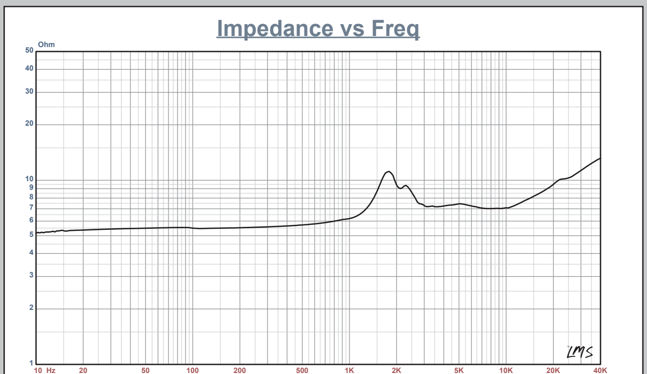

Figure 1: Faital Pro FD371's free-air impedance plot

tweeter on-axis and shows the sensitivity varies between 102 and 109 dB. The company-specified 107-dB sensitivity falls between those measurements. This SPL profile measures ±3 dB from 2 to 9 kHz. The FD371 pro sound tweeter's recommended minimum crossover frequency is 2.5 kHz, which is low for this type of device. The response is ±3.75 dB from 2 to 20 kHz.

Figure 3 shows the on- and off-axis SPL out to 60° off-axis, with the normalized view shown in **Figure 4**. You can get a graphic look at the device's directivity in the factory beamwidth plot shown in **Figure 5**. **Figure 6** shows the two-sample SPL comparison, indicating both samples are well matched.

For the remaining tests, I used the Listen SoundCheck analyzer, 0.25" SCM microphone and power supply to measure distortion and generate time-frequency plots. For the distortion measurement, the FD371 tweeter was mounted on the same baffle used for the frequency-response measurements, and a pink noise stimulus was used to set the SPL to 104 dB at 1 m (0.64 V). I measured the distortion with the Listen microphone placed 10 cm from the horn's mouth. This produced the distortion curves shown in **Figure 7**. I then used SoundCheck to measure the Faital Pro driver's

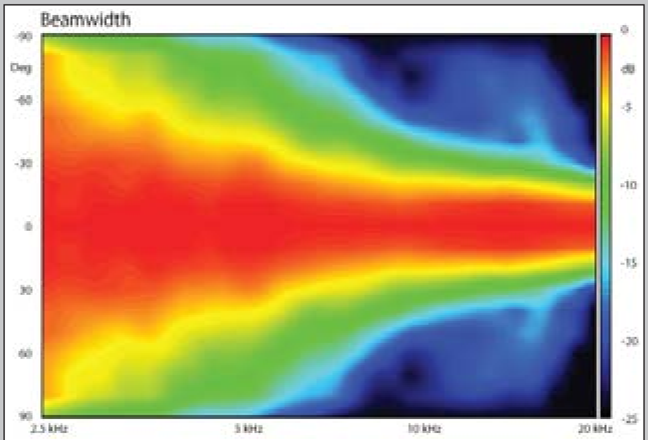

Figure 5: Faital Pro FD371's factory beamwidth plot

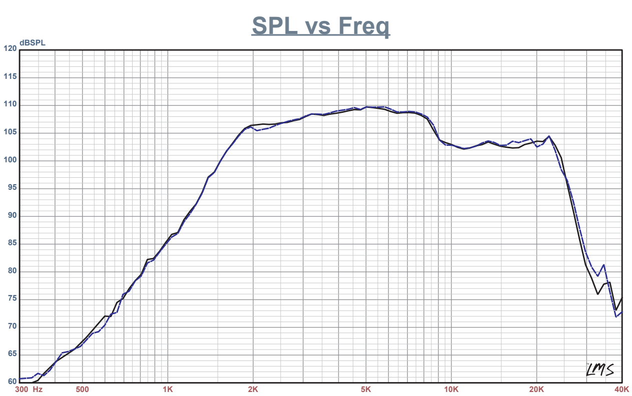

Figure 6: Faital Pro FD371's two-sample SPL comparison

2.83-V/1-m impulse response and imported the data into Listen's SoundMap time/frequency software. The resulting CSD waterfall plot is shown in **Figure 8.** The short-time Fourier transform (STFT) plot is shown in **Figure 9**.

I haven't had a chance to verify the results as yet, but Keith Gronsbell, Faital Pro's U.S. representative, said the Ketone polymer diaphragm yields a smooth subjective performance that is considerably more musical than the bullet tweeters from the pro sound's past.

For more information about this driver and other Faital Pro pro sound products, visit the company's website at www.faitalpro.com, or in the U.S., contact Faital USA, Inc., Keith Gronsbell (kgronsbell@faital.com), 220 West Parkway, Unit 13, Pompton Plains, NJ, 07444; (516) 779-0649; Fax (973) 835-5055. *aX*

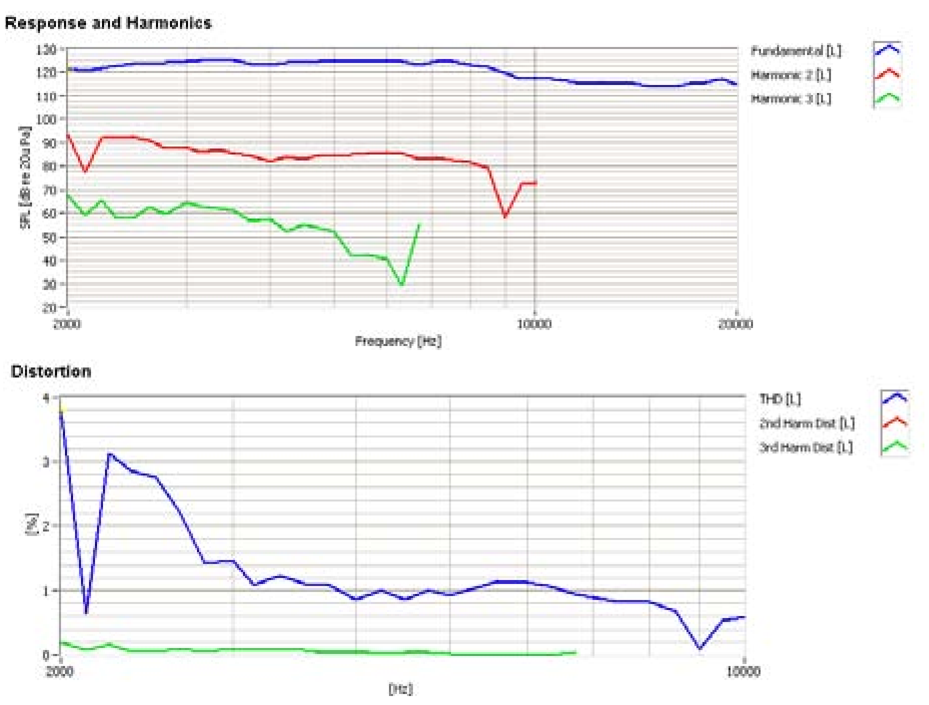

Figure 7: Faital Pro FD371 SoundCheck distortion plots

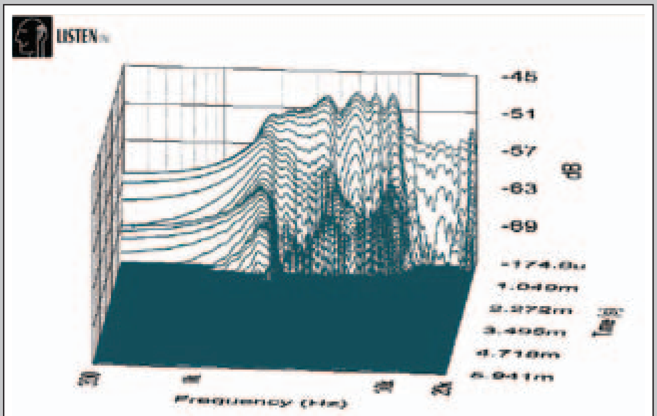

Figure 8: Faital Pro FD371 SoundCheck CSD waterfall plot

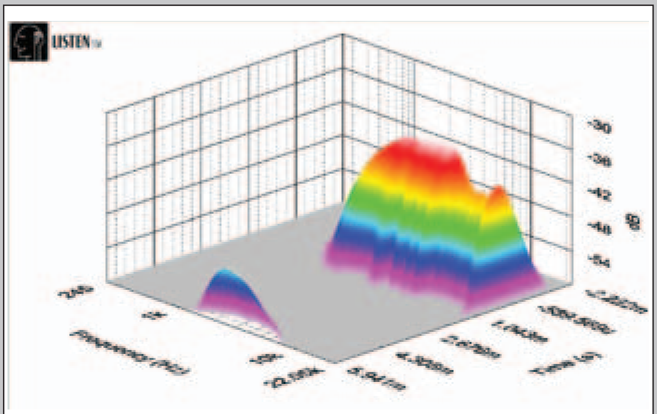

Figure 9: Faital Pro FD371 SoundCheck STFT surface intensity plot

By Giovanni Bianchi (Italy)

Generalized Linkwitz Equalizer

A possible solution to the equalizer's limitations

"Io stimo più il trovare un vero, benchè di cosa leggiera, che l' disputar lungamente delle massime questioni senza conseguir verità nissuna." Translation: "I consider it more important to find the truth about a small thing than to engage in long discussions about big questions without finding any truth."—Galileo Galilei (1564–1642), Opere, IV, *p. 738*

Any closed-box loudspeaker presents a response similar to the one shown in **Figure 1,** at least in the low-frequency range. A loudspeaker is a high pass with 40-dB/decade asymptotic slope. Sometimes that slope is referred to as a 12-dB/octave, which is $40 \times \log_{10}(2) \cong 12.041$, and is equivalent to 40 dB per decade.

Moreover, the six curves differ from their damping (or quality) factor. As some readers know, $Q_C \leq 1/\sqrt{2}$ means no peak in the frequency response. $Q_C = 1/\sqrt{2}$ defines the "maximally flat" or second-order Butterworth response, and $Q_C = 1/2$ corresponds to the critical damping (i.e., the fastest decay in the transient response). As Q_C becomes greater than $1/\sqrt{2}$, the frequency response assumes positive peaks with amplitude increasing with Q_C itself, while simultaneously in the time domain, the transient response becomes slower and slower.

The curves of **Figure 1** are the logarithm representation of transfer functions of the type:

$$T(f)=\frac{\left(j\frac{f}{f_C}\right)^2}{1+j\frac{1}{Q_C}\frac{f}{f_C}+\left(j\frac{f}{f_C}\right)^2} \qquad (1)$$

where f_c is the resonant frequency, Q_C is the quality factor, and $j = \sqrt{-1}$ is the imaginary unit. Note that:

$$20\log_{10}\left[\left|T(f)\right|\right]=10\log_{10}\left[\left|T(f)\right|^2\right] \qquad (2)$$

The x-axis of **Figure 1** is indeed the normalized frequency f/f_C.

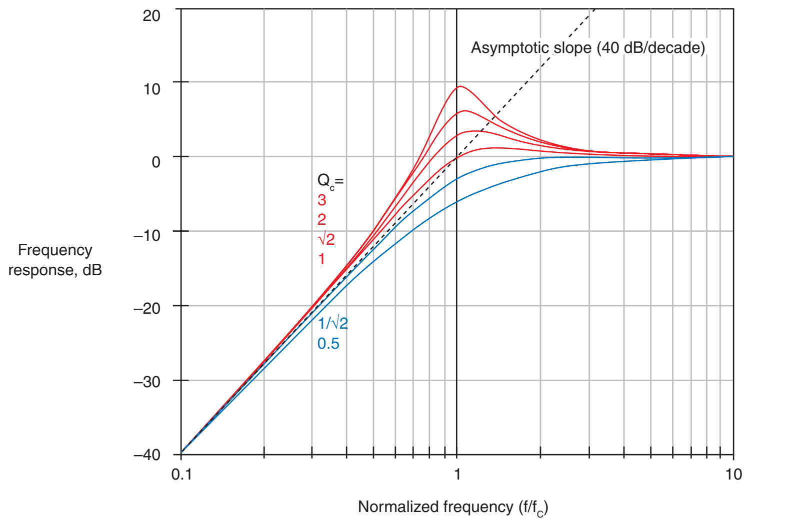

Figure 1: This graph shows the doubly normalized amplitude response of a closed-box loudspeaker. The x-axis normalization is in response to the resonant frequency (f_C), the y-axis normalization is in response to the high-frequency value. The quality factors are labeled from the highest to the lowest curve.

The complex-values transfer function (1) corresponds to the squared amplitude:

$$|T(f)|^2=\frac{\left(\frac{f}{f_C}\right)^2}{\left[1-\left(\frac{f}{f_C}\right)^2\right]^2+\frac{1}{Q_C^2}\left(\frac{f}{f_C}\right)^2} \qquad (3)$$

The function (3) reaches its maximum ($Q_C = 1/\sqrt{2}$) when:

$$f=f_{PEAK}=\sqrt{\frac{2Q_C^2}{2Q_C^2-1}}f_S \qquad (4)$$

That maximum is:

$$\max\left\{|T(f)|^2\right\}=\left|T\left(\frac{f}{f_S}=\sqrt{\frac{2Q_C^2}{2Q_C^2-1}}\right)\right|^2=\frac{4Q_C^4}{4Q_C^2-1} \qquad (5)$$

The 3-dB frequency is provided by:

$$\frac{f_{3dB}}{f_C}=\sqrt{\frac{1}{2Q_C^2}-1+\sqrt{1+\left(\frac{1}{2Q_C^2}-1\right)}} \qquad (6)$$

The 3-dB frequency is defined as the one that makes square amplitude response (3) equal to one half: $10 \times \log_{10}(1/2) = 3.0103$ ("precisely").

Table 1 lists the values of the parameters (4) to (6) for different values of the quality factor. One example will clarify the meaning and the use of **Table 1**. Assume a loudspeaker with $f_C = 50$ and $Q_C = 1$. The interesting values are on the table's fourth row, then the response presents a 1.249-dB peak at the frequency of 70.711 Hz (i.e., approximately $50 \times \sqrt{2}$) and the 3-dB point at 39.3 Hz (i.e., 50×0.786).

In some cases, the loudspeaker's "natural" resonant frequency is higher than the required value, typically when low-mass and/or low-compliance drivers are used. A small enclosure also contributes to increase f_C. High Q_C are typical with small enclosures ($V_B << V_{AS}$), speakers without powerful enough magnets (small $B \times l$ and thus small Q_{ES}), or in the case of current driving, which is sometimes claimed to present lower distortion in comparison with the more classical voltage driving.

Less common, but still possible, is the case of a loudspeaker with a lower resonant frequency as required that occurs when the loudspeaker is used as satellite in a three-box (i.e., two satellites plus one subwoofer) system.

In all the aforementioned cases, it is possible to equalize the natural loudspeaker response by means of a circuit—the equalizer—inserted between the source output (e.g., the RIAA preamplifier, the CD player, the tuner, etc.) and the amplifier input. Typically, the equalizer is inserted between the preamplifier (or control unit) output and the power amplifier input, or equivalently inside the tape monitor loop.

THE CLASSICAL LINKWITZ EQUALIZER

The equalizer needs to transform the natural loudspeaker response into another one, which still has the form (1), but with different values of the resonant frequency and/or of the quality factor.

I denoted the loudspeaker's parameter and the transfer function with the apex LS (0) as it is (the response I want). By multiplying the equalization function by the loudspeaker transfer function I obtain the desired equalized response. This means the equalization function is the ratio between the required transfer function and the loudspeaker's function. In a formula:

$$T_{EQ}(f) = \frac{T_0(f)}{T_{LS}(f)} = \frac{1 + j\frac{1}{Q_{LS}}\frac{f}{f_{LS}} + \left(j\frac{f}{f_{LS}}\right)^2}{1 + j\frac{1}{Q_0}\frac{f}{f_0} + \left(j\frac{f}{f_0}\right)^2}\left(\frac{f_{LS}}{f_0}\right)^2 \quad (7)$$

Note that the function (7) transforms T_{LS} (f) into T_0(f) in both amplitude and phase, although **Figure 1** and **Figure 2** plot only the amplitude.

Figure 2 shows the response of three hypothetical loudspeakers (thick curves): red (f_C = 80 Hz, Q_C = 1.5), green (f_C = 100 Hz, Q_c = 1.5), blue (f_C = 45 Hz, Q_C = 1/√2). The two thin curves (red and green) are relative to two equalizers, which transform the loudspeaker response of the corresponding color into the blue one.

Figure 3 shows a very classical network that implements the transfer function (7), beside a "–" sign, which corresponds to the polarity inversion in the loudspeaker wires, and can then be easily compensated (if required) by a further inversion of the loudspeaker polarity.

The component values of the network in **Figure 3** can be calculated by choosing a convenient value for the capacitor

Q_C	$\frac{f_{PEAK}}{f_C}$	$10\log_{10}\left(\frac{4Q_C^4}{4Q_C^2-1}\right)$	$\frac{f_{3dB}}{f_C}$
0.5	–	–	1.554
1/√2≈0.707	–	–	1
1	√2≈1.414	1.249	0.786
√2≈1.414	1.155	3.59	0.707
2	1.069	6.301	0.674
3	1.029	9.665	0.657

Table 1: The peak frequency, peak value (in decibels), and 3-dB frequency of a closed-box loudspeaker is shown with different quality factors.

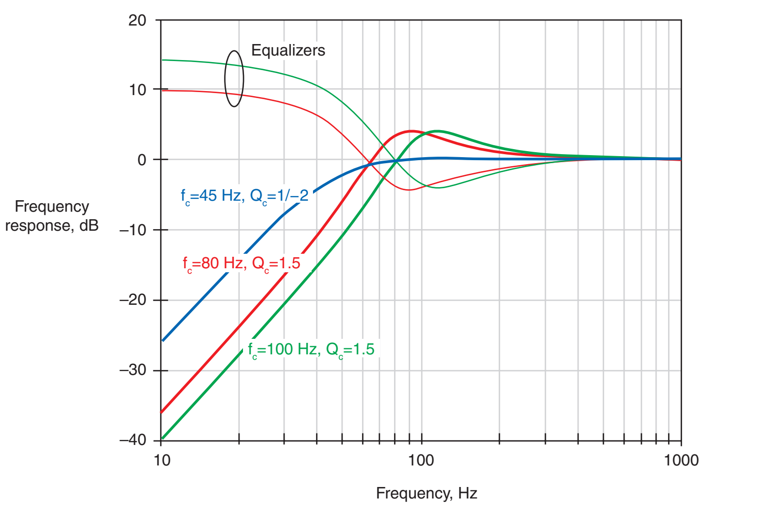

Figure 2: Here is the response of two closed-box loudspeakers. Red (f_C = 80 Hz, Q_C = 1.5) and green (f_C = 100 Hz, Q_C = 1.5) represent typical cases of a loudspeaker small and/or with a too small box. The blue curve (f_C = 45 Hz, Q_C = 1/√2) presents a much better response. Finally, the green curve is the equalization curve to transform the red response into the blue one.

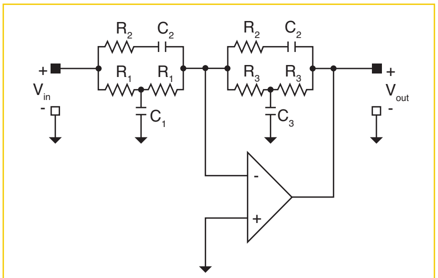

Figure 3: The Linkwitz equalizer implements the transfer function (6).

C_2 and computing the expressions. All the values are in ohm, farad, and hertz:

$$R_1 = \frac{1}{4\pi}\frac{1-\frac{Q_0}{Q_{LS}}\frac{f_0}{f_{LS}}}{f_{LS}^2 - f_0^2}\frac{f_0}{Q_0}\frac{1}{C_2} \quad (8)$$

$$R_2 = \frac{1}{2\pi}\frac{f_0 - 4\pi R_1 C_2 Q_0 f_{LS}^2}{Q_0 f_0^2}\frac{1}{C_2} \quad (9)$$

$$R_3 = \frac{f_{LS}^2}{f_0^2}R_1 \quad (10)$$

$$C_1 = \frac{1}{4\pi^2 R_1^2 C_2^2 f_{LS}^2} \quad (11)$$

$$C_3 = \frac{R_1}{R_2}C_1 \quad (12)$$

R1 and R2 are both positive if, and only if, one of the two following bilateral constraint on the ratio is fulfilled:

$$\frac{f_0}{f_{LS}} < \frac{Q_0}{Q_{LS}} < \frac{f_{LS}}{f_0} \quad (13a)$$

$$\frac{f_{LS}}{f_0} < \frac{Q_0}{Q_{LS}} < \frac{f_0}{f_{LS}} \quad (13b)$$

This means the circuit shown in **Figure 3** is not always realizable. Rather, it requires that natural and equalized parameters fall within the limits (13a) or (13b).

In order to clarify the problem, consider the two cases of **Figure 2**. Choosing (100 nF) the formulas (8–12) provide the values in **Table 2.**

Hence, the "Red case" is not possible to implement with the Linkwitz equalizer, although its parameters are not that much different from the ones of the safe "Green case." Furthermore, the network shown in **Figure 3** needs

	R_1	R_2	R_3	C_1	C_3
Green	5,003 Ω	604 Ω	24,707 Ω	1.012 µF	205 nF
Red	8,506 Ω	–3,749 Ω	26,883 Ω	547 nF	173 nf

Table 2: The "Red case" is not possible to implement with the Linkwitz equalizer, although its parameters are not that much different from the ones of the safe "Green case."

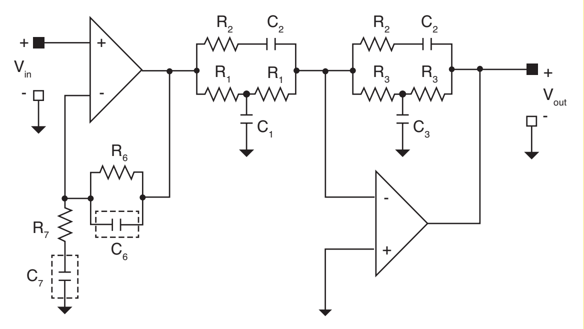

Figure 4: The Linkwitz equalizer's generalized version has two operational equalizers.

an input buffer, because it is sensitive to the driving source's impedance and because its input impedance is typically low and frequency depending in the order of a few hundred ohms, see the value of R2 as obtained from the first (Green) case. In other words, the response of the network in **Figure 3** coincides with the function (7) only considering the voltage effectively applied to its input, and not just the nominal output of the source. The two values are reasonably equal only if a buffer with high input impedance, low output impedance, and flat gain over the frequency is placed between the source and the network of **Figure 3**.

LINKWITZ EQUALIZER'S GENERALIZED VERSION

Figure 4 shows my proposed solution for the two limitations of the Linkwitz equalizer. The circuit has two operational amplifiers instead of one. Its input (output) impedance is ideally infinite (zero), and presents a frequency response virtually independent from the load conditions, unless the current driving requirements do not exceed the capabilities of the used operational amplifiers.

When one of the two conditions (13a) or (13b) is fulfilled, then $R_6 = 0$ Ω (or short-circuited), R_5, C_5, and C_6 are not in place, and R_1, R_2, R_3, C_1, and C_3 are computed with the formulas (8–12). The circuit in **Figure 4** simplifies into the one in **Figure 3**, with the addition of a unit-gain input buffer.

With none between (13a) and (13b), the solution is based on a less demanding requirement on the Linkwitz network, which makes a partial compensation of the response. The remaining part of the compensation has the transfer function:

$$T_{GB}(f) = \frac{1+\frac{j2\pi f}{s_{PA}}}{1+\frac{j2\pi f}{s_{PB}}}\frac{s_{PA}}{s_{PB}} \qquad (14)$$

where the real numbers s_{PA} and s_{PB} are the so-called zero and pole of the function.

The first-order function (14) is much simpler than the second-order one (7). The latter is implemented with the frequency-depending buffer shown in **Figure 4**, realized with the first operational amplifier and the components R_5, R_6, C_5, and C_6. The relative component values can be calculated through the formula listed in **Table 3**.

The remaining components of the network in **Figure 4**, can be computed by assuming a convenient value for C_2, through the expressions:

$$s_{PB} = 2\pi\sqrt{\left[\frac{f_0}{Q_0}-\frac{f_{LS}}{Q_{LS}}\left(\frac{f_0}{f_{LS}}\right)^2\right]\frac{f_0}{Q_0}} \qquad (15)$$

$$s_{PA} = \frac{2\pi f_{LS}}{Q_{LS}} + \left(\frac{f_{LS}}{f_0}\right)^2\left(s_{PB}-\frac{2\pi f_0}{Q_0}\right) \qquad (16)$$

$$C_1 = 4\frac{(2\pi f_{LS})^2}{s_{PA}^2}C_2 \qquad (17)$$

$$R_2 = \left(\frac{1}{Q_{LS}}-\frac{s_{PA}}{2\pi f_{LS}}\right)\frac{1}{2\pi f_{LS}}\frac{1}{C_2} \qquad (18)$$

$$R_1 = \frac{1}{2}\frac{s_{PA}}{(2\pi f_{LS})^2}\frac{1}{C_2} \qquad (19)$$

$$C_1 = 4\frac{(2\pi f_0)^2}{s_{PB}^2}C_2 \qquad (20)$$

$$R_1 = \frac{1}{2}\frac{s_{PB}}{(2\pi f_0)^2}\frac{1}{C_2} \qquad (21)$$

	$s_{PA} > s_{PB}$	$s_{PA} > s_{PB}$
C_6 =		Open circuit
C_7 =	Short circuit	
R_6 =	$\frac{1}{s_{PB}C_6}$	$\left(\frac{1}{s_{PA}}-\frac{1}{s_{PB}}\right)\frac{1}{C_7}$
R_7 =	$\frac{1}{s_{PA}-s_{PB}}\frac{1}{C_6}$	$\frac{1}{s_{PB}C_7}$

Table 3: These are component values for the additional first-order equalizer of Figure 4.

The formulas (15–21) together with the ones in **Table 2** applied at the "Red case" of **Figure 2** (impossible for the standard Linkwitz equalizer) give (C_2 = C_4 = 100 nf) R1 = 3,061 Ω, R2 = 7,141 Ω, R3 = 21,438 Ω, C1 = 4.225 µF, C3 = 272 nF, R4 = 64,653 Ω, R5 = 53,164 Ω, and C5 short-circuited.

KEEP IT SIMPLE

I sought a possible solution for the classical Linkwitz equalizer's well-known limitations. Moreover, I tried to keep the added complexity to a minimum. It seems I was succesful. *aX*

By Richard K. Mains (United States)

Experimenting with Electrostatic Speakers (Part 3)

Electronics to drive a hybrid electrostatic system

In the final installment of this article series, I discuss issues related to the electronics required to drive hybrid electrostatic systems.

ELECTRONICS

Electronics is difficult for many ESL builders, and I suspect it discourages some people from attempting an ESL project. If you are not experienced with electronics and high voltage, it is not necessary to build your own. There are commercial alternatives available. I have included schematics of the circuits I used for the benefit of experienced builders who would like to design and build their own electronics.

The required electronics consist of basically three parts. You need a high-voltage DC power supply to charge the diaphragm, an active crossover to divide the signal between the ESLs and the low-frequency drivers in the system. Finally, you need a way to drive the ESLs. I think the easiest way is to use low-voltage audio amplifiers and step-up audio transformers. In this article, I will discuss all these components except for the low-voltage audio amplifiers.

HIGH-VOLTAGE POWER SUPPLY

I have some words of caution before I present the circuit used for the high-voltage DC power supply. If you have little experience working with high-voltage circuits, I would advise you not to attempt to build a high-voltage supply yourself. Assembled high-voltage supplies are available (e.g., ER Audio, which sells them in assembled and kit form). Also, you can find suggestions about scavenging high-voltage supplies from other products (e.g., copiers) at various places on the Internet. Whichever way you decide to go, if you are unfamiliar with high-voltage circuits please be careful. Under no circumstances should you directly touch any part of the circuit when it is powered. Also, my circuit contains capacitors that can remain charged long after the power to the unit is switched off. The circuit I use is powered directly from 120 V or 240 VAC, and of course, you must not come into contact with this voltage. If you are working with these circuits, it is imperative to have a way to safely measure the output voltage and to discharge the circuit. I use a high-voltage probe, the kind TV service people use to measure the high voltage on a TV picture tube. These probes are typically rated to measure up to 40 kV. The probe's output is an attenuated voltage you can plug into a voltmeter to read. Keep in mind that even though these probes' input impedance is typically 600 MΩ or so, the probe can significantly load down the circuit so the measured voltage may be less than the actual voltage. The high-voltage probe can also be used to discharge the circuit after it is powered off through its input impedance.

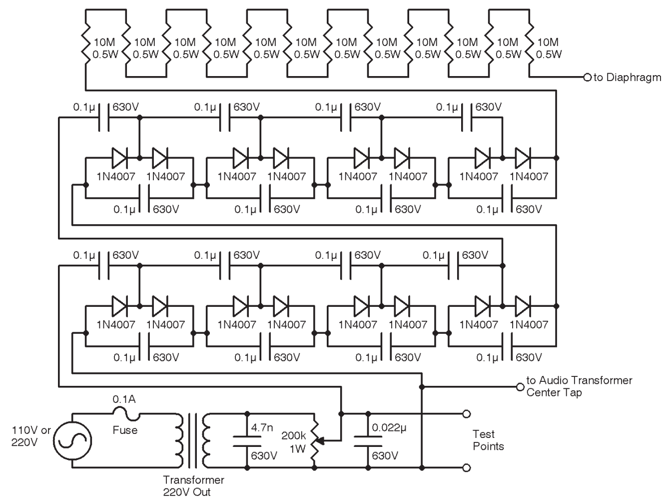

Figure 1: I used this circuit for the high-voltage power supply to charge the diaphragm.

Similar cautions apply to the audio step-up transformers. The transformers typically boost the input voltage by a factor of 50. You should never touch these transformers' output when there is a live source connected to the input. Even if you think there is no voltage at the input, disconnect or turn off the source before coming into contact with the transformer output. And of course, once the electrostatic cells are completed, never touch the stators when the cells are operating. The step-up transformers' output is not current-limited and can be lethal.

Figure 1 shows the circuit I used for the high-voltage power supply to charge the diaphragm. This circuit uses a cascade of voltage multipliers to obtain the required DC voltage. Each multiplier section consists of two 0.1-µF capacitors and two 1N4007 diodes. Each multiplier section develops twice the peak voltage applied at the input to the cascade. First, I isolated the input AC voltage with a power transformer. This is a safety precaution. The 0.022-μF and 4.7-nF capacitors on the transformer's output side may not be necessary, but I added them to reduce the effect of possible power line spikes. The power transformer's output, which I designed to be 220 VAC, is applied across a 200-kΩ potentiometer (pot). The voltage at the input to the voltage multiplier cascade is controlled by adjusting this pot. The maximum output voltage obtainable from the circuit can be determined as follows. The voltage across the pot is 220-V RMS or 311-V peak. If the pot is adjusted for maximum voltage, each voltage multiplier section will produce about 622 V. Since there are eight sections altogether, a 4,980-VDC maximum output voltage is expected. You can adjust the pot to obtain any intermediate voltage value you require for your ESLs. The voltage at the test points may be monitored for the desired input AC voltage to the cascade. (Be sure to turn the pot down so the input voltage will be zero when you first power up!) For my hybrid speaker system, I set the voltage at the test points to be about 96.4 VAC, which corresponds to a 2,180-VDC output voltage.

I put 120 MΩ between the output of the voltage multiplier sections and the diaphragm. This resistance serves two purposes. First, it helps ensure the diaphragm operates in "constant charge" mode, which is desirable because it reduces distortion in electrostatic speakers. But this resistance has the added benefit of limiting the output current from the supply, in case you inadvertently come into contact with it. (If someone touches the circuit on the input side of these resistors there is no current limiting!) You don't need much current from the supply to charge the diaphragm, so it makes sense to limit the output current capability.

If you use the circuit shown in **Figure 1**, enable sufficient clearance between points of high voltage difference in the circuit. For example, you don't want the voltage multiplier sections to come near the input to the circuit or to the Center Tap connection.

ACTIVE CROSSOVER

Figure 2 shows the circuit I used for the active crossover in my hybrid electrostatic speaker system. This would be the circuit for either the left or right channel, so the circuit in **Figure 2** would have to be duplicated for stereo output.

I used NE5532 op-amps throughout. You may decide to use a different op-amp. In **Figure 2**, U1 is a buffer stage that prevents the crossover from loading down the circuit at its input, perhaps some kind of pre-amp. The U2 and U3 stages process the signals for the woofer. U2 is a variable-gain stage used to set the woofer's level by setting the 50-kΩ pot. This adjustment is extremely important in hybrid electrostatic systems. This pot should be a high-quality, multi-turn type. Another option is to have a plug-in resistor at this spot. Whatever the case, you need to have a precise method of controlling the gain since

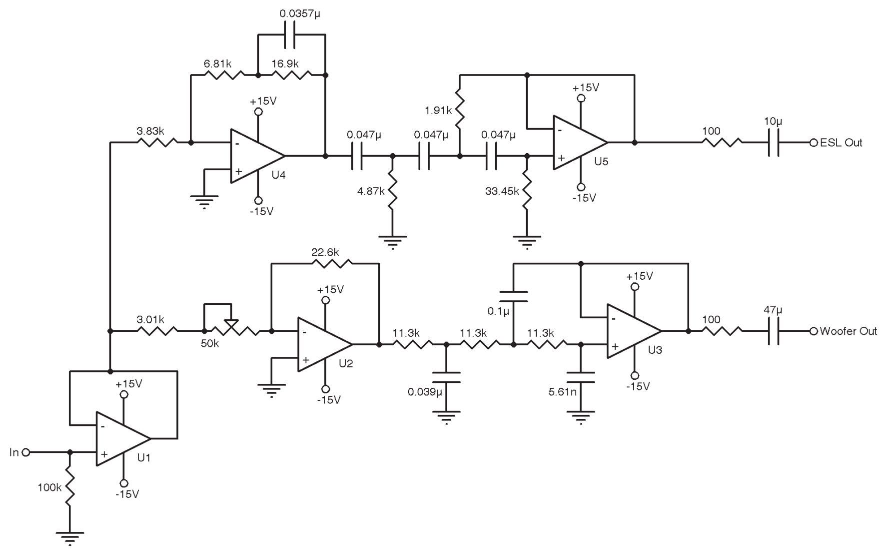

Figure 2: I used an active crossover circuit for my hybrid system.

this is a sensitive adjustment. The part of the circuit involving U3, starting with the 11.3-kΩ series resistors, is a third-order Butterworth low-pass filter with a 500-Hz corner frequency. If your system requires a different crossover frequency, you can frequency or impedance scale the filters' components in **Figure 2**.

U4 and U5 process the electrostatic speaker's signals. U4 implements a low-frequency equalization, which Roger Sanders discusses in "An Electrostatic Speaker System Part 1," *Speaker Builder*, February 1980. The boost's upper corner frequency is 918 Hz, which differs from the value recommended in Sanders' articles. However, I found using FFT spectrum analyzer measurements, this is the optimum equalization for my system.

The U5 stage, beginning with the 0.047-µF series capacitors, implements a third-order Butterworth high-pass filter with a 500-Hz corner frequency that complements the low-pass filter in the woofer circuit. Again, you can scale the filter's components if you need a different crossover frequency.

I also have a relay circuit to protect the speakers from op-amp transients (see **Figure 3**). This circuit would have to be duplicated for stereo output. During power-up, RT1 and CT1 introduce a delay during which transistor TT1 is off and the crossover outputs are connected to ground, enabling sufficient time for the op-amp turn-on transients to complete. When TT1 turns on, the outputs are released from ground. At power down, the −15 V comes up toward ground and turns on transistor TT2, which quickly discharges CT1 and turns off TT1, thereby grounding the outputs before the op-amp turn-off transients begin.

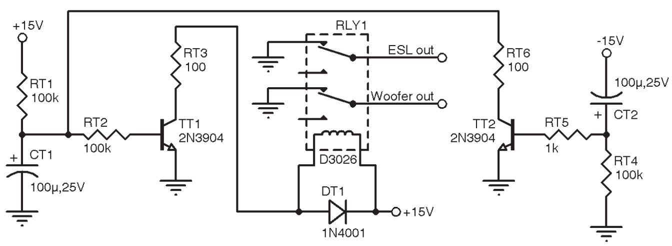

Figure 3: I used a relay protection circuit in the active crossover.

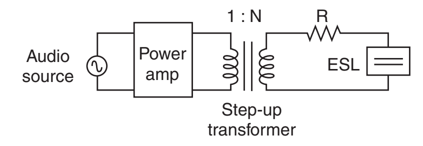

Figure 4: The diagram shows the tests I performed on my audio step-up transformers.

STEP-UP TRANSFORMERS

Many people think of electrostatic speakers as the "perfect" transducer, and certainly a well-made system can be stunning in sound quality. However, I have also found electrostatics are demanding speakers. I have worked on electrostatic speakers off and on for many years, and only recently can I say I am satisfied with my system. Everything has to be just about right: the crossover, the method of driving the speakers, and the speaker positioning, to name a few important factors. Here is one example.

For a long time, the electrostatics I made sounded harsh. Some sounds (e.g., the upper registers of violins) were almost unbearably strident. I could see in my measurements a rising high end, which I initially tried to compensate for in the active crossover circuit. However, it was never enough to eliminate the problem. (The effect I am talking about occurs at a higher frequency than is addressed by the ESL equalization circuit in **Figure 2**.) I finally discovered the difficulty was with the audio step-up transformers.

I had read about transformer resonance in Roger Sanders' articles and elsewhere, but somehow I had not recognized it as a significant problem. What happens is the transformer exhibits some leakage inductance that resonates with the ESL capacitance. In a series-resonant circuit, the voltage across the individual elements (inductor and capacitor) can be larger than the input voltage. I think this effect is not emphasized nearly enough. When I finally started doing measurements on the step-up transformers, I was surprised to see the effect's magnitude.

Figure 4 shows the setup I used to test the transformers. You need an audio generator you can trust to produce constant amplitude sine waves versus frequency. I used a digitally programmable sine wave generator for the audio source. The generator's output is passed through an audio power amplifier. It is a good idea to use the power amp you intend to use in the final system, because some power amps are not flat with frequency when driving capacitive loads. The power amp output goes through the step-up transformer and the transformer output is connected to the ESL, just as in the real system. In **Figure 4**, I have indicated the ESL as a capacitor enclosed in a box, since the ESL looks electrically almost like a capacitor. Note in **Figure 4** there is now the option of a resistor R in series with the ESL. If R = 0, there is the usual situation where the step-up transformer output is connected directly to the ESL stators.

You don't need much voltage from the audio generator to perform this

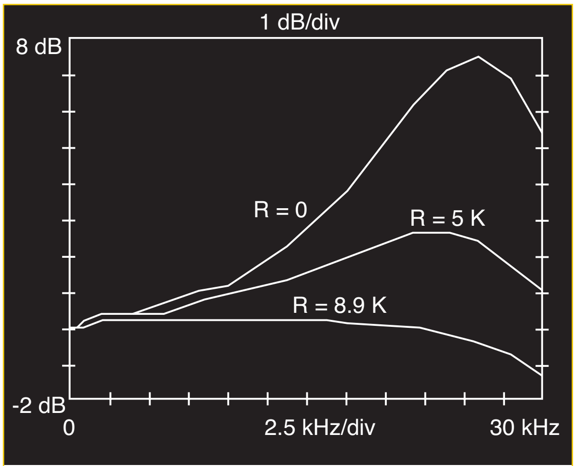

Figure 5: VESL versus frequency for the Triad S-142A for three different values of series resistance, R.

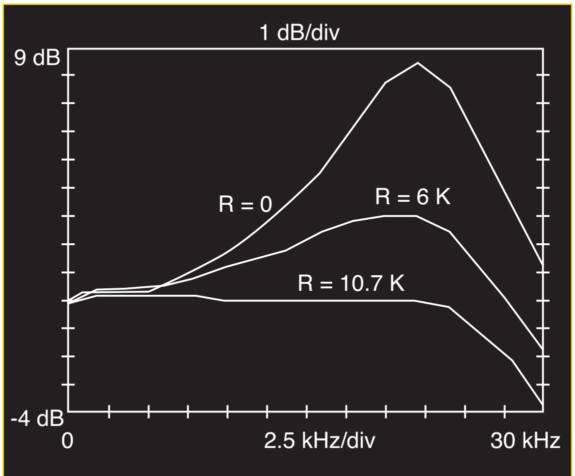

Figure 6: VESL versus frequency for the Hammond 1650E for three different values of series resistance, R.

test. Let's say the power amplifier multiplies its input voltage by 20 and the step-up transformer multiplies its input voltage by 50. With only 10 mV from the generator, there will be about 10 V at the ESL. Be careful not to turn the generator amplitude up so much you damage whatever you are using to measure the voltage at the ESL. (I must re-emphasize, don't touch the transformer output with anything connected to its input).

I used an oscilloscope to measure the voltage VESL versus frequency, because I don't trust my VOM to be accurate over the entire audio frequency range. As a precaution, I powered the scope through an isolation transformer. I currently use two different kinds of step-up transformers. One is the Triad S-142A, which I got because Roger Sanders recommended it. Unfortunately, it is no longer available. The other is the Hammond 1650E transformer, which as far as I know is still available from Antique Electronic Supply (www.tubesandmore.com). It is similar, but not identical, to the Triad S-142A. **Figure 5** shows the magnitude of VESL versus frequency I measured for the Triad S-142A, for three different values of series resistance R. For this test, the Triad's 4-Ω tap was used as the input, and the full winding on the 8,000-Ω side was used as the output.

For many years, I connected my ESLs directly to the Triad step-up transformers, which corresponds to the R = 0 case in **Figure 5**. I think it is clear why my speakers sounded harsh—at 20 kHz, the response is up by 5 dB, and it reaches a maximum of about 7.5 dB at 26 kHz before starting to come down again.

You can almost completely eliminate this effect by putting resistance in series with the ESL. In **Figure 5** you can see the peak is much reduced with R = 5 kΩ, and the response is nearly flat for what I consider to be this transformer's optimum value, R = 8.9 kΩ. Unfortunately, the optimum value of R is somewhat different for each transformer. **Figure 6** shows the result of the same measurement for the Hammond 1650E transformer, for which the optimum resistance turns out to be about 10.7 kΩ.

For the Hammond 1650E, the resonant effect is even more severe than it is for the Triad S 142A. In the R = 0 case, the response is up by 7.7 dB at 20 kHz, and it reaches a maximum of 8.4 dB at 22 kHz.

In addition to flattening the voltage supplied to the ESL stators, this series resistance has an added benefit. Because the ESL looks essentially like a capacitor, it presents a low-impedance load at high frequencies. The addition of this series resistance somewhat reduces the high-frequency load placed on the power amplifier.

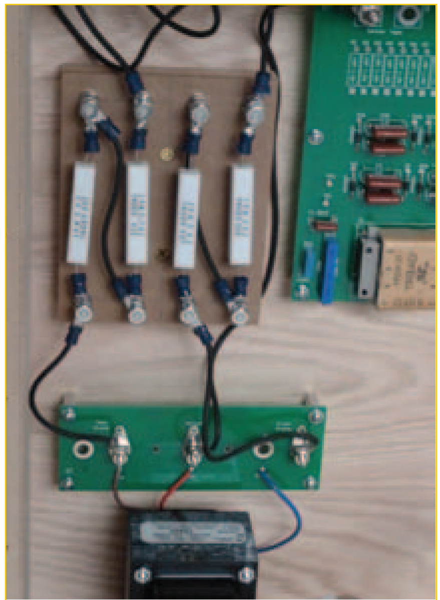

Photo 1: Here is how I mounted the ESL series resistance in the base of my cabinet.

Needless to say, I have modified my ESLs to include this series resistance. I did not originally plan on it when I designed my ESL cabinets, but it was not difficult to modify them to include the series resistance. **Photo 1** shows how I did the modification. I cut a small piece of MDF and used plastic bolts to mount four 10-W resistors. I don't know that it is necessary to divide these resistors evenly in the outputs going to each stator, but I did so to maintain symmetry. The values of the four resistors in **Photo 1** add up to the optimum value of R obtained from **Figure 6**.

ELECTROSTATIC CELLS

The making of electrostatic cells is well within the grasp of amateur speaker builders. If you are considering an electrostatic speaker project, I hope the tips I presented in this article are useful. If I can help with any questions related to electrostatics, e-mail me at: rkm@acoustic-instruments.com. *aX*

REFERENCE

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

 By Richard Honeycutt (United States)

Differences in Amp Sound

What's the Truth?

The tube-versus-solid state debate may have resolution

Back in the 1960s, after crossover distortion was tamed in the better solid-state amplifiers, many serious audiophiles remained convinced that tube amps sounded better than any solid-state ones. Then in the 1970s, after Matti Otala and others had demonstrated the effects, cause, and prevention of slewing-induced distortion, the percentage of serious audiophiles who preferred tube amps may have declined slightly, but it certainly did not drop to zero.

THE GREAT DEBATE

The first tests to determine the cause of the difference in sound, if any, were mostly performed by engineers, who concluded that since the frequency response and distortion performance of the best tube and the best solid-state amplifiers were comparable, there must not be any difference (see **Photo 1**). Thus began the so-called "Great Debate." I believe we are now in a position to put this debate to rest.

First, I must affirm, as mentioned in a previous article, that subjective amplifier judgments are, by nature, individual perceptions. Every perception is a fact to the person perceiving. Thus, if you tell me you can hear a difference between two amplifiers, I have to believe you. If I can hear no difference, then perhaps you are a more critical listener than I am. If neither I nor any of a dozen other skilled listeners can hear a difference, then you have indeed heard a subjective difference, but since the perceived difference is not a shared reality, we cannot say there is an objective difference. If you cannot prove in a properly designed double-blind test that you can repeatedly hear a difference, then we must conclude that the difference you do hear does not proceed from physical, and therefore, determinable, causes.

THE GOLD STANDARD

In last month's article, I discussed blind and double-blind testing. Both assume that the equipment under test is presented in a pair: two amplifiers or two speakers and so forth are being compared. In a blind amplifier test, the test subject (i.e., the person providing opinions about the equipment under test) does not know which amplifier is playing at a given time, in order to avoid the effects of extraneous variables (e.g., manufacturer preferences or finish details). Long ago, experimenters found that test operators (e.g., set-up technicians or other persons involved with the experiment) would sometimes unintentionally provide clues via facial expression or body language that could give away the identity of the equipment under test. Thus, double-blind tests, in which no one involved in the test operation knows which piece of equipment is being used at a given time, were developed. Double-blind tests are the gold standard for any tests involving human perception.

Some people do not like double-blind tests, believing themselves to be immune to hidden biases. However, there is no evidence to prove that even the best of intentions enable a person to avoid subconsciously tilting his/her test responses in a preferred direction. (Perhaps a Vulcan could?)

Resistance to double-blind testing is not limited to those who believe tube amplifiers sound better, although some of those listeners still offer objections to the elaborate test protocols. Some people firmly believe that any well-designed IC-based amplifier sounds better than any hollow-state amplifier and still claim that double-blind testing is unnecessary for them. Perhaps the most convincing for me personally was an individual who told me he made a change that was expected to make an amplifier sound better, but he was surprised to find that it sounded worse. No details of the test protocol were given. Of course, this amounts to hearsay, and thus cannot be considered scientific evidence.

DISTORTION MEASUREMENTS

In 1977, the British magazine *Hi-Fi News and Record Review* published an article by Jean Hiraga detailing sensitive distortion measurements made in Japan on a variety of tube and solid-state amplifiers that operated well below clipping. The article was reprinted by *audioXpress* in the March, 2004 issue. This article showed conclusively that differences in the distortion spectra of excellent amplifiers do exist, but no effort was made to correlate these measured differences with perception.

In the March, 1980 issue of *High Fidelity* magazine, "The Great Ego-Crunchers: Equalized Double-Blind Tests" by Daniel Shanefield was published. This article directly addressed the perception issue. Shanefield mentions the division of audiophiles into "golden ears" who insisted that they could hear differences in amplifiers and "nonbelievers," who constitute the

Photo 1: Tubes or transistors? A listener's preference is a subjective choice.

majority. He makes the statement: "In the next few years (after 1970) several small-circulation magazines espoused the golden-ear point of view, though they often disagreed with each other about which components were truly excellent and even changed their minds drastically from issue to issue." Shanefield's conclusion was that audible differences among good-to-excellent amplifiers do indeed exist, but that if the frequency response of all amplifiers under test was equalized to be flat within 0.25 dB, no perceptible difference remained. He includes the somewhat startling detail that when he compared three Dynaco 400 samples, frequency response differences of a few tenths of a decibel did exist, and the amplifiers did sound different. His experimental protocol ensured that the amplifiers were operated well below clipping.

Shanefield's tests were subsequently replicated by several members of the Boston Audio Society.

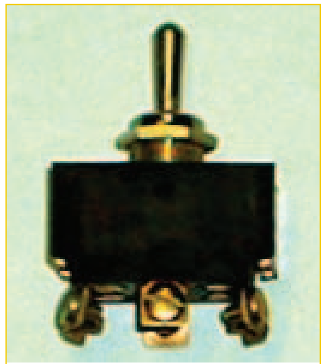

Photo 2: A flick of the switch enables test subjects to switch amplifiers. Sometimes "X" was the same amplifier as "A" or "B."

A/B/X TESTS

In October 1991, David Clark of DLC Design presented the paper, "Ten Years of A/B/X Testing" at the 91st Convention of the Audio Engineering Society. A/B/X amplifier tests compare an unknown amplifier "X" with two known amplifiers, "A" and "B." The test subject's goal is to determine whether "X" is "A" or "B." For example, an excellent amplifier can be used as "A," and a medium-grade amplifier as "B." The test subject can switch at will among "A," "B," and "X," but the switches may be connected so that position "X" is actually amplifier "A" (see **Photo 2**). If the subject can reliably identify "A" is "X," then clearly the difference between "A" and "B" is perceptible to him. If not, we cannot conclude that there are physical causes for the differences that some listeners perceive under less-controlled conditions.

A/B/X tests are usually double blind, but they do not require the equipment's brand/model under test be kept from the listeners, since neither the listener nor the test operator knows which is "A," "B," or "X." Test subjects are not permitted to communicate with each other.

In an A/B/X test, the listener chooses when to flip the switch, allowing whatever amount of time he feels is needed to properly identify the unit under test as "A" or "B." A test may span several listening sessions, if the listener so chooses, or may be finished quickly if the listener is confident he has determined the identity of "A" or "B" as "X" in a short time.

A/B/X testing excels at finding perceptible differences, if any exist, but is not designed to establish levels of accuracy or preferability. The A/B/X test itself was compared with long-term listening as a method of identifying a calibrated 2.5% total harmonic distortion (THD) component that was added to a musical signal. The Audiophile Society acted as the "golden ears." The Southwestern Michigan Woofer and Tweeter Marching Society (SMWTMS) acted as the "engineers." Neither group could identify the distortion at a 5% confidence level in long-term listening tests. However, using A/B/X testing, the SMWTMS not only correctly proved the audibility of the distortion in 45 min. of testing, but also correctly identified a lower amount of distortion. In the complete series of tests, THD was found to be audible at 4% using big-band jazz music, 2% using flute music, and 0.4% using a sine wave. The spectrum of the harmonic distortion was not specified in Clark's AES paper (AES Preprint 3167).

Clark added a note in his 1991 paper, based on a private conversation with Thomlinson Holman (the "TH" of THX). Holman has found that a number of professional power amplifiers do distort audibly when driving highly reactive loads (e.g., some theater speakers) when playing explosive movie sounds. The cause could well be that under such severe load conditions, the amps' power supplies experience instantaneous drops in voltage. This condition would be easy to identify using an oscilloscope.

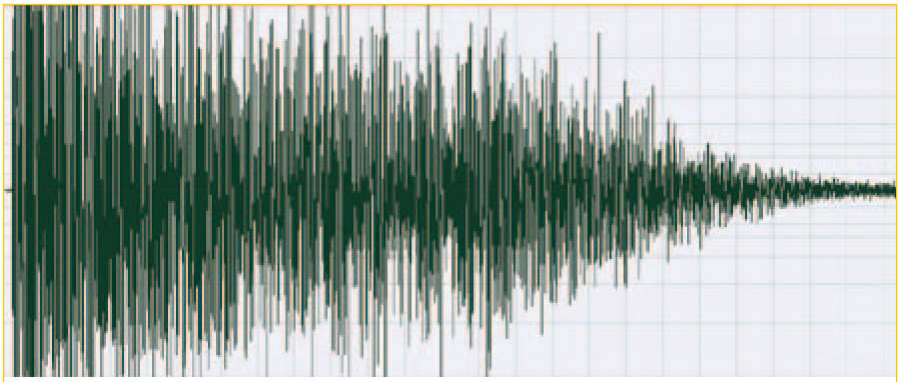

Photo 3: Listeners must often determine for themselves what they hear (e.g., if this sound plotted here is actually clipping on transients).

AUDIO PRO'S CHALLENGE

Audio professional Richard Clark (note: this is a different Clark), originally a believer that different amplifiers sound different, set up a $10,000 challenge: anyone who, by listening only, can identify which of two amplifiers is which, under rules he has established, will receive the prize. The rules can be found at http://tom-morrow-land.com/tests/ampchall/rcrules.htm. They primarily include minimum-quality levels for participating amplifiers, level matching, and so forth, all of which are essential to the validity of any comparative test.

In the years since the challenge was first offered, most large groups have obtained accuracy of 49–51%, which are essentially the results one would expect to occur by chance. Smaller groups have never gotten more than 60% correct. In any statistical sampling, small test groups are more likely to deviate from chance results. As the test population is increased, test results converge toward a specific value. For a random process, a larger test population is likely to converge toward chance results. The fact that Clark found more nearly chance results when measuring larger groups is itself a strong indicator the test subjects' responses were random, not ordered as would be the case if there were perceptible differences between the amplifiers being tested. These are averaged scores for the groups. No individual has ever reached 65% correct. These results do not permit us to say no person can ever hear differences between two good amplifiers, but they do strongly indicate that any such differences must not be very robust (see **Photo 3**).

No test such as David Clark's or Richard Clark's can ever show that there are no perceptible differences in amplifier sound. Proving the nonexistence of anything is philosophically problematic because we can truthfully say only that any experiment did not find such-and-such a thing. However, we cannot perform all possible experiments. As an analogy, we cannot say there are no white crows, because we cannot look everywhere at once. If we postulate the nonexistence of white crows, the person who finds a single example will prove us wrong. So far, however, no person has demonstrated publicly in a scientific fashion that he can reliably distinguish between the sound of two good amplifiers with identical frequency response and low noise driven below clipping. And yet the perception remains that there are real differences in amp sound. As a consultant in acoustics and sound/video system design, I regularly encounter people who assume I must have a vacuum-tube stereo system "because everyone knows they're better."

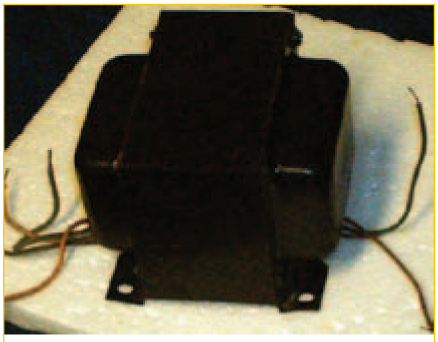

Photo 4: One factor could be whether or not the output transformer affects the sound.

It is true that almost all tube power amplifiers have a very slight high-frequency rolloff (tenths of a dB), but no test has convincingly shown that such a small deviation from flat is perceptible (see **Photo 4**). (If it is, and makes the sound better, should we all add inexpensive RC low-pass filters to our amplifiers to improve the sound so they sound like tube amps?)

THE SEARCH CONTINUES

There are still serious researchers who are trying to find the elusive ingredient to tube sound. In October, 2011, Shengchao Li of Potomac, MD, presented a paper, "Why Tube Amps Have Fat Sound While Solid-State Amplifiers Don't," to the Audio Engineering Society's 131st convention in New York. The paper was reviewed by two qualified anonymous reviewers. Li begins from the assumption that tube amps do indeed sound different. He proceeds to explain that the differences arise from output-tube nonlinearities, amplifier output impedance, and output-transformer nonlinearity resulting from the core material's B-H curve. These nonlinearities, Li says, interact to reduce the low-frequency output of the amplifier under some conditions. The speakers, whose nonlinearity is worst at low frequencies, thus have less signal at those frequencies and thus produce less distortion. In this manner, some low-frequency output is traded for reduced low-frequency distortion. The first two mechanisms suggested by Li have been discussed in earlier Hollow-State columns. The third may be significant in low-feedback amplifier designs, but in more typical Williamson designs, transformer nonlinearities are largely compensated by negative feedback. Certainly all three mechanisms will be exacerbated at levels approaching or exceeding clipping. It would be instructive to see if a low-feedback tube amplifier could be identified in A/B/X testing, and just what level of overdriving is necessary to permit objective identification of any amplifier.

CHOOSING A SIDE

Let us now step bravely into the hornets' nest. After four decades of testing by a number of very capable scientists, no evidence has been published that shows any objective difference among the sound of good-to-excellent audio amplifiers operated well below clipping, if the frequency responses are equalized within 0.25 dB of flat. Does this indicate there is no "tube sound"? It does not, for several reasons.

First, almost no home-music listener (or even recording studio engineer) equalizes the amplifiers flat within 0.25 dB. Virtually no speaker pairs are matched that closely all across the passband. And a flat response is not always everyone's first choice. Hi-fi systems of the 1950s usually had "scratch" and "rumble" filters to remove the sound of artifacts on vinyl recordings. These typically applied a 3-dB/octave high cut above 8 kHz, and low cut below 80 Hz, respectively. Presumably they were included because equipment manufacturers found that their customers wanted and used them: at least under some conditions, listeners did not prefer flat response.

Second, a significant number of audio amplifiers are made for instrument amplification. The distortion used intentionally with electric guitars is well-known. I have also played with professional keyboardists who preferred tube-type Hammond organs because of the "growl" they produce at high volumes. This growl comes from distortion—largely intermodulation distortion—in the tube power amplifiers. A smaller percentage of audio listeners, but still a non-negligible number, prefer some distortion even in their music reproduced from recordings. Naturally, the distortion spectrum is quite important to such people. As shown in a previous column and in Hiraga's 1977 tests, that spectrum is very different for a tube amplifier versus a solid-state amplifier. It is a safe generalization to say that most people prefer tube distortion over the distortion of a solid-state amplifier. Most probably prefer triode tube distortion specifically.

Third, there are conditions under which distortion, even if not desired, occurs in an audio chain. It can happen when a person is listening to music at high levels, especially if low-efficiency speakers and/or underpowered amplifiers are being used. I believe this occurrence is the rule rather than the exception for many serious music listeners. Not only rock music, but also symphonic music, pipe organ music, and big-band jazz require a lot of amplifier power in order to play through typical home-stereo speakers at realistic levels without clipping. Aside from amplifiers for music listening, another place where "tube sound" is much hyped is in microphone preamps for audio recording. In this application, it is not uncommon for the microphone to put out surprisingly high voltages. Capacitor microphone cartridges, especially, are almost immune to clipping, so when exposed to high sound pressures, they can produce output voltages close to 1,000 times their normal output levels. Under these conditions, preamp clipping is pretty much inevitable. Again, the distortion spectrum is very important, and most recording engineers prefer the spectrum provided by tube pre-amps.

So in this columnist's codgerly opinion, there is indeed a "tube sound" under some conditions, and many excellent technicians, engineers, and audiophiles find it preferable for specific applications. It is not, however, scientifically accurate to claim that hollow-state amplifiers are better or worse than—or even perceptibly different from—solid-state ones for all applications.

Purely audio considerations aside, many audiophiles prefer the aesthetics of a "warm," artistically designed (perhaps handcrafted) amplifier over the usual "high-tech" appearance of most solid-state amplifiers. And many of us enjoy the "legacy" feel of the equipment setup when operating vacuum tubes are visible. These are valid reasons to buy hollow-state equipment, if it appeals to you.

TEST FOR YOURSELF

If you are still not satisfied about the topic of "tube sound," you are invited to take part in an online test. The test has two parts. In the first part, you will be invited to use quality headphones (please, not computer speakers) to listen to a pair of .wav files of a short clip played by the lead guitarist of Darrell Harwood and the Coolwater Band. One file was made by recording directly to digital from the guitar, then playing the recording through a solid-state power amplifier adjusted to produce an acceptable amount of distortion for proper artistic effect. The other file uses the same digital recording, played through a tube amplifier adjusted in the same way. Both recordings were made using the same speaker and cabinet, and were recorded using a type 1 calibrated measurement microphone with a tensioned stainless steel diaphragm. The same physical setup was used for both recordings, and the levels of the recordings were carefully matched. You may play the files as many times as you wish, and will then be requested to send an e-mail stating which file, "A" or "B," sounded better to you. If you choose to include your opinion as to which is the tube-amp and which is solid-state recording, please do that as well. The purpose of this part of the test is to illustrate the sound of the different distortion spectra of tube versus solid-state amplifiers. The second part of the test consists of three .wav recordings of a brief clip of classical music. There is a reference clip made directly from the CD, a clip of the selection being played through a McIntosh hollow state power amplifier, and a clip of the selection being played through a commercial solid-state power amplifier. Both of the "amplifier" recordings were made using an Evenstar Pro Peregrin studio monitor speaker, recorded using the measurement microphone described above, with the same physical setup. No equalization was applied to either amplifier, and the amplifiers were operated well below clipping. The reference ("X") straight-through recording is identified, and you are asked to listen to the "A" and "B" recordings, determine which sounds more like the reference, and e-mail your results. The test can be found online at www.edc-sound.com/amptest. It will be available until the end of 2012. Instructions for participating are included on the website. *aX*