Audio Designer Details His Experiences, p. 29

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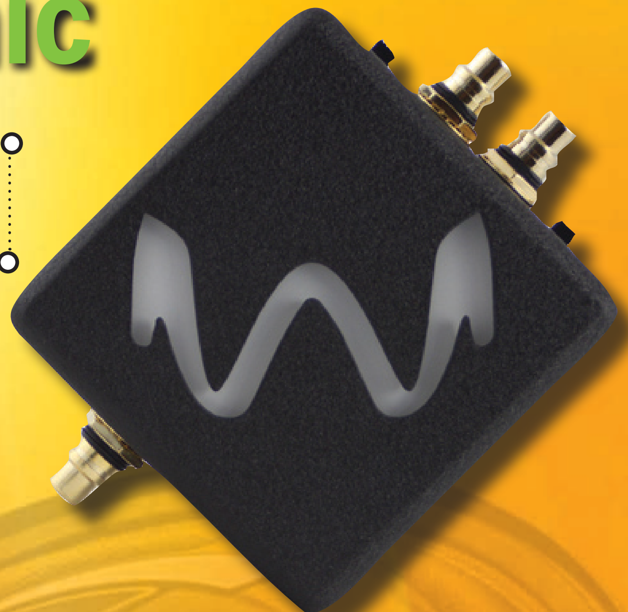

Small Driver Put to the Test

Tweeter Styles Compared

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From the Editor's Desk

Audio Update

We're now a month into the new year. What types of audio projects have you designed and installed? We're interested, so send us an e-mail with "audio update" in the subject line and include details about your latest project. We're interested in tube projects, DIY speaker setups, hi-fi product reviews, and more.

If you haven't decided on your first 2013 project, you'll likely get some excellent ideas from this issue. We feature audio product reviews, speaker design tips, and more. Let's look at the details.

On page 8, Mike Klasco and Steve Tatarunis deliver the second article in their series, "Tweeter Talk." They present different tweeter styles and cover the balanced drive tweeter.

Turn to page 12 for Gary Galo's review of Wyred 4 Sound's µDAC USB D/A Converter. He covers its features and sound.

The third "session" in Richard Honeycutt's "Speaker Design School" series begins on page 16. He details "the nuts and bolts of speaker design."

Even if you attended CEDIA 2012, check out David J. Weinberg's thorough review of the expo on page 20. He presents information about the innovative sound processing, speaker, headphone, and technologies that were on display.

If you're an electric guitar player, you'll enjoy Chuck Hansen's review of Helmuth Lemme's book, *Electric Guitar: Sound Secrets and Technology* (Elektor 2012).

On page 29 you'll find an interview with loudspeaker expert Charlie Hughes. He highlights his audio interests and describes some of his past and present projects.

The third article in Richard Honeycutt's series "Power Supplies for Hollow-State Equipment" appears on page 34. He covers power, tube selection, and more.

Vance Dickason wraps up the article portion of the issue with a look at the Motus UH25CTi. You'll find it's an interesting high-frequency dome tweeter.

Regards,

C. J. Abate
editor@audioxpress.com

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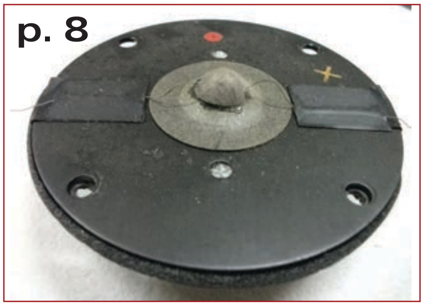

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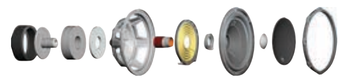
By Mike Klasco and Steve Tatarunis (United States)

Tweeter Talk (Part 2)



Dome Variations and a Balanced Drive Tweeter

Tweeter styles vary, but they all are designed to enhance music quality

In our speaker parts articles, we have covered many loudspeaker components, including the enclosure. But until now, we have omitted a direct exploration of the tweeter. Tweeters are a class of transducer (speaker) designed to reproduce the upper range of audio frequencies. Last month, in Part 1 of "Tweeter Talk," we covered a bit of tweeter history, discussed the cone tweeter, and introduced the common dome tweeter. This month we will explore dome tweeter variations and introduce the balanced drive tweeter.

DOME TWEETER DISPERSION

Dome tweeters generally have better dispersion than cone tweeters, but many factors come into play. Even dome tweeters have many variations including: soft and hard domes, back chambers, phase plugs, voice coil attachment (at the peripheral or modally driven), and magnetic structure topology.

First, let's explore the speaker diaphragms' "effective radiating area" and how this consideration impacts the dispersion of tweeter cones and domes. Simply measuring the tweeter diaphragm's diameter often does not directly correlate to the transducer's dispersion. When a tweeter's diaphragm works as a perfectly rigid piston, the theoretical would more closely track the reality. But, this is not always the case.

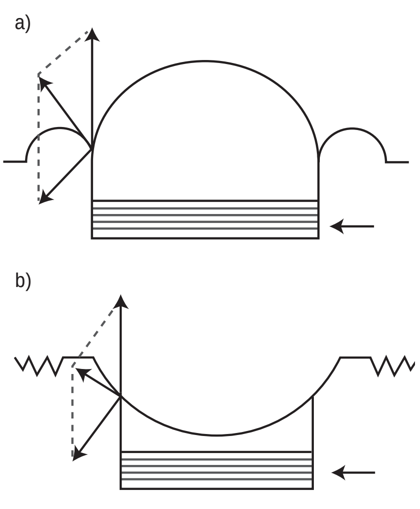


Figure 1: The topology of the voice coil-to-dome attach can be changed, depending on whether it is a standard dome (a) or an inverted dome (b).

The designer can intentionally try to stiffen the diaphragm assembly to increase its piston characteristics or intentionally design for the decoupling and flexing of the diaphragm. Soft dome tweeters are designed to blow with the breeze, while metal dome tweeters stand tall and stiff in the storm.

In the case of hard domes, there many ways to stiffen the diaphragm. The diameter can be reduced, but this complicates many other design aspects. In general, smaller dome tweeters provide wider sound dispersion at the highest frequencies. However, smaller dome tweeters have less radiating area, which limits their output at the lower end of their range. They also have smaller voice coils, which limit their overall power output.

Another way to increase stiffness is to have the diaphragm with a deeper contour (e.g., a deeper dome or cone). As with anything else, too much will get you into trouble. A higher Young's modulus (stiffness) of diaphragm material can be selected. It is possible to switch from a polyester like Mylar (a molten polyethylene terephthalate or PET) to a higher stiffness polyester like Teonex (a polyethylene naftenate or PEN). In the case of hard domes, you can switch from aluminum to titanium.

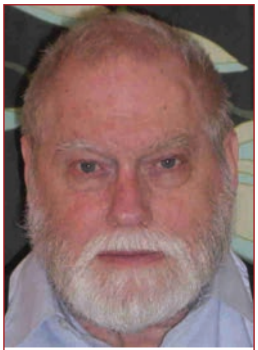


Photo 1: Winslow Burhoe invented the inverted dome tweeter. (Photo courtesy of Burhoe)

The topology of the voice coil-to-dome attachment can also be changed (e.g., a modally driven voice coil). Many inverted dome tweeters have used modal drive, which is where the voice coil is attached at the diaphragm's half diameter. By driving the dome where its first break-up mode is physically located (halfway up the diaphragm), you can shift the break-up frequency up an octave. This is especially practical for inverted dome tweeters (see **Figure 1**).

INVERTED DOME TWEETERS

Winslow Burhoe invented the inverted dome tweeter (see **Photo 1**). In this design, the dome is suspended at its edge but driven by a smaller diameter voice coil (see **Photo 2**). Regular dome tweeters are suspended at the edge and have the voice coil attached at the edge. Burhoe introduced the inverted dome tweeter when he launched Epicure (also known as EPI) in 1970. Epicure designed and sold many successful designs. In the 1990s it became one of the Harman brands. I had the privilege of meeting and talking with Burhoe, an industry icon, at an informal Association of Loudspeaker Manufacturing and Acoustics (ALMA) meeting in the Boston, MA, area last summer. He is still designing and building audiophile speakers under the Direct Acoustics brand name. In addition to Epicure, other brands have used inverted domes with modal drives, including Genesis Physics in the 1980s and later Focal's famous beryllium inverted dome tweeters that are still highly regarded today.

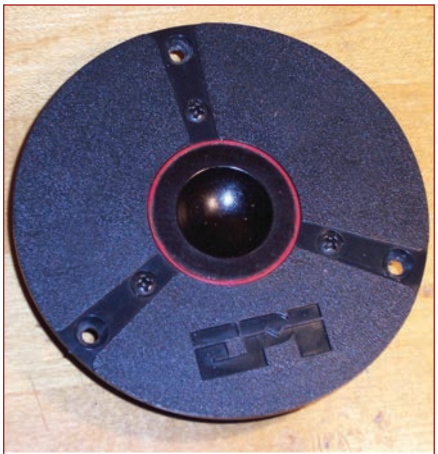


Photo 2: The EPI tweeter is suspended at its edge but driven by a smaller-diameter voice coil.

Jacques Mahul, founder of JMLab and Focal, made the following points during an interview with Jonathan Scull, in the April 1998 issue of *Stereophile*: With an inverted dome, you have the coil in the middle of the dome diameter. The mechanical coupling is better. Consider that the coil of a [positive] dome tweeter is attached tangentially—it's outside, on the circumference. As a result, you have no angle created between the coil and the "base" of the hemisphere—they lie in the same plane. In this way, you actually lose a significant amount of energy.

With the inverted dome (i.e., the coil in the center) there is an angle created between the dome and the coil. The voice coil is fixed at mid-height on the dome and uniformly moves the cone's entire surface. The energy transfer is more efficient, so you lose only perhaps 45% or 50% of the energy between the coil and the dome. This is important, as it helps bring out all the micro-information and inner detail contained in the musical source. Sometimes the tweeter's size does not reflect the dispersion.


Photo 3: The Allison tweeter has a larger diaphragm with a small effective radiating area.

CURLINEAR DECOUPLING

One particularly well-engineered approach is to have a larger diaphragm with a small, effective radiating area (e.g., the innovative Allison tweeter, which is shown in **Photo 3**). In an interview with Steven Mowry Roy Allison said, "The midrange and tweeter units are not really domes in the usual sense. They are convex radiators, but each is driven by a voice coil at approximately half the distance from the center to the suspension edge. Neither has a spider. The tweeter's cone part, although convex overall, is curved inward and the outer edge is clamped through a thin ring of latex foam to the mounting plate. Thus, the cone's radius of curvature changes as the voice coil moves, simulating a pulsating hemisphere that puts large amounts of high-frequency energy into the reverberant field all the way up to 20 kHz. The midrange unit has a straight-sided cone with a flexible polyethylene edge suspension. It has extremely wide and uniform dispersion over its operating range, as does the tweeter. Neither is proprietary inasmuch as they're not protected by patents, but they are difficult and time-consuming to make properly, and they do project relatively far out from the front cabinet panel, which complicates the grille design. They have not been imitated, probably for those reasons." ("*Voice Coil* Chats with Mr. Roy

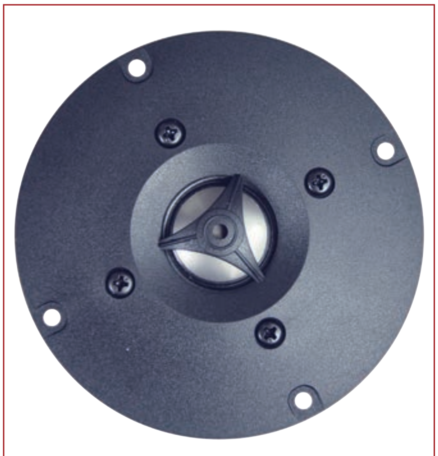


Photo 4: Hard-dome tweeters often contain a phase plug. (Photo courtesy of Electrovision)

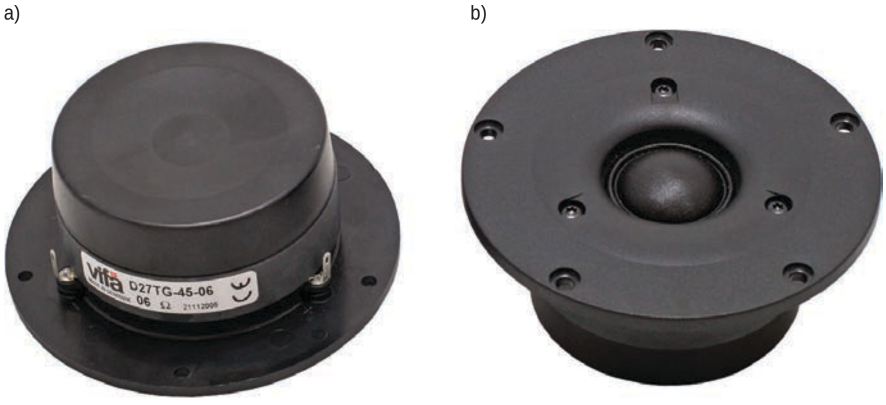


Photo 5: A tweeter's back chamber typically consists of the volume behind the dome (a) and the magnetic structure's inner cavities (b). (Photo courtesy of Vifa Professional Audio)

F. Allison," December 2010).

The idea is the flexing (squeezing) of the outer part of the diaphragm provides the decoupling reducing the effective radiating area, which provides improved power response in the driver's upper range.

PHASE PLUG

The phase plug is a small orifice suspended over the tweeter dome often found in hard-dome tweeters (see **Photo 4**). The wavelengths of the last octave or so of the top-end response are smaller than the diaphragm (if you optimistically assume piston motion). So, the phase plug's purpose is to equalize the path length to the sound energy's listener from the diaphragm's center and sides. For example, in compression drivers, the phase plug optimizes phase coherence and extends (peaks) the frequency response at the treble's very top end.

Photo 6: Balanced drive tweeters are popular in auto-sound coaxial speakers. (Photo courtesy of Clarion)

BACK CHAMBERS

Except for dipole tweeters (e.g., some ribbons and electrostatics) most tweeters have a rear chamber (i.e., captive air volume behind the diaphragm). Typically, it consists of the volume behind the dome and the magnetic structure's inner cavities. Sometimes felt or other acoustically absorbent material is placed behind the dome to absorb reflections from the magnetic structure bouncing back through the dome (see **Photo 5**).

In tweeter designs that target a lower resonance frequency, the pole piece may be vented through to a cup attached to the rear of the magnetic structure. These loosely coupled chambers will lower the resonance frequency, hopefully maintain smooth phase response a bit lower into the midrange frequencies, and enable the tweeter to have a lower crossover point. Yet, back chambers also tend to impact the rest of the tweeter's response so the designer's skill and experience will come into play when the prototype is first tested.

BALANCED DRIVE

Balanced drive is essentially derived from a combination of the common microspeaker, the cone tweeter, and the headphone driver, which is sometimes described as the donut with a dome. Basically, the diaphragm has a center dome with the voice coil directly behind it and a cone shape that joins the diaphragm to the frame. Usually, the cone acts as part of the radiator and part of the suspension. Early versions tended to have a separate dust cap and some of them had spiders. Most current designs have polyester film diaphragms without spiders, often using ferrofluid in the gap for centering the coil. The diaphragm size (typically 1") and small voice coil diameter (often 12-mm diameter or approximately 0.5") result in high output, good efficiency, and low cost. Audax (France) and Foster (Japan) were early evangelists of this design, followed by many Taiwanese and, later, Chinese vendors. Balanced drive tweeters are also popular in auto-sound coaxial speakers as the design is robust and can be easily paired with small woofers (see **Photo 6**).

Next month we'll continue with more Tweeter Talk—discussing esoteric tweeters including air-motion transformers, electrostatic, polyvinylidene difluoride (PVDF), and piezoelectric polymer designs. *aX*

 By Gary Galo (United States)

Wyred 4 Sound µDAC USB D/A Converter

Small footprint USB DAC offers fine performance

Wyred 4 Sound µDAC
Wyred 4 Sound 4235 Traffic Way Atascadero, CA 93422 (805) 466-9973 www.wyred4sound.com support@wyred4sound.com
Price: $399

Wyred 4 Sound is a small California-based audio company with an interesting history. Beginning in the early 1990s, owner and designer E. J. Sarmento worked for the now-defunct Cullen Circuits. (It ceased operations in the fall of 2010.) That company's founder, Rick Cullen, was PS Audio's production manager in the 1980s, and he had a hand in the design of its original Digital Link D/A converter. When Cullen left PS Audio to form his own company, he remained closely tied to his former employer. Cullen Circuits served as PS Audio's design consultant, and his company was also involved with the manufacturing of some of its products. Cullen Circuits earned something of a cult following among audiophiles with its various modifications to PS Audio's Digital Link III D/A converter, done with PS Audio's blessing.

THE DAC

Wyred 4 Sound originally began as an "after hours" endeavor for Sarmento, but he eventually left Cullen to devote all his efforts to his own company. (A more detailed company history is posted on its website). Wyred 4 Sound has gained considerable respect in high-end audio circles for its DAC2 D/A converter, which uses ESS's flagship Sabre32-series ES9018 DAC chip, and for its power amplifiers based on high-efficiency ICE technology (www.icepower.bang-olufsen.com).

Photo 1: Wyred 4 Sound's µDAC features a S/PDIF coax, Toslink optical, and USB inputs on one end, plus analog outputs, input switching, and a power switch on the other. Measuring about 3.5" square, this diminutive product offers remarkably fine sonic performance for its size and price. The manufacturer's logo is backlit in soft blue when a digital input signal has been locked. (Photo courtesy of Wyred 4 Sound)

The µDAC, or "microDAC," is Wyred 4 Sound's latest digital audio product, and addresses the increasing popularity of music playback via streaming and digital music servers (see **Photo 1**). This diminutive product measures 3.5" square and a little over 1" high. The DAC includes both S/PDIF coaxial, Toslink optical, and USB inputs. The S/PDIF and Toslink inputs will accept datastreams up to 192 kHz/24 bit, and the USB input operates at sampling rates up to 96 kHz.

The µDAC has two PC boards (see **Photo 2**). The main board contains the digital input connectors and all the digital circuitry. The piggyback board, which plugs into the main board with the component side down, contains the analog output circuit, output muting, and the analog RCA output connectors. The RCA connectors used for the S/PDIF input and the analog outputs are audiophile-quality, gold-plated, and Teflon-insulated, which is unusual in such an affordable audio product. The S/PDIF coaxial input is transformer-coupled with a 65612 pulse transformer bearing Wyred's logo. I believe it is actually manufactured by Pulse Corp. It manufactures a transformer with this same part number designed to be used with Cirrus Logic's input receivers. The receiver for the S/PDIF and Toslink inputs is Cirrus Logic's industry-standard CS-8416. Wyred 4 Sound uses its own decoder chip for the USB input.

At the µDAC's heart is an ESS Sabre-series ES9023 DAC chip. The Sabre series is an excellent example of trickle-down

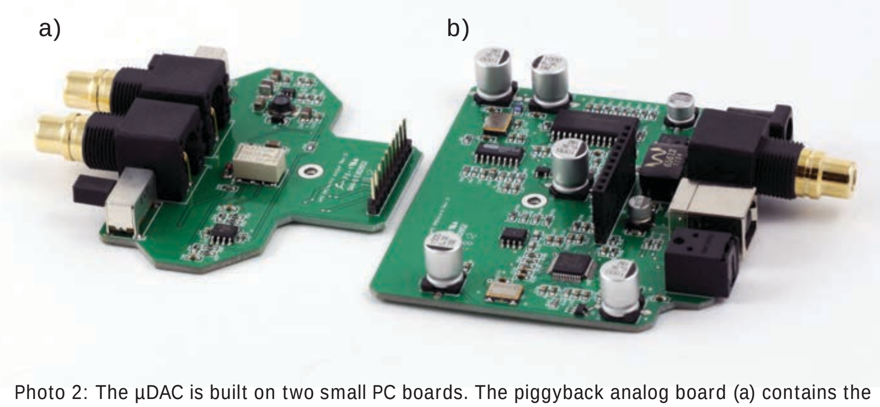


Photo 2: The µDAC is built on two small PC boards. The piggyback analog board (a) contains the LM4562 output op-amp and output connectors, along with the input selector and power switches. The main board (b) houses the input connectors and all the digital circuitry, including the ES9023 DAC chip. (Photo courtesy of Wyred 4 Sound)

technology. This chip inherits several performance features from the state-of-the-art Sabre32 chips, including the Hyperstream architecture and the Time Domain Jitter Eliminator, discussed in my Oppo BDP-95 Universal Player review (*audioXpress*, January 2012). In the ES9023, the Hyperstream architecture operates at 24 bits of resolution, whereas the flagship Sabre32 chips are 32 bit. Unlike the ES9018 used in the Oppo player, the ES9023 isn't multichannel and it isn't compatible with the Direct Stream Digital system used for SACDs. The ES9023 is a stereo chip that decodes PCM datastreams at sampling rates up to 192 kHz/24 bit. The dynamic range is 112 dB and the chip will operate from a single 3.3-V power supply (see **Table 1**).

The ES9023 is manufactured with an integrated op-amp driver that can supply 2 V_{RMS} at the left and right analog outputs. But, Wyred 4 Sound wanted greater output voltage and current capability, so it added a National Semiconductor LM4562 dual op-amp at the DAC output. The µDAC is rated for a 2.5 V_{RMS} maximum output voltage. The LM4562 will deliver 26 mA into a 600-Ω load. (Strangely, ESS doesn't specify the DAC chip's output current capability in the ES9023 datasheet.) The ES9023's outputs are designed to remain at ground reference, as far as DC is concerned, and the typical LM4562 input offset voltage is 0.1 mV. Since the op-amp is fed from the DAC chip's stable, low-impedance output, the entire audio signal path is free of coupling capacitors. I measured an extremely low—and safe—2.2 mV of offset at the output of each µDAC channel, and the output muting circuit ensures there are no turn-on spikes.

Maximum output level	2.5 V
Frequency response	20 Hz to 20 kHz, ±0.1 dB
Signal-to-noise ratio	–102 dB
Noise (A weighted)	13 µV
Output impedance	50 Ω
THD + Noise (A weighted)	0.0035%
Channel tracking	±0.05 dB
Dimensions	3.5" × 3.5" × 1.25"

Table 1: The manufacturer's specifications are listed for the µDAC.

The µDAC is supplied with a "wall-wart" style external switching-mode AC/DC adaptor, which supplies 9 VDC at 100 mA. The unit can also be powered from a computer's USB port, which will be especially useful when the µDAC is being used with a laptop. If a 120-VAC line is available; however, I'd recommend using the supplied AC adapter. The AC adapter's switching supply may actually offer lower noise levels than the USB supplies on many computers. Both the power supply source and the desired digital input are selected with small slide switches located on the case's output end. Wyred 4 Sound notes there are three internal power supply regulation points and multiple filtering points for each regulator.

INSTALLATION

The µDAC is extremely simple to install and operate. The owner's manual is included on the driver CD as a PDF file, and it can also be downloaded from the manufacturer's website. The manual contains easy, step-by-step instructions for installing the Windows USB driver, including screenshots of the various windows that will appear during the process. Drivers are supplied for Windows XP, as well as both the 32- and 64-bit versions of Microsoft's post-XP operating systems (OSes). The driver disc is not needed for installation on a Macintosh. The OS automatically installs the driver when the µDAC is connected. The manual notes the USB device currently does not support Mac OS 10.5 or earlier. A 6' USB cable is supplied with the µDAC. Connection to CD players or other S/PDIF sources is similar to any other outboard DAC, just connect a coax or Toslink interconnect. Be sure to select the correct input and power source using the small selector switches on the end of the unit. When the unit has locked onto a digital source, the Wyred 4 Sound logo, cut into the top of the chassis, is backlit in a soft blue.

THE SOUND

I gave the µDAC the acid test, using it as an outboard DAC in my main stereo system, fed from both my reference Oppo BDP-95 player, and my older NAD M55, both with a coaxial S/PDIF connection. I would not expect a product of this size and price to compete with expensive, high-end DACs and players, but Wyred 4 Sound has achieved a remarkable performance level from the µDAC. The µDAC is a warm and smooth-sounding DAC with a surprisingly fine level of detail and soundstage capability. Deviations from *ne plus ultra* performance are those of omission, and the sonic presentation is refreshingly free of the digital harshness and grain often found in inexpensive digital audio products. The µDAC delivers musical, fatigue-free sound that's good for the long haul.

THE RESULTS

The µDAC has a number of competitors in the same price range, including the NuForce Icon HDP that I reviewed in the April 2011 issue of *audioXpress*. I was very impressed with that product at the time, and I am still extremely pleased with its performance as the main DAC/preamp in my home digital editing system. Two minor weaknesses in the Icon HDP's performance that I noted in the review were "some roughness and graininess in the upper midrange and treble region" and the "lack of ultimate smoothness and detail in the treble." I find that the µDAC improves on the Icon HDP in these areas. The upper midrange and treble are smoother and more detailed than the Icon HDP. In fairness, however, the Icon HDP incorporates a headphone amplifier and a preamp complete with a volume control, making it a more versatile product.

The µDAC offers a worthwhile improvement over the sound of many stand-alone digital players. It provides warmer, more detailed sound and a smoother treble region than Oppo's affordable BDP-93 (reviewed in *audioXpress*, June 2011). The ES9023 may offer performance advantages over the Cirrus Logic CS4362A used in the BDP-93, particularly the Time Domain Jitter Eliminator used in the ESS chip. One thing is certain. The National 4562 op-amp with the DC-coupled topology used in the µDAC are superior to the 5532 op-amps and electrolytic coupling capacitors used in the BDP-93, by several orders of magnitude.

It's not entirely surprising that a $399 outboard DAC would improve the sound of a $500 stand-alone player. But, I was surprised to find that it even acquitted itself admirably when compared to my NAD M55 used as a stand-alone player. The M55, recently discontinued, sold for $2,000 during its five-year lifespan, and was considered one of the best digital players in its price class. The M55 has a Burr-Brown/TI DSD1608 DAC chip along with OPA2134 op-amps in the analog output stage. The DSD1608 has a dynamic range of 108 dB compared to 112 dB for the ES9023, but the improved immunity to jitter may be the ES9023's greatest advantage. The OPA2134 used by NAD is certainly a good audio chip, but it has never been one of my favorite op-amps sonically, despite its excellent measured performance. Though the OPA2134 and LM4562 both have 20-V/µs slew rates, the LM4562 boasts a gain-bandwidth product of 55 MHz compared to 8 MHz for the OPA2134. I wouldn't say I preferred the M55 when used with the µDAC, but the µDAC held its own when replacing the M55's internal DAC and audio circuitry.

Like most audiophiles, I loathe internal computer sound cards. Most of them offer decidedly unrefined sound, lacking in detail, with harsh treble and limited soundstage reproduction. The µDAC is a serious antidote for the typical computer sound card, making it ideal for interfacing a computer with a high-performance audio system.

Wyred 4 Sound may want to consider offering a version of the µDAC with a volume control. This would make the unit more versatile, and it could probably be accomplished by simply inserting a high-quality stereo potentiometer between the ES9023's analog outputs and the LM4562 output op-amp. In the meantime, the µDAC raises the bar for small, affordable outboard DACs. It's a fine-sounding product at a very fair price. *aX*

By Richard Honeycutt (United States)

Speaker Design School

Session 1C: Meet the Speakers

Prepping for the speaker design process

In an effort to help audiophiles achieve the ultimate goal—designing and building their own audio equipment—the third article in my "Speaker Design School" series addresses the nuts and bolts of speaker design.

REQUIREMENTS & APPLICATIONS

The first design step is to set your target specifications. But before actually choosing our targets, let's remind ourselves of a few human perception issues, since most of us will be designing systems for human listeners. No one has unlimited resources, so it makes sense to focus our resources on the most important areas. First, frequency response. Many loudspeaker system design specifications are designed toward marketing rather than user needs. This is an industry fact, and not necessarily a bad thing. But for those who have the luxury of writing specifications based on actual needs, three facts stand out. The ear/brain system is not particularly sensitive to response anomalies at frequencies below about 150 Hz. Absent bass is certainly objectionable, but presence of a 2-dB ripple in the bass response is not generally detectable, even by very well trained listeners, on a double-blind A-B test. Humps wider than one-third octave begin to be a problem at about 2 dB, but even fairly deep dips in response below 150 Hz are unlikely to be noticed. Most of the bass notes that live at those frequencies are recognized mainly by their harmonics, which occur largely above 150 Hz.

The ear/brain system—designed, as it is, primarily for speech communication—is exquisitely sensitive in the midrange or voice range: between 150 and 5,000 Hz. In that range, 1-dB humps in response are audible, if they are one-third octave wide or more. Narrower humps must be much higher before they will be perceptible. Dips also must be one-third octave or wider to be noticeable unless they are very deep.

And now to high frequencies. The BBC performed a carefully designed series of tests in the 1960s in which it determined that only a small percentage of trained listeners could tell whether a music-reproduction system was low-pass-filtered at 12 kHz or had a 20-kHz upper limit. Yet, some recent tests indicate humans can perceive acoustical vibrations up to at least 50 kHz; some say 85 to 100 kHz. Certainly some musical instruments produce harmonics in this range. However, the perception seems to occur via facial cilia, and so cannot be absolutely proclaimed to be "hearing." It is a matter of debate whether reproduction of frequencies beyond 20 kHz adds to the listening experience. At least one speaker manufacturer believes it does. But one thing on which there is agreement is that sensitivity to response flatness is reduced at frequencies above 5 kHz. As in the low end, humps less than 2-dB high are generally not perceptible, and one-third-octave humps must be progressively higher to be audible as the frequency is increased. As in other ranges, dips are much less perceptible than humps.

Figure 1 shows a sample frequency response with deviations from "flat," along with comments on how these deviations would be perceived by listeners.

Next, I need to discuss distortion. After decades of semi-fruitless arguing about distortion, there is some useful information. In this discussion, I will limit the focus to nonlinear distortion. So-called frequency distortion (changes in signal waveforms caused by uneven frequency response) has already been discussed in a previous article. Basically, distortion in a musical program is equivalent to wrong notes. Some of the wrong notes are in tune: these represent harmonics of the desired notes. Others are out

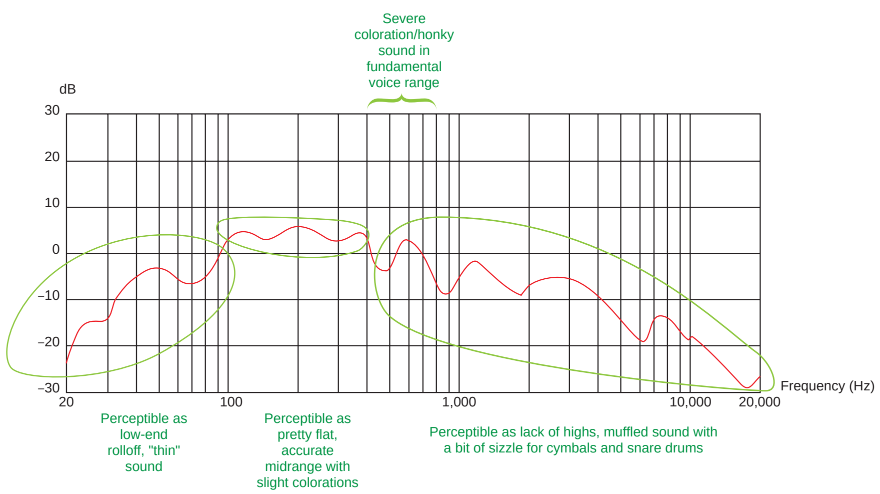


Figure 1: This sample frequency response with deviations from "flat" demonstrates how listeners would perceive these deviations.

of tune: these represent modulation distortion, quantizing distortion, and various noise-modulation forms of distortion. The latter forms of distortion are quite objectionable, and they have been shown to add to listening fatigue even when the listener is not consciously aware of them. Harmonic distortions are much less objectionable, as they primarily modify the music's timbre. A good example of harmonic distortion's sound is the honkiness of used-car lots' paging systems.

Loudspeaker systems create primarily harmonic distortion, although there is also significant modulation distortion. The severity of these distortions is directly related to the cone's excursion. Thus, the larger the woofer and the better job the system does at extracting maximum sound from minimum excursion, the less the distortion, in general.

Since the greatest excursions occur in reproducing low frequencies, most speakers produce their greatest amounts of distortion in that range. Full-rated-power distortion figures at low frequencies are seldom quoted on loudspeaker systems, to protect whatever sanity the marketing department may still possess. But the subjective effect of harmonic distortion—especially low-order odd-harmonic distortion—at low frequencies, is enhanced bass. Only very critical listeners can easily distinguish between this "false bass" and true bass.

In the midrange, speakers tend not to produce much distortion. However, at high frequencies, magnetic nonlinearities mediated by eddy currents in the steel parts cause the distortion to rise again. Of course, if the limits of hearing are defined as 20 to 20,000 Hz, as is usually done, harmonic distortion of fundamental waves 10 kHz or higher is of no consequence, since the lowest harmonic is at 20 kHz. Further, the comments made earlier about the insensitivity of the ear/brain to frequency-response anomalies at higher frequencies also applies to distortion. Thus, as long as the distortion products are harmonically related to the desired signal, it can safely be said that a few percent harmonic distortion above 5 kHz is unimportant. Modulation distortion is another matter, since it produces "out-of-tune wrong notes" in virtually any frequency range, regardless of the range in which the fundamentals appear.

A very good speaker for live sound may have harmonic distortion below 6% at low frequencies, below 2% in the midrange, and below 3% at high frequencies, measured at an output of 115 dB$_{SPL}$. Speakers for home use are unlikely to be much better at an output level 20 dB lower, and may be worse. Since this distortion is primarily low-order harmonics, it does not sound as bad as the numbers might make us expect.

Next, consider the room. There is widespread agreement that the loudspeaker-room interaction is quite important, but less agreement as to details. The point at which this discussion intersects loudspeaker system design primarily concerns directivity. In general, virtually any loudspeaker system is nondirectional at low frequencies, and much more directional at high frequencies. Since the ear/brain distinguishes between the direct sound and the "room sound" after about 50 ms, there was once a question as to whether a room's direct or the reverberant field should have flat response. In other words, should a speaker produce a flat pressure response on axis or a flat total acoustic power response? The former is now agreed to be preferable, since the latter creates a very harsh sound, especially for listeners on axis. However, it is important that total acoustic power response be smooth, with no large peaks or dips.

Any room reinforces some frequencies and partially cancels others, depending on where the listener is located in the room. At low frequencies, these "room modes" are fairly broad and spaced rather far apart. For example, a living room that is 3.12 m × 4.69 m × 6.24 m will have its lowest mode at 27.5 Hz, its second mode at 36.6 Hz, and its third mode at 45.9 Hz. In this range, there are only four modes per octave. At 100 Hz, the modal density increases so there are about 25 modes per octave. Thus, above this frequency, the modes overlap to the point that they no longer cause irregularities. Interference from direct reflections becomes the dominant concern. The effect of lower-frequency modes can be lessened using a single subwoofer and by avoiding directional (doublet or cardioid) subwoofers, which excite fewer modes. (Modal density is our friend!)

The primary room-related concern for speaker system designers, in midrange frequencies, is reflections. These usually reach the listener in such a short time that they are not perceived as echoes, but they do cause reinforcement and cancellation at some frequencies—humps and dips in the frequency response. To help control deleterious reflections in home systems, a good target for midrange and high-frequency vertical directional patterns is about 20°. The horizontal radiation should cover about 90°, although this depends on the room's shape and layout. Requirements for permanent-install and concert systems will vary widely with the intended venue. At low midrange frequencies, this target will be difficult

Speaker Specifications	
Frequency response	Flat +1 dB from 20 Hz to 20 kHz
Directivity	90° horizontal and 20° vertical
Maximum SPL	130 dB_{SPL}
Aesthetics	Small, unobtrusive solid walnut cabinet with spherical front
Distortion	Less than 0.1% at all frequencies, measured at full output SPL
Magnetically shielded	100%

Table 1: These may be the specifications for a "dream" design for some applications, but they may not be practical to achieve.

to achieve in many applications, but driver size, angular mounting, and the correct use of array technology (e.g., in the D'Appolito array) can help. In many cases, the most deleterious reflection is the bounce from the floor, which causes a low-midrange dip in the response of many floor-standing speaker systems.

At high frequencies, excessive reflections from walls will interfere with spatial localization in home systems and will decrease intelligibility in professional installations. Certainly some wall reflections are desirable to impart a sense of spaciousness, but since the out-of-pattern sound level never drops to zero, this requirement will always be met.

Now, the other shoe. Yes, we want the on-axis frequency response to be as flat as possible, but the balance of frequencies we feed into the room from off axis also matters. The perception of naturalness in the loudspeaker system's sound depends significantly on the smoothness of the total acoustic power response. This simplifies to a requirement for a gradual transition from the non-directivity at low frequencies to the high directivity at high frequencies. If a 30-cm woofer is crossed into a 25-mm dome tweeter at 1 kHz, the directivity will abruptly change from perhaps 60° to almost 180° in a small span of frequencies. This will prevent the system from ever sounding integrated and seamless. The requirement for a smooth transition in directivity is not often met, except in the best systems.

The last issue to be discussed regarding the loudspeaker-room interaction is the speaker system's transient response. All other things being equal, the best possible transient response award must go jointly to electrostatic and horn systems. (Since most electrostatic systems use ancillary subwoofers, all-horn systems may win by default.) Second would be closed-box systems; third, vented systems—with lower-order systems better than higher-order ones. Passive-radiator systems fall just above bandpass systems at the bottom end of the scale. However, many astute observers have questioned how much the transient response of the speaker system really matters, since the system will be operated in a room having a non-zero decay of reverberation. (The use of the term "reverberation time" in relation to home listening rooms and recording studio control rooms is not necessarily justified, as room-acoustical conditions for meaningful use of the term are seldom met, at least at lower frequencies, in home and control rooms.) A speaker having poor transient response may exhibit a 20- or 30-ms decay of an impulsive sound. Rooms having a decay time under 0.1 s (100 ms) are quite rare and certainly not desirable as listening rooms. It is not at all clear that the speaker decay time matters under these listening conditions.

SETTING SPECIFICATIONS

The most obvious, and yet often overlooked, principle in setting system specifications is to first consider the application. Most speaker engineers became interested in the field as a result of a prior interest either in music listening or in music performing. The kind of music that was heard or performed tended to color the engineer's future biases concerning speaker design. For example, a classical-music listener is more likely to think in terms of frequency-response extension and spaciousness of sound, while a rock music performer may be primarily concerned with blistering bass and sizzling highs. Simply as a consciousness-raiser, I'll mention some important areas related to certain kinds of speaker-system applications. These are given in order of their relative importance.

Home stereo:

- frequency response flatness, extension
- directivity
- aesthetics
- distortion performance
- maximum SPL

Aftermarket autosound:

- maximum SPL
- aesthetics
- maximum power handling
- frequency response flatness

Professional sound (permanent-install and concert):

- directivity
- maximum SPL
- frequency response flatness
- distortion performance
- aesthetics

Multimedia:

- small size
- magnetic shielding
- low cost
- aesthetics
- frequency response extension

These lists are not intended to be either prescriptive or exhaustive, but simply to stimulate thinking.

With these priorities in mind, you can take the first step in designing a speaker: deciding on the target numbers for each specification. If you are only dreaming, and have no intention of actually building the speaker, you can certainly choose the set of specifications in **Table 1**.

Such a speaker might seem ideal to some people, for some applications. Actually designing one is a pipe dream. Engineering has been described as "the art of the possible." I hope this article has helped to clarify what the specifications mean, and what level of performance is possible and desirable.

In a future series, I will discuss the factors of sound radiation and loading of speakers, and how these factors influence performance. Then I will be ready to discuss design methods. *aX*

 By David J. Weinberg (United States)

CEDIA 2012

Evolution in Home Technology

A review of CEDIA 2012

The Custom Electronic Design & Installation Association's (CEDIA) Expo 2012 emphasized integrated home monitoring and control, custom-installer productivity, and featured Dr. Michio Kaku's keynote about his expectations for the intuitive home of 2016.

I did not find Dr. Kaku's predictions especially advanced, even for only four years out. Technological improvements and functional integration are well underway, and I don't expect the home's intuitive responsiveness to be adequately sophisticated for the better part of a decade from now.

THE SHOW

CEDIA (www.CEDIA.org) reported approximately 17,000 attendees (including 15% first-time) from 71 countries, a 4% non-exhibitor attendance gain over last year, as well as more than 450 exhibitors (including 90 new) and more than 80 product debuts. CEDIA Expo 2012 offered 175 training courses (including manufacturers' training) with a focus on educating electronic systems contractors about home computer networking (with 22 new advanced IP-networking courses) in addition to fundamental and advanced home technology education.

CEDIA has used comments from a number of people, including me, to update the *CEDIA Electronic Systems Technical Reference Manual*, releasing the second edition at this CEDIA Expo. I reviewed the first edition and failed it for its many scientific and technical errors. CEDIA's new education director told me that errata for the first edition will be available soon, and that I likely will be given the opportunity to review them. I have not yet reviewed the manual's second edition, and I advise readers to critically evaluate its contents.

SURROUND PROCESSING

Datasat Digital Entertainment (www.DatasatDigital.com; from 1993-2008 known as the DTS cinema division), respected for its AP20 professional cinema processor, exhibited the RS20i home theater processor (http://DatasatDigital.com/consumer/products/rs20i.php; $18,500 SRP), substantially based on its AP20 professional product, and including capabilities required for home theater application that are not needed for movie theater use. The RS20i supports up to 16 channels of digital audio (including 11.1-, 12.2-, and 12.4-channel configurations); bandpass filters, third-octave graphic and parametric equalization for each channel; four subwoofer outputs; Dirac Live system optimization technology ("impulse-response correction"; http://DatasatDigital.com/consumer/home-cinema/room-optimization.php); and 20 user-defined setup and operational profiles.

SPEAKERS & HEADPHONES

Atlantic Technology (www.AtlanticTechnology.com) demonstrated its H-PAS PowerBar 235 ($900 SRP), which it claims delivers relatively high-output bass down to 47 Hz and doesn't require a subwoofer (although there is a subwoofer output "for optional subs if you're an incorrigible bass hound"). The 43"-wide device uses multichannel DSP to "create a 2-, 3-, or 5-channel virtual sonic landscape for immersive sound." In two-channel mode, the music had bass that was deep enough and sufficiently smooth for most customer environments. Listening in two-channel mode to a recording of Diana Krall ("My Love Is"), I heard good, strong bass, faintly muffled vocals, and a bit of sibilance. The speaker seems to slightly throw the vocals at the listener. The sound room was a bit dead toward the back, so in five-channel mode the Dolby basketball trailer's surround effect was weak (as with all soundbars, the surround effect depends a great deal on room acoustics and symmetry). However, when playing an excerpt from the movie *Super 8*, dialog sounded fine, and the overall sound was good for a soundbar.

GoldenEar Technology (www.GoldenEar.com) debuted the 49"-wide SuperCinema 3-D Array Soundbar ($1,000 SRP), comprising six 4.5" mid-bass drivers and three high-velocity folded ribbon (HVFR) tweeters. It is said to have a 60-Hz low-frequency limit. Its DSP (always active) is claimed to cancel "crosstalk distortion between the left and right channels," stretching the front soundstage to "effectively envelop the listener in a true 180° soundfield." In its sound room, the soundbar was demoed with a 120-Hz crossover to the Forcefield 5 subwoofer ($1,000 SRP). It was a pleasure to hear Mel Torme's voice ("Once In Love With Amy" from *Mel Torme Swings Schubert Alley*) sound so natural, with a very slight bottom-end boost. Perhaps the subwoofer's level was a bit high, as the bass sounded a little strong on the Dolby bouncing-ball trailer. GoldenEar founder, president, and chief designer Sandy Gross played an excerpt from Mahler's *Symphony No. 1* (Michael Tilson Thomas conducting the San Francisco Symphony Orchestra) through the soundbar with a subwoofer and two small wall-mounted surround speakers, which sounded pretty good.

PSB Speakers' (www.PSBSpeakers.com) M4U 2 active-noise-canceling over-the-ear headphones were comfortable. Their noise-canceling function slightly attenuates music's bass as it cancels a lot of the ambient lower-frequencies. Their overall sound is mellifluous—not especially accurate, but appealing.

VIDEO

3DTV was marketed by many display manufacturers, but was not as strongly hyped as in recent years. So-called 4k resolution (twice the horizontal and vertical pixel count of 1920 × 1080 HD displays) was this year's hyped video technology, in spite of the almost total lack of consumer-accessible 4k content and network-distribution limitations.

JVC (http://Newsroom.JVC.com/news-blog/performance-enhancements-in-new-expanded-jvc-projector-line/ and http://Pro.JVC.com/prof/attributes/category.jsp?productId=PRO2.2) demonstrated what I call their pseudo-4k projector, as it uses e-shift2 DLP technology to upscale and display 2-D images. e-shift2 is similar to what several years ago was called wobulation, which half-shifts pixels and interpolates data to generate simulated 4k images. JVC's definition of 4k is 3,840 × 2,160, which is twice the horizontal and the vertical pixel counts for most HDTVs (1,920 × 1,080), but differs from Digital Cinema 4k (maximum pixel count: 4,096 × 2,160). JVC projector models with e-shift2 are DLA-X95R, DLA-X75R, DLA-RS66, DLA-RS56, DLA-X55R, DLA-RS48, and DLA-RS4810 with prices ranging from $5,100 to $12,000. The DLA-X95R and DLA-RS66 come with two pairs of 3-D glasses and one RF emitter. JVC claims that the RF connection to the 3-D glasses offers better synchronization stability than IR connections, and avoids interference with IR remote-controlled devices. Additional PK-AG3 RF 3-D glasses cost $180 SRP, and a PK-EM2 3D RF emitter is $100 SRP.

Sony demonstrated the VPL-VW1000ES SXRD three-panel LCoS 4k projector (https://DealerSource.Sel.Sony.com/dsweb/p/builtin/home_theater_projectors.html; $25,000 SRP), which was also shown at the 2011 CEDIA Expo. The picture was pretty, and seemed fairly well calibrated. Its black level was a bit above inky black, but the PLUGE was set correctly and the colors appeared reasonably accurate, if a little oversaturated (a SMPTE color-bars with a PLUGE test pattern appeared briefly). There were some grain and sparkles in the image, possibly due to Sony having chosen a Stewart StudioTek 130 screen, which is not smooth enough for such a high-resolution image (the grain in the screen looks like video noise and can mask video noise in the projected image). Sony's 84"-diagonal 176-lb XBR-84X900 4k (3840 × 2160) LCD LED-edge-lit TV ($25,000 SRP) supports passive 3-D imaging (http://Store.Sony.com/webapp/wcs/stores/servlet/CategoryDisplay?catalogId=10551&storeId=10151&langId=1&identifier=S_4KTV).

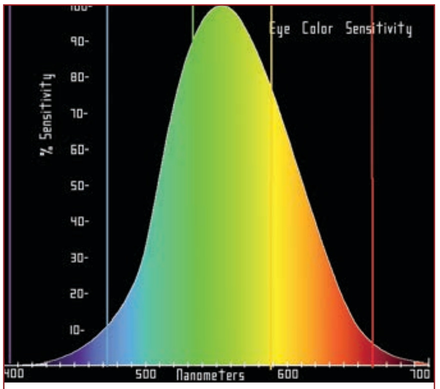


Figure 1: This graph shows overall human visual relative spectral sensitivity versus wavelength. (Photo courtesy of Public domain image authored by Skatebiker at the Wikipedia project, https://en.Wikipedia.org/wiki/File:Eyesensitivity.png)

One step below the always reference-quality image at the Da-Lite booth (www.Da-Lite.com; Joe Kane [Joe Kane Productions; www.VideoEssentials.com] with his Samsung SP-A900B projector and Da-Lite JKP Affinity screen), the Sony 4k and JVC pseudo-4k projected images that appeared to be reasonably accurate, which means that there were more apparently calibrated displays at the 2012 CEDIA Expo than I have seen at this show for many years.

HOME CONTENT DISTRIBUTION

The latest version of the HDMI (www.HDMI.org) specification (1.4a) requires that no version numbers be used on the cable, its packaging, or in its documentation. All are now identified as one of five HDMI cable types ("each with specific performance characteristics"), with a list of supported additional connectors and options. There are 11: HDMI Ethernet Channel, Audio Return Channel, Dolby True HD and DTS-HD with lip sync, 3-D, 4k/2k Support, Content Type—real-time communication between source and display enabling optimized picture settings based on content type, additional color spaces (e.g., xvYCC, 24-to-48-bit Deep Color, plus for digital photography or computer graphics), bandwidth up to 10 Gbps, and the HDMI Micro Connector.

Accell's (www.AccellCables.com)

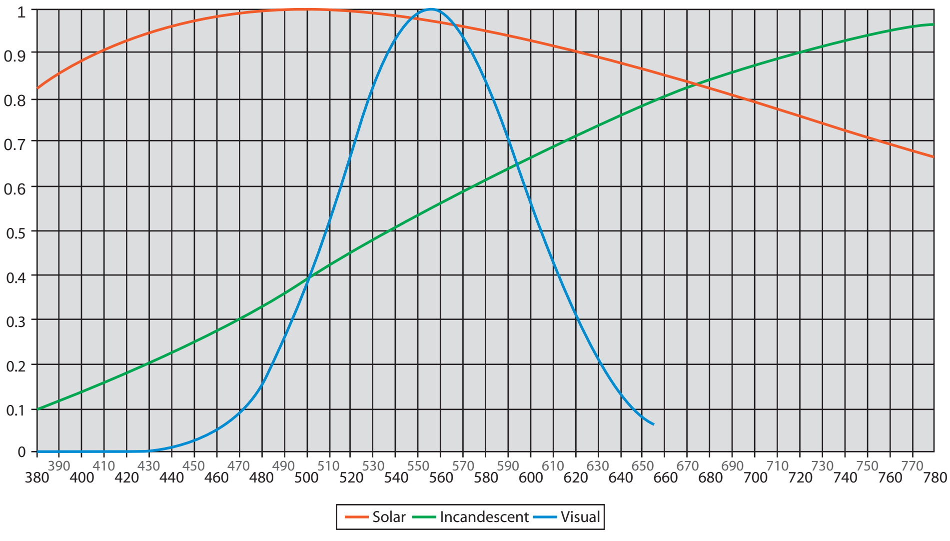


Figure 2: This graph shows the approximate spectra of: the sun (curve near the top of the image), an incandescent light bulb (curve that slopes from bottom left to top right), and human vision (centered curve that peaks near the middle of the horizontal axis), over the 380-to-780-nm wavelength range of visible light. Each curve is normalized to show peak light energy near the top of the graph. (Photo courtesy of www.RoperID.com/science/electromagneticspectraoflightbulbs.htm)

expanded line of AVGrip Pro Locking High-Speed HDMI cables supports Ethernet and an audio return channel (HEAC).

Celerity Technologies (www.CelerityTek.com) announced its fiberoptic HDMI "solution" that supports the HDMI audio return channel. Its fiberoptic cables have a small proprietary connector at each end, which makes it easier to fish the cable inside walls or through smaller-diameter conduit. Two adapters come with each cable that converts the proprietary connection at each end to HDMI. Costs range from more than $250 to $1,800 for lengths more than 40' to 1,000'. The company is working on adapters so the fiberoptic cable can be used as an Ethernet cable.

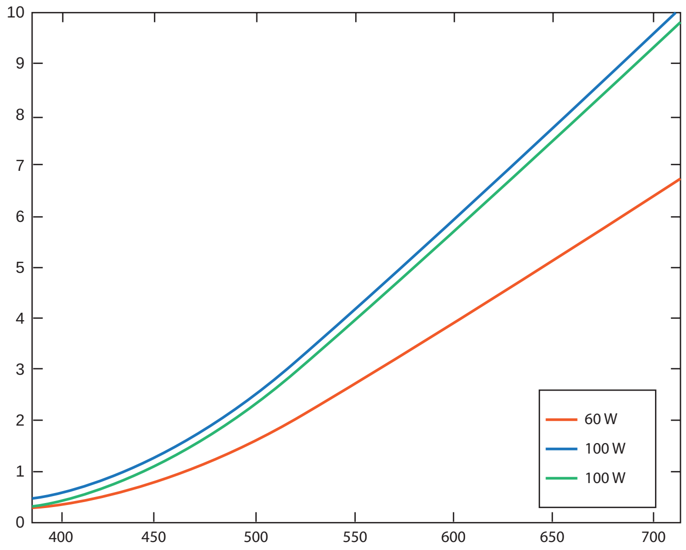


Figure 3: Spectra of GE incandescent light bulbs "GE Super Soft, Soft White" shows how light bulbs generate light in different ways. (Photo courtesy of www.Graphics.Cornell.edu/online/measurements/source spectra/index.html)

DVDO's (www.DVDO.com) Air (www.DVDO.com/air.aspx; $400 SRP) uses 60-GHz wireless technology (generally limited to approximately 30' distance) to deliver uncompressed digital video (up to 1,920 × 1,080 p60) and up to 7.1-channel losslessly data-reduced surround sound. The Air is compliant with HDMI 1.4a (3-D and CEC options included) and HDCP. The DVDO Quick6 (www.DVDO.com/Quick6.aspx; expected less than $500 SRP) is an eight-port (six inputs, two outputs) HDMI Smart Switch (which means it continuously maintains the HDMI handshake with each connected source and display, internally switching the signals without needing to validate each connection each time; this substantially reduces the time for the new source's image to appear). Quick6's InstaPrevue enables display of a live picture-in-picture preview of each HDMI-connected source, facilitating selection among them.

Wireless Speaker & Audio (WiSA) Association (www.WiSAAssociation.org) is a year-old industry group whose "main charter is to certify interoperability between WiSA-compliant transmitters, such as DTVs, set-top boxes, and Blu-ray players, with receivers, comprising home-theater speakers from many brands." WiSA released its "first Compliance Test Specification for interoperability testing of the wireless link between WiSA-compliant speakers and home theater and consumer electronics devices such as DTVs, set-top boxes, Blu-ray Disc players and game consoles." WiSA-approved wireless technology transmits 2 to 7.4 channels of 24-bit non-data-reduced digitized audio at up to 96 ksps over the U-NII frequency spectrum (international unlicensed 5-GHz band that requires dynamic frequency selection (DFS). A WiSA-compliant system is required to prevent conflicts with weather and military radar transmissions, and tries to find an unoccupied in-band channel (among the 24 possibilities) while it looks ahead for an alternate unoccupied channel to facilitate jumping, in case of interference, without losing audio data. WiSA-compliant systems deliver the audio with a fixed 5-ms latency to each loudspeaker, with a less than 0.00016-ms speaker-to-speaker differential. WiSA-compliant systems incorporate forward error correction and error concealment. Although WiSA-compliant systems eliminate wires from a central location to all loudspeakers, any technology that wirelessly transmits audio to speakers requires an amplifier at or inside each speaker, and an AC outlet nearby.

LET THERE BE NEW LIGHTS

We see light with varying color sensitivity, as approximated in **Figure 1** and again in the centered-peak curve in **Figure 2**. The spectrum of the sun, man's original source of light, is approximated by the continuous spectral curve near the top of **Figure 2**.

Light bulbs generate light in different ways, some with a continuous broadband color spectrum (see the upward sloping curves in **Figure 2** and **Figure 3**),

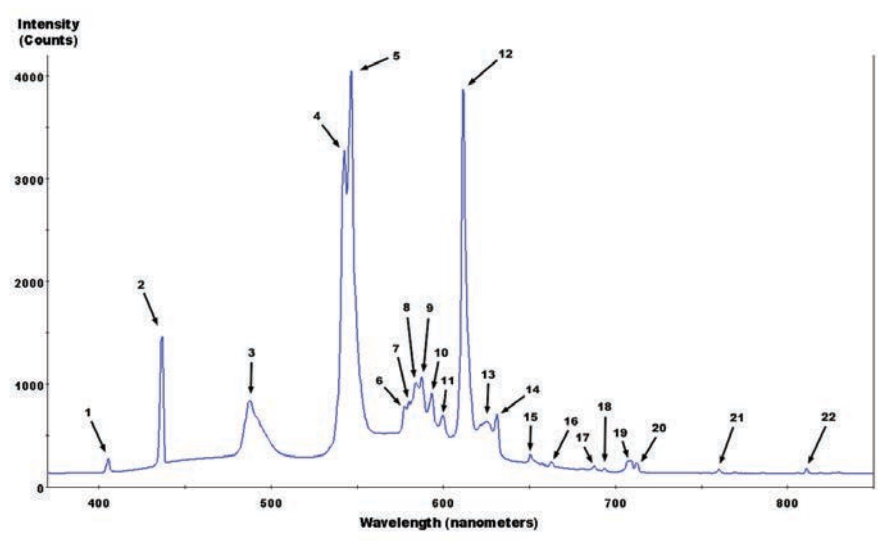


Figure 4: Relative-intensity spectrum of a typical cool-white fluorescent lamp shows the typical fluorescent spectrum. (Photo courtesy of http://commons.wikimedia.org/wiki/File:Fluorescent_lighting_spectrum_peaks_labelled.gif)

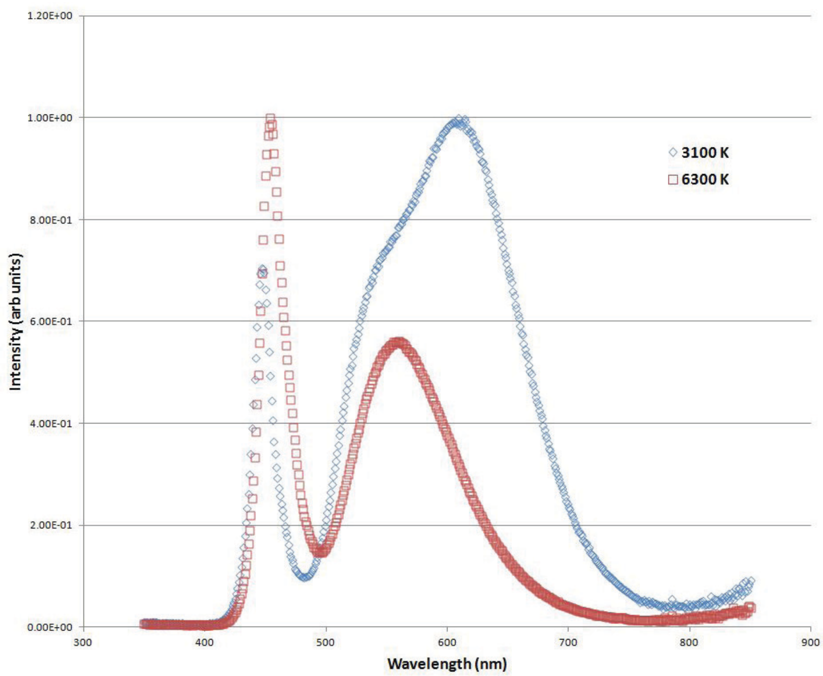


Figure 5: Spectra of two Firefly LED lamps reveal correlated color of 3,100 K and 6,300 K (Photo courtesy of Firefly LED Lighting)

others with a set of scattered narrow peaks (see the typical fluorescent spectrum in **Figure 4**).

LEDs typically generate light in very narrow spectra (although not as narrow as a laser, which is almost a single-spectrum line at its specific frequency). There are LED technologies that can generate somewhat broader spectra, although they cannot easily match the continuity and smoothness of incandescent or solar light. This is why many of us find reading in fluorescent lighting relatively uncomfortable, and why some experts claim that using LEDs for video backlighting and front/rear projection lamps can't deliver a video image with completely correct spectral balance, even when the advertised correlated color of the lamp is the correct 6,500 K, or the lamp has a high color-rendering index (CRI).

There is information about the Academy of Motion Picture Arts and Sciences' (AMPAS) solid-state lighting project at www.Oscars.org/science-technology/council/projects/ssl/index.html (especially the series of short symposium videos accessible via links along the right side of the webpage), with additional technical material at www.Oscars.org/science-technology/council/projects/ssl/technicalinfo.html. These sources demonstrate the spectral changes that occur when different light spectra illuminate the Macbeth color test chart.

NuLeds (www.NuLEDs.com) LED lighting offers lower heat generation and higher efficiency than incandescent bulbs, color balance that I prefer over fluorescent lamps (especially for reading), and fixtures that receive power and can be managed over Ethernet. The NuLed models on display come in a correlated color temperature of 2,700 or 3,500 K, far below (warmer; "redder") than the video reference color temperature of 6,500 K. Thus, these lamps are inappropriate to use as low-level background lamps in home theaters. No spectral data was made available.

Firefly LED Lighting (www.FireflyLEDLight.com) offers alternatives to incandescent and fluorescent bulbs. Its lamps fit in standard lamp sockets. It offers different lenses for different light dispersion patterns. Its LED lamps are offered in one of two color temperatures: 3,000 K (in the range of typical incandescent bulbs) and 6,000 K (which is reasonably close to the 6,500-K standard for video's correlated color of white). They can be used as soft ambient lighting in home theaters, although the user should be aware the overall spectrum is not a perfect match to sunlight or incandescent light bulbs (see **Figure 5**).

MISCELLANEOUS

BlueVolt (www.Bluevolt.com) is a "provider of online learning management systems for the manufacturing, construction and service industries, [delivering] training across an entire company and its suppliers, sales channel, associations, and customers." More than 223,000 registered users have taken almost 1.4 million BlueVolt training courses on Internet-connected computers, tablets, or smartphones. CEDIA has adopted BlueVolt technology for some of its training courses.

Jensen Transformers (www.JensenTransformers.com) continues to be my preferred source for accurate and helpful information on the causes, solutions (including products), and preventive measures related to system grounding problems [see its white papers, many written by Bill Whitlock, who was a key member of the group that developed AES 48-2005 (revised 2010) Standard "AES standard on interconnections—Grounding and EMC practices—Shields of connectors in audio equipment containing active circuitry" and who correctly explains the balanced interface that is so often incorrectly described in print].

One testament to the growth of mobile commerce is the Square (https://SquareUp.com) card reader that plugs into the iPhone, iPad, or Android device and links to your bank account, enabling you to accept American Express, Discover, MasterCard, and Visa payments wherever you are. The company, written up in MIT's Technology Review, also has developed Square Wallet that enables hands-free checkout with certain smartphones at a growing list of businesses, just like you have been doing with credit/debit cards and cash.

A competitor is Pay Anywhere (www.PayAnywhere.com), which provides the service and a small adapter for many models of Apple, Android, and Blackberry mobile devices that enables you to accept American Express, Discover, MasterCard, and Visa payments.

EDUCATION & CERTIFICATION COURSES, PLUS TRAINING SESSIONS

Industry consultant and journalist Michael Heiss delivered his 18th annual **"New Technologies Update,"** which is always informative and entertaining. The course objective is to give attendees "a greater understanding of the relationship between the technologies that drive [the home-technology] industry, the current status in key technology areas, and an idea of where they are going over the next 12 to 24 months." He discussed: strengths and weaknesses of smart TVs versus over-the-top broadcasting versus IPTV; 3-D, 4k, and high frame-rate video; various wireless and wired in-home and to-home data pipes and protocols; new surround sound schemes (e.g., Dolby Atmos); new smartphone and "pad" products and software from Apple, Samsung, Google, and Microsoft; and display trends. As Heiss concluded, "So many technologies, so little time!"

Consultant and professor Rich Green told his audience about **"Future Technologies: The Scoop From Silicon Valley."** He emphasized that economic factors have reduced profit margins while service and operation costs have substantially increased. He added that custom installers need to become IP-focused, and that mobile devices and the cloud, along with company agility and diversification, also are keys to a successful custom installation business. Green is cognizant that we "tend to overestimate short-term changes and underestimate long-term change." Theodore Levitt (in Marketing Myopia, 1960) wrote "the history of every dead and dying 'growth' industry shows a self-deceiving cycle of bountiful expansion and undetected decay." Green believes that traditional business models need to be re-evaluated. He sees a five-year trend leading to in-home wired and wireless networks with everything connected, plus live connections to family and friends; computing devices becoming more like appliances—ubiquitous and controlling everything; media streaming from the cloud to all types of devices; a growth in user interfaces that are based on conversational voice and gesturing; increasingly immersive entertainment, gaming, social computing and virtual reality; and growth in digital home healthcare. He quotes a study that "8- to 18-year-olds in the U.S. spend one-quarter of their media time using multiple media." He sees Apple, Amazon, Facebook, and Google leading the software industry's efforts to, as Marc Andressen put it, "take over large swaths of the economy." Green emphasized that trend with discussions of the Square (which enables iPhone and other Mac users to swipe credit cards for payment) and other methods of ordering and payment using smartphones. He calls Google and other Internet-activity trackers as representing "the era of frictionless surveillance," and that with the plethora of mobile connected devices we are in the "post-PC era." Content is becoming available anytime on any device, we are even able to migrate during a program from one device to another. He talked about 4k and 8k monitors, physically flexible OLED video displays, active and passive 3-D, automated system/room equalization, Dolby Atmos, online music and video program sources, DACs, audio codecs, apps, games, controllers, healthcare robotics and apps, and so forth. He quotes Danny Hillis (Long Now Foundation), "We humans have linked our destinies with our machines. Our technology has gotten so complex that we no longer can understand it or fully control it. We have entered the Age of Entanglement." His final comment is that "it's all about the services. Software will overtake hardware. Be the Digital Concierge."

Vic Cypher, Cypher Communications, told us there should be **No New Wires: Using Retrofit Technologies to Increase Efficiency**. He is a proponent of custom installers pursuing retrofit work to maintain, and even expand, their businesses. He talked about unshielded twisted pairs (UTP), carrying Ethernet, analog and digital video and audio, including HDMI extenders, coaxial cable (MoCA; www.MoCAlliance.org), HomePlug (powerline; www.HomePlug.org) technology, Wi3 (wired wireless wideband; http://Wi3Networks.com), WirelessHD (www.WirelessHD.org), Wi-Fi (www.Wi-Fi.org), WHDI (www.WHDI.org), WiGig (www.WirelessGigabitAlliance.org), and more. He emphasized the need for extensive communication with the client throughout the process, to determine and meet their needs and wants.

Floyd Toole delivered his well-received three-part **Home Theater Audio and Acoustics**, primarily based on his book *Sound Reproduction: The Acoustics and Psychoacoustics of Loudspeakers in Rooms* (approximately $40 at www.Amazon.com). He explained the fundamentals of sound and acoustics as they relate to the choice and placement of speakers and acoustical materials, how the room affects bass frequencies, the general requirements for optimizing a surround system in a given room, specific speaker characteristics to get good sound and appropriate directional and spatial illusions, how to distinguish between important and uninformative specifications, and how to think about the customer's needs and constraints in designing a home-theater system. Toole worked to "provide an understanding of how acoustics and psychoacoustics influence what we hear and how to take advantage of that knowledge to optimize the listening experience," discussing strategies and using illustrations.

Acoustic Isolation and Noise Control informed attendees on the bidirectional nature of this problem (keeping home noise out of the theater, and home-theater sounds inside the theater), addressing topics including selecting the best location for the home theater, deciding on "transmission loss requirements for walls, ceilings, floors, windows and doors, [plus] how, using common materials and some proprietary ones, to achieve the necessary performance." The course also discussed "what can and cannot be done in real-life situations, especially in retrofits."

Headphone Audio and Human Hearing: What You Need to Know was developed by the Home Acoustics Alliance "to teach engineers, product designers, and salesmen how humans hear and specifically how headphones differ from loudspeakers." Many aspects of theory and design were addressed and live demos underscored the science behind imaging and psychoacoustics.

Audio Setup and Calibration addressed the need for calibration, sound-quality metrics (e.g., clarity, focus, envelopment, dynamics, frequency response, and seat-to-seat consistency), electronic versus mechanical (e.g., speaker position and acoustic treatment) calibration, the importance of proper equipment design and system layout, the interaction between design and calibration, system components, required tools, a step-by-step calibration procedure, and system performance verification (including listening to thoroughly known source material [see **"Learn to Listen"** below]). [Some of the recommendations strictly follow CEDIA, International Telecommunication Union (ITU), and other standards, when deviation based on thorough understanding can yield better results.] The course directs the custom installer to "balance customer priorities with proper design principles," and to market system calibration as an added-value line item. Its rational approach to speaker cables maintains the course's credibility: "Stranded copper wire is still unbeaten for cost-effective top performance, strength, and durability. This is not a statement condemning high-end cables, but it is worth noting that many of the sound quality ills supposedly cured by high-performance cables are much more effectively handled by other means of calibration." However, how balanced interconnections work and their benefits are incorrectly described.

Video Setup and Calibration stepped through digital-video signals and levels, colorimetry, Society of Motion Picture and Television Engineers (SMPTE) standards, test patterns and tools needed, plus a 12-step procedure for calibrating the display. The course explains the effects of user controls (e.g., sharpness and how incorrect adjustment can distort the image) and recommends verifying the results with known reference material.

Frederick J. Ampel's (Technology Visions Analytics) course **"Learn to Listen"** returned this year with enhancements. This two-hour session, aimed at helping custom installers sell surround-sound systems, had two theses:

- A customer's own CDs can be used to demonstrate the benefits of having a surround sound system (extracting and spatializing ambient sounds from the recordings).
- While using measurements can result in fairly decent calibration of a surround-sound system, final adjustment should be performed while listening to carefully selected content very well-known by the calibrator (a technique that is recommended in the annex of SMPTE standard ST 202:2010 for movie-theater sound system calibration).

Ampel demonstrated both assertions, explaining the effects as he played two-channel music sources (switching between stereo and surround processing), and describing what to listen for from multichannel movie soundtracks. It was clear that almost any sound system can make loud noises, as few listeners know what they sound like. (When was the last time you were at the location and heard a train crash?) However, we all know the sound of footsteps in the grass or on a wood porch, or the sound of a pencil falling onto the floor. It's the realism of the small, quiet sounds that convince us the scene is real and pull us out of our living rooms and into the movie.

As part of this course, I delivered a brief explanation of the components of movie soundtracks and how they are created and combined, including the importance of SMPTE standards to this process.

Setting up the sound system for the course, Ampel used the AudioControl (www.AudioControl.com) Iasys HT (www.AudioControl.com/t37/60172/19893/Stand-Alone/Iasysreg-HT-with-HT--100-Home-Theater-Analyzer.html; $4,000) to get the initial EQ and relative speaker-to-speaker time delays needed to generate a coherent soundfield at the prime listening position, entering the settings into the AudioControl Diva (www.AudioControl.com/19872/products/Room-Correction-Processor----Diva.html; $10,000). I have experimented with these two devices in my home surround system and found them easy to use and quite helpful.

Other equipment used for course demonstrations was provided by manufacturers: AudioControl (Maestro surround processor, Diva, Savoy multichannel amplifier, Iasys HT), Straight Wire (www.StraightWire.com; interconnects and speaker cables), and Triad Speakers (www.TriadSpeakers.com; InRoom Gold LCR speakers; four InRoom Gold in spaced pairs for left/right surrounds; five InRoom Gold subwoofers, each with a DSP BASH-technology amplifier).

As part of CEDIA's focus on the increasing importance for custom installers to understand IP technology, **Remote System Access: Methods, Security, and Best Practices** described various methods to remotely access home control, monitoring, and security systems; plus how to set up and troubleshoot the Internet connection. I think far too much of this course was spent extolling the benefits of remote access. Security was addressed, with some emphasis, but I don't believe enough, given to the need and procedures for protecting the remote access from hackers. For example, I didn't see any mention of the need to change default passwords in almost all routers, modems, and other Internet-connected devices, nor the characteristics of good passwords, such as being of adequate length, varying case (UPPER/lower), a mix of alphanumerics, not being word-based, and so forth.

OBSERVATIONS

This CEDIA Expo seemed to be somewhat more reserved than in recent years. Perhaps the industry is showing greater maturity. CEDIA Expo 2013 will be held September 25–28, 2013 in the Colorado Convention Center, Denver, CO. Perhaps I'll see you there. *aX*

A Look Inside *Electric Guitar—Sound Secrets and Technology*

New book hits the right note for both musicians and technicians

Electric Guitar—Sound Secrets and Technology
Helmuth Lemme Elektor, 2012, 476 pages ISBN 978-1-907920-13-4
$47.60

In his book, *Electric Guitar—Sound Secrets and Technology*, Helmuth Lemme addresses the average consumer/musician who is technically interested in and familiar with certain fundamentals of electronics (see **Photo 1**). The communication between musicians and technicians is often difficult. He wrote his book as a bridge between these two worlds. A cult following has arisen around certain electric guitars and basses that has been crafted in part by clever marketing and sponsorship by popular performers.

Advertising works on an emotional level and glorifies consumer products. While every musician wants to have the best sound, it is possible to improve on many commercial instruments for little money. It just depends on the application of accurate and proven knowledge. This book aims to expose the fictitious, idolized world and bring it back down to earth.

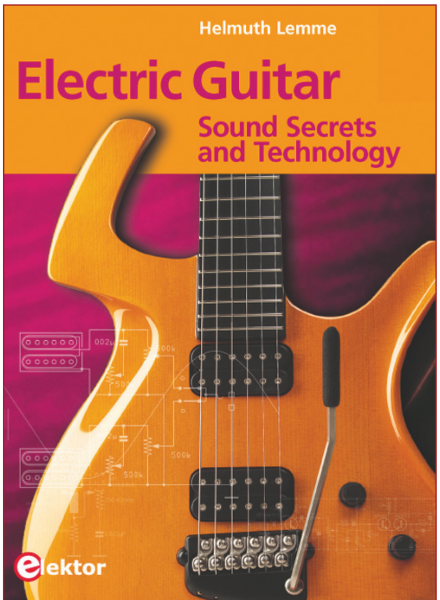


Photo 1: This book (available at www.elektor.com) explores the guitar's mechanical and electrical parts. Manufacturer's secrets are unveiled, pickups and their electrical environments are explained, and several tips for the technically inclined musician, can improve the sound for many instruments.

THE FOUNDATION

Chapter 1 presents a concise yet comprehensive historical review of the breakthroughs and development of all types of guitars—from traditional Hawaiian guitars to flat-top classical, full-body archtops, solid body, and semi-acoustic guitars. This chapter also has a section on electric basses.

Chapter 2 gets into the real substance of the book—how the strings, body, neck, frets, bridge, and all their various mechanical resonances determine the guitar's sound before any electronic components are added. Other factors that influence the guitar's acoustic sound are room acoustics, any venue amplification that may be used, and even the way a guitarist holds and plays the instrument.

PICKUPS

Chapter 3 is the book's longest chapter, and the author aims to demystify magnetic pickups. A pickup creates an electrical signal with its own transfer characteristic by absorbing vibration energy from the guitar. The fine nuances are defined by the mechanical properties examined in Chapter 2.

A poorly constructed guitar cannot be improved by even the best magnetic pickup design. I learned that when I naïvely tried to electrify a cheap archtop when I first learned to play guitar. I obtained two "lipstick" pickups from the nearby Danelectro factory and cut holes in the top of the guitar (and some of the bracing) to mount them. The guitar howled from 60-Hz hum and acoustic feedback when I plugged it into my small guitar amplifier. The feedback may even have been damped somewhat by the heavy layers of Testors Candy Apple Green plastic model spray paint I had applied to the body.

My next naïve indignity on the now-mangled archtop was to try to make it into a bass guitar using the Guild humbucker bass pickup and bridge that I purchased from a long-forgotten commercial supplier. I filed notches in the nut to enable four bass strings to reside in the space once occupied by six guitar strings. I traded some of the resulting fret buzz for loss of intonation by adjusting the truss rod so the fingerboard was seriously concave. The neck eventually folded upward due to the structural damage I created with these modifications. But I digress.

The magnetic pickup market is competitive and diverse. The designs for magnetic pickups are equally diverse. The first magnetic pickups were single-coil, followed by the dual-coil Gibson humbucker. Lemme discusses the many different single-coil and humbucker types available. While "vintage sound" is hyped as highly desirable, the manufacturing and processing quality was highly variable in the early days.

The magnetic pickup electrical transfer characteristic is that of a parallel L-C circuit with the significant wire resistance in series with the inductance. This gives each magnetic pickup its own resonance frequency and high-frequency peaking due to the Q of the coil. The author calls this response peaking the "superelevation" characteristic, and it can be affected by the volume and tone control resistances, the guitar cord impedance, and the amplifier's input impedance.

The author describes in detail the equipment and techniques he has developed to objectively measure the response of guitar and bass magnetic pickups. This gives the reader the electrical parameters needed to optimize the guitar's sound by simple modifications to the guitar's electrical components, cord, and perhaps the amplifier input circuitry.

Lemme also noted measurements of the important parameters of many common guitar pickups (e.g., Fender prints the coil inductance on the bottom of its pickups). These parameters are determined by the number of wire turns on the coil, whether it is a single-coil or humbucker, the flux density and reverse permeability of the magnet, the effect of the steel pole pieces and pole adjustment screws, metal pickup covers and plating, and so forth.

He discusses the linear and nonlinear distortions that are produced by these parameters in conjunction with the interaction of the pickup magnetic field and the vibrating strings, the type of string (e.g., wound or plain) and the physical location of the pickup or pickups in the guitar.

He also discusses the installation of active pickup preamps in the guitar, which can be a very effective method to achieve reproducible sound by isolating the pickup from the control pots, guitar cord, and amplifier impedances. Of course, this active circuitry requires electric power in the form of one or more 9-V batteries.

While there are a variety of commercially available magnetic pickups, section 3.9 in Chapter 3 is devoted to DIY construction and modification of magnetic pickups. DIY repair or rewind makes sense for unusual vintage guitars with no available replacement pickups or for extending the treble response of a top-quality acoustic guitar.

Chapter 4 has a relatively short explanation of the construction, operation, and electrical properties of piezo pickups.

Chapter 5 examines the positioning of a pickup or combinations of pickups on the guitar body. The tone is different for every position on the body, warm at the neck and harsher treble at the bridge. Also, the resonance and impedance will change depending on whether pickups are connected in parallel or series. The harmonics are determined by the location of the pickup poles relative to nodes and peaks in the string vibration.

Incidentally, the Guild bass pickup I attempted to use on the previously mentioned archtop guitar eventually found a happy home on a 1967 Hagstrom bass guitar that I modified after learning a whole lot more about pickup position relative to the bridge and neck.

GUITAR WIRING

Chapter 6, the second-longest chapter, addresses guitar wiring. Lemme explains the advantages and weaknesses of both classic passive circuits and the passive circuits used in modern guitars. He explains why the resistance and connection of the volume control is the most important factor in resonance peak and superelevation.

Next, he presents a number of improved passive wiring circuits that enable the tonal qualities of the pickups to come into full effect. Good sound is a matter of individual taste, and the author's ideas are meant to encourage experimentation. Modifications range from extensive custom changes to those that retain the instrument's original appearance, and are easily restored to the original circuit design (assuming you had the foresight to keep the removed components).

Active electronics are a necessity with piezoelectric pickups. The author delves into active pickup design, which has the advantage of more tonal variety corresponding to state-of-the-art electronic design.

Lemme offers a number of DIY active electronic designs, and finished units and kits are available commercially. Active controls take up more space and require a battery compartment. He discusses the desirability of changes that can be completely undone without a trace. He also discusses good grounding and shielding techniques and describes how to solve hum and noise problems in both active and passive pickups.

Finally, Chapter 6 ends with a discussion of wireless RF transmitters.

Chapter 7 briefly covers guitar synthesizers. Organ guitars were the predecessors to modern analog and digital guitar synths. While these devices all produce a wide variety of synthesized sounds, the "feel" of a guitar is hard to imitate and can be lost. Some manufacturers have developed modeling synthesizers that use samples of real instruments to get closer to the original sounds of vintage and modern guitars and basses.

FEEDBACK

Chapter 8 discusses feedback in electric guitars. The sensitivity to feedback is very low with solid-body guitars and increases with semi-solid, semi-acoustic, archtop, and flattop classical guitars, respectively. Feedback can occur due to microphonic magnetic pickups, magnetic fields from speaker magnets or power transformers in the amplifier, or pickups that are poorly fastened to the instrument. Sometimes it is desirable in rock music to have some sustain. The author also reviews electronic sustain devices (e.g., the Sustainiac and E-Bow).

MODIFICATIONS

Chapter 9 is entitled "The DIY Electric Guitar." Lemme describes how to repair electronic guitar components and how to perform the difficult removal and reinstallation of pickups, controls and jacks in semi-acoustic guitars. (I would love to have had this information when I wanted to replace a noisy tone control in my Gibson ES-335.)

He also discusses changing out pickups, improving pickups (including making entire pickups from scratch), and installing additional pickups on your guitar. He does not cover repairs to wood, neck truss rods, varnish, or hardware in this book. Lemme feels there are other good books that specialize in these topics.

DIY DESIGN

Finally, Lemme discusses making your own electric guitar. There are unfinished body/neck kits available, or you could use the neck and body from donor instruments (not necessarily from the same one). The solid-body guitar project is the easiest to undertake. The pickups, hardware, and electronic components are readily available. He describes the solid-body 12-string guitar he made from a reshaped Framus body, Ibanez neck, and custom-designed electronics.

GUITAR COLLECTIONS

Chapter 10 discusses guitar collecting as a passion, and how to select a suitable used guitar or bass. The high prices of older collectible guitars have kept many guitarists from being able to afford the guitar they might want. But over the past few decades, these high prices have also driven up the quality standards for new guitars.

The machine-wound pickups on new guitars are uniformly consistent. The hand-wound pickups on older guitars vary widely in number of turns, inductance, magnet strength and polarity, and winding consistency. As a result, one older guitar may sound radically different from another "identical" guitar due to pickup differences, as well as the amount of usage, and exposure to extreme temperature and humidity conditions. There is also the danger of fakes or altered units made to appear to be real classic collectible guitars or basses.

Chapter 10 also provides an extensive discussion on the mechanical and electrical test you need to perform before buying an instrument.

Chapter 11 presents tips for buying. You have to try a guitar before you buy it. This makes buying by mail order or over the Internet riskier, and Lemme does not recommend it. Playability and good sound are the most important factors in buying a guitar. Famous brand names are not always a guarantee of quality, and a no-name instrument may have the high quality you seek. Ignore advertising and sponsorship, any reference to "vintage," or fluctuations in the value of the instrument you ultimately choose to own.

THE BOTTOM LINE

I found this to be a fascinating and valuable book. The author is an experienced electronics professional and an active musician. He has thoroughly tested everything described in this book, in practice. In my opinion, the index is a very important part of any book, and a generous 15 pages are devoted to indexing. This book is highly recommended! *aX*

Loudspeaker Expert Focuses on Design

Hughes credits his success to knowledge, contacts, friendships, and his wife.


Charlie Hughes is shown in his home work space.

SHANNON BECKER: Tell us a little about your background and where you live.

CHARLIE HUGHES: I've been involved with designing loudspeaker systems professionally for about 25 years. Prior to that, I worked mixing and recording live music in high school and a little bit in college. I earned a Physics degree from the Georgia Institute of Technology. I was very fortunate to have studied under Dr. Eugene Patronis while I was there. I currently live in Gastonia, NC, about 20 minutes from Charlotte.

SHANNON: How did you become interested in audio electronics?

CHARLIE: My interest isn't so much electronics as it is loudspeakers. Of course, I have an interest in amplifiers, DSP, mixers, and the like. However, this is secondary to loudspeakers, how to measure their performance, and how to design them to sound good.

SHANNON: Describe your career as a loudspeaker design engineer. What are some of the highlights?

CHARLIE: I started working at Peavey Electronics in Meridian, MS, right after graduating from Georgia Tech. I was there for almost 14 years. After I left Peavey, I spent a short time at Altec Lansing in Milford, PA, as the chief engineer for its pro audio division, which the company was trying to resurrect. In late 2004, when I left Altec, I started Excelsior Audio, a consulting and loudspeaker design and measurement company working primarily with loudspeaker manufacturers.

I did a lot of horn design at Peavey, being the primary horn guy while I was there. I received my first patent (6,059,069) for a horn design concept. Having that patent issued was really cool. I also presented my first Audio Engineering Society (AES) paper on this type of horn design. I remember seeing Don Keele and Earl Geddes in the audience as I began my presentation. That was a bit intimidating at the time, although I had worked with both of them previously.

In 2000, I was asked to be an instructor at a TEF Level II workshop along with Don Eger and Russ Berger. That was a thrill because those guys are acoustical heavyweights! I knew a lot about the TEF platform and time delay spectrometry (TDS) measurement as applied to loudspeaker systems, so I guess I was able to hold my own.

Around the same time, I was programming MATLAB code to help optimize the crossover design for loudspeaker systems. This was mainly done in an effort to make the directivity response more consistent through the crossover region and to reduce the time in doing so by eliminating lots of repeated polar measurements. The resulting program, PolarSum, was very useful in this regard (www.excelsior-audio.com/Publications/Alternative_Ways_of_Viewing_Polar_Data_%28SAC_tt013%29.pdf).

More recently I was asked to join the Ahnert Feistel Media Group (AFMG) based in Berlin, Germany, to consult on its software and help with advanced tech support and product training globally. This is the company responsible for the acoustical modeling program EASE. It also produces a couple of measurement programs, EASERA and SysTune. One of its other programs is SpeakerLab. This is an amazing program that, while doing some of the same things as PolarSum, takes loudspeaker modeling and design far beyond what anyone else I am aware of is doing in terms of directivity response.

Being asked to speak as a panelist for two State of the Art of Live Sound Loudspeaker Design seminars at AES conventions in 2008 and 2009 with Tom Danley (Danley Sound Labs), Dave Gunness (Fulcrum Acoustic), Aleš Dravinec (ADR Audio), and Pete Soper

(Meyer Sound Labs) was a thrill. In 2010, I was asked to speak at an AES convention workshop with Floyd Toole and Peter Mapp about audio system coverage. To be included in the same company with all these gentlemen was indeed a great honor.

Beginning in 2004, I became the chairman of a CEA standards committee working group, CEA R3-WG1. I'm also active on several AES standards committees.

SHANNON: While working at Peavey Electronics you were responsible for the design and development of many commercially successful products for the musical instrument, cinema, PA, studio monitor, and professional/install markets. Can you share some of the challenges involved with a couple of the most well-known designs? Are they still used today?

CHARLIE: At Peavey, some of the most popular loudspeakers are probably the SP series (e.g., SP-2, SP-3, etc.). I worked on several iterations of a couple of the models in this line. For these, there always seemed to be the issue of improving performance, maintaining reliability, and keeping the cost where it needed to be. Of course, this is probably no different than the challenges facing design engineers at many other companies. There was one instance when it was decided that the horn in the SP line needed updating. I think many of the models were using the older CH-2 at the time. I was tasked with the new horn design (what became the CH-941). Since it would be used across the board on the SP line, I wanted to make sure it was as good as I could make it at the time. There were some challenges to getting the directivity control as consistent as I wanted just above the low end cut-off of the horn. There was some narrowing of the coverage pattern in the midrange. I had to go back and study some of Don Keele's AES papers as well as the work of Cliff Henricksen and Mark Ureda. Getting the secondary flare angles correct near the horn mouth helped with this.

The Q Wave series (later the QW series) was another line where a new horn design was used. This time it was one of the new Quadratic Throat horns. The same horn was used on several of the models in this line. It worked very well. I was to design a three-way system for the line. This required a new MF horn along with a new HF horn that would use a smaller compression driver. The MF horn required a closed-back 6.5" MF driver. It was difficult getting the right driver for this system. I know I had to have made Tom James (Eminence Speakers) crazy with all of the back and forth to get this driver dialed in to get the right performance, low frequency extension, and power handling. He did it though, and it is the heart of that loudspeaker system.

Charlie recently finished designing a speaker for SoundTube. The speaker features a line-array point source (LAPS) design.

SHANNON: What made you venture out on your own and start Excelsior Audio?

CHARLIE: The truth is my wife had a lot to do with it. She had been kicking me in the backside for quite some time to start my own business. She had more faith in me than I did! When I left Altec Lansing in 2004 it seemed like the right time to give it a shot.

SHANNON: Can you discuss your best experience thus far?

CHARLIE: That's a tough one because I've been blessed to have so many. If I had to pick one thing, I would have to say it would be getting involved with Synergetic Audio Concepts (SynAudCon). This is an education consortium started by Don and Carolyn Davis to help raise the level of knowledge in the audio industry. When Don and Carolyn retired it was taken over by Pat and Brenda Brown.

I have learned so much from other SynAudCon members: guys like Jay Mitchell, Dave Gunness, Tom Danley, Bruce Olson, Jim Brown, Peter Mapp, Ray Rayburn, Bill Whitlock, and so many others. Without the knowledge, contacts, and friendships I have made stemming from SynAudCon there is no way I would be where I am today.

SHANNON: What was your first personal project? Is it still in use?

CHARLIE: Wow, time to go way back. I think the first personal project I did was the loudspeaker system I designed and built for my senior design project at Georgia Tech. A rather substantial project was required to graduate. Mine was a three-way loudspeaker system. I thought they sounded fairly good at the time, but they looked atrocious! Woodworking was not one of my accomplished skills at the time, nor is it now. I like to think I design them a lot better than I can build them.

I had these and used them from 1987 until about 2001. I made some major changes (complete redesign and rebuild) to the passive crossovers around 1995 that greatly improved the sound. Several years later, the drivers became too deteriorated and they were not worth trying to salvage.

SHANNON: You wrote an article, "Subwoofer Alignment with a Full-Range System" (*audioXpress*, January 2012). Did you create this project for your personal system? Do you still use the system?

CHARLIE: Not really. This was the result of some work from 2002. I was at a SynAudCon workshop about loudspeaker and acoustical testing and measurement. A question was asked of the instructors regarding the alignment of subs and full-range systems. This sparked a, shall we say, enthusiastic discussion amongst a couple of the instructors. However, the question never really did seem to get answered. I spent a lot of time thinking about this topic and what the right answer should be. By investigating the behavior of individual low-pass and high-pass filters, as well as their summation, I think I found the answer.

I think this is really only an issue when the distance between the subwoofer and full-range system is quite large acoustically (i.e., in excess of one-quarter wavelength in the crossover region). Otherwise, there is not enough phase difference to adversely affect system performance (home systems). In large concert PA systems this distance can be quite large and the cause of substantial problems. I'm hopeful that more people will implement my method and find it to be a useful solution. A video of a talk I gave on this can be found on YouTube (www.youtube.com/watch?v=yoASUFGPWwg) and a PDF of the slides downloaded from my website (www.excelsior-audio.com/Publications/AES129_RH_Charlie_Hughes_Subwoofer_Alignment_with_a_Full-Range_System.pdf).

SHANNON: Are you currently working on any audio projects? If so, could you tell us a little about them?

CHARLIE: Yes, I finished the design work not too long ago for a project I did for SoundTube. This was a three-way fixed, single-box line array with an add-on low frequency directivity extension (LFDE) box. One of the interesting things I was able to do with this design is to frequency shade each pass band (except the HF) to maintain fairly constant vertical directivity control (with respect to frequency) as well as have a fairly short extent of the near-field across the entire operating range of the loudspeaker. Neither of these useful performance aspects is typically embodied in a line array. It is essentially a line array that performs as a point source. Thus, we dubbed the design methodology LAPS (line-array point source).

Another project that is nearing completion for Bag End is a line of three two-way loudspeakers, each with a 10", 12", and 15" woofer, respectively. I designed the high-frequency horn, the vented enclosure for each, as well as the internal passive crossovers. We employed a passive all-pass filter in these crossovers to help time align the output from the LF and HF pass bands. I've also developed DSP filtering for use as front-end EQ as well as for driving the loudspeakers in a biamp configuration.

An interesting aspect of the EQ used for these loudspeakers is that it is based on the frequency response at multiple listening positions, both on- and off-axis. By post-processing many off-axis measurements we were able to better determine what frequency regions would benefit the best from EQ and which ones to leave alone. Equalizing a loudspeaker based solely on a single frequency-response measurement

Charlie recently designed three two-way loudspeakers for Bag End. Each has a 10", 12", and 15" woofer.

typically will not yield results as good as this method.

There are some other projects currently in the works but they're not close enough to completion to be able to discuss them.

SHANNON: What do you see as some of the greatest audio innovations of your time?

CHARLIE: I think the TDS measurement method by Richard Heyser was a great innovation for loudspeaker measurement and analysis. There are others methods, such as dual-channel FFT using a log-swept sine stimulus (Angelo Farina) that are also useful.

The realization that the "sound" of a loudspeaker has more to do with its off-axis radiation than the on-axis frequency response is an unassailable truth when listening to loudspeakers in rooms, but it is amazing how many people still don't get it.

The use of FIR filtering in the DSP available today to make loudspeakers more accurate in terms of the reproduction of the input signal to the system is also a great advance.

SHANNON: Do you have any advice for *audioXpress* readers who are thinking of building their own sound systems?

CHARLIE: Read Floyd Toole's book *Sound Reproduction, Loudspeakers and Rooms*, (Elsevier, 2008). Pay just as much attention to the off-axis response of loudspeakers as the on-axis response. Get the directivity response of the system as consistent as possible across as wide of a frequency range as possible. Avoid abrupt changes in the directivity response. These are most likely to occur through the crossover region. Also, get it right in the time domain. I think that, audibly, this is more important than the frequency domain.

And, have fun! If you don't enjoy what you're doing, it's time to find something else to do. *aX*

 By Richard Honeycutt (United States)

Power Supplies for Hollow-State Equipment (Part 3)

A Power Supply Design

A sample power supply design demonstrates previously discussed concepts

It's time to work through a power supply design example to tie together the ideas we've discussed in the preceeding two articles. The first step in any design is to set specifications. Let's design a supply for a 50-W guitar amplifier with a sort of "hard southern-rock" type of distortion. A pair of 6550 beam power pentodes as output tubes in an ultra-linear circuit, combined with a fairly stiff power supply, will help us achieve this sound. By "fairly stiff," I mean the B+ voltage regulation will be 10% or better, as the current varies from minimum to maximum. We will also design for total hum and noise to be at least 70 dB below maximum output. This power supply will also be suitable for one channel of a home-stereo amplifier.

TUBE SELECTION

Next, we need to calculate the current that must be provided by the B+ and filament supplies. Then, we'll be able to calculate the capacitor and inductor values.

We will populate our amp with the tubes shown in **Photo 1**. **Table 1** details each tube's function and quantity. Checking a tube manual, we find these tubes will require the voltages and currents shown in **Table 2**.

Photo 1: These are the tubes used in the amplifier whose power supply is discussed in this column.

We will connect the filaments in parallel. To do this, we will feed the 7025s' filaments by tying the ends together and connecting the filament supply from center to the two ends. **Figure 1** shows this wiring arrangement.

The 6550s will be operated in Class AB1 to prevent crossover distortion, which means the tubes don't cut off at the signal's zero-crossing. Instead, they are biased, in this case, for a 60-mA plate current at the zero-crossing (per the datasheet at www.dougstubes.com/6550-tungsol.pdf). The maximum signal plate current is about 190 mA. When one of the output tubes is conducting 190 mA, the other is cut off. At zero signal, each is idling at about 60 mA, for 120 mA total. The screen grids will draw another 38 mA or so for whichever tube is conducting maximum current. Thus, the B+ power supply needs to be able to put out 228 mA (i.e., 190 + 38) maximum while staying within its regulation and hum specifications.

Summing the tubes' requirements, we find that the filament supply must produce 6.3 V at 5.9 A. The B+ supply needs to be able to supply 450 V at 228 mA with a 10% safety margin = 251 mA, plus 180 V at 14 mA.

RECTIFIERS

To provide the desired regulation and low ripple, we'll use a full-wave rectifier.

Function	Model	Quantity
Preamp, reverb preamp, reverb return, tremolo	7025	5
Reverb driver, phase splitter	6FQ7/6CG7	2
Power output	6550	2

Table 1: These are the functions, models, and number of tubes used in this amp's design.

Model	B+ V & I	Filament V and I	Total for all tubes
7025	180 V at 2 mA each	6.3 V at 0.3 A each	I_P, 10 mA; filament, 1.5 A
6FQ7/6CG7	180 V at 2 mA each	6.3 V at 0.6 A each	I_P, 4 mA; filament, 1.2 A
6550	450 V at 190 mA + 38 mA screen grid	6.3 V at 1.6 A each	I_P, 228 mA; filament, 3.2 A

Table 2: The individual tubes require specific voltages and currents.

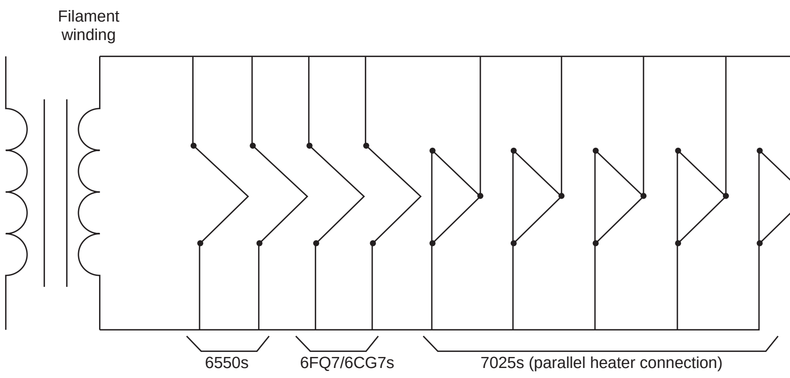


Figure 1: Filament supply wiring is connected from the center to the two ends.

Since we want a fairly stiff supply, not a very stiff one, we will choose a PI filter. A simple capacitor filter would make hum control difficult, but an LC filter would make the supply very stiff, not providing the feel we want for this guitar amp. As far as voltage regulation is concerned, a PI filter can be analyzed like a capacitor filter, with the capacitance equal to C1 + C2 (since they act effectively in parallel). In last month's article, we gave the equation for the RMS transformer secondary voltage for a power supply using a capacitor filter (not accounting for the effect of load current upon the DC voltage):

$$V_{AVG} = 0.636\ V_P = 0.9\ V_{RMS}$$

Solving for the transformer secondary voltage, and remembering that the output DC voltage with no load current is V_P, we have:

$$V_{RMS} = \frac{0.636\ V_P}{0.9} = 0.707\ V_P$$

With V_P being 450 V, the transformer secondary voltage needs to be 318 V (i.e., 450 V × 0.707). If we use a bridge rectifier, we can use a single 318-V winding, but if we use a full-wave center-tapped rectifier, we'll need a center-tapped 636-V winding (sometimes listed as 318-0-318 V) rated at a minimum of 251 mA.

In last month's article, we also provided the equation for the output voltage from a capacitively filtered full-wave rectifier:

$$V_{LOAD_{DC}} \cong V_P - \frac{I_{LOAD_{DC}}}{2\ fC}$$

As mentioned earlier, this equation also applies for a PI filter, provided that the DC resistance of the inductor (or "choke") is accounted for and the capacitors are regarded as being paralleled so that their values add. Guessing at a Hammond P-T156R choke (we'll verify later), the DC resistance should be about 56 Ω, which equates to a 19-V (i.e., 56 Ω × 0.340 A) drop across the choke. Solving for the capacitor, we have:

$$C \cong \frac{I_{LOAD_{DC}}}{2f\left[V_P - \left(V_{LOAD_{DC}} + V_{CHOKE}\right)\right]}$$

Remembering that the charging frequency f for a full-wave rectifier is 120 Hz in the U.S., this provides:

$$C \cong \frac{0.228}{2 \times 120\left[450 - \left(450 \times 0.9 + 19\right)\right]} = 36\ \mu F$$

(With a 10% sag at full-load, a "450-V" supply puts out 450 × 0.9 V.) This would mean the sum of the values of C1 and C2 is at least 36 µF.

CAPACITORS

A well-known supplier of components for hollow-state circuits offers a 32-µF, 500-V dual electrolytic capacitor that should work well for our purposes. The slightly higher value will slightly improve voltage regulation and reduce ripple. We can solve the equation provided last month for the ripple of a PI filter, to find the size of the inductor. But first, we'll need to translate 70 dB below maximum output into percent. We do this by converting –70 dB to a fraction:

$$\gamma_{FRACTION} = \log_{10}^{-1}\left(\frac{-70}{20}\right) = 0.000316$$

This equates to 0.0316% ripple.

$$\gamma\% \cong \frac{1}{32\sqrt{2}\,\pi^3 f^3 R_L C_1 C_2 L} \times 100\%$$

Or we can ignore the "× 100%" conversion factor and simply substitute:

$$0.000316 \cong \frac{1}{32\sqrt{2}\,\pi^3 f^3 R_L C_1 C_2 L}$$

The equivalent load resistance R_L is just the load voltage divided by the load current:

$$R_L = \frac{450\ \text{V}}{0.228\ \text{A}} = 1{,}974\ \Omega$$

Solving for L, we obtain:

$$L = \frac{1}{32\sqrt{2}\pi^3 f^3 \times 1{,}974 \times (32 \times 10^{-6})^2 \times 0.000316}$$

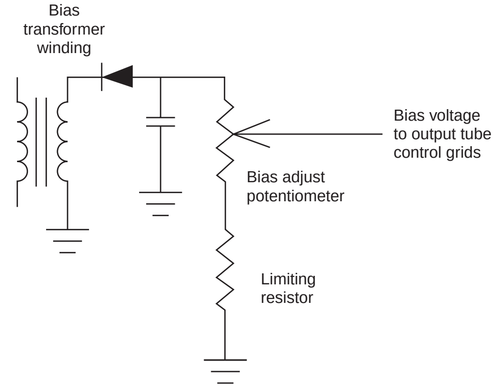


Figure 2: Bias supply topology uses a half-wave rectifier.

We have already entered the values of C1 and C2. This works out to a 0.646-H inductor. A 1.5-H one, such as the Hammond unit mentioned earlier, will be easier to find and will give slightly better performance. Since the low B+ supply only produces 14 mA, the choke can be rated as low as 14 mA. A 1.5-H inductor with so low a current rating may be hard to find. A higher current rating is not needed, but will not be a problem.

RESISTORS

The "low B+" (180-V) supply only has to deliver about 14 mA. We will use a dropping resistor fed from the main B+ supply. This resistor must drop 270 V at 14 mA, so its value must be:

$$R = \frac{V}{I} = \frac{270}{0.014} = 19.3\ \text{k}\Omega$$

We'll use a 22-kΩ resistor, as it is a close enough standard value. This resistor will dissipate $W = V^2/R = 270^2/22{,}000 = 3.31$ W. Normally, a resistor's power rating should be at least twice the average power it dissipates, which would be about 7 W. We'll use a 10-W resistor, since those are easier to find, and we will gain some extra reliability by over-rating it a bit more.

The decoupling capacitor can be chosen from the equation:

$$C = \frac{1}{2\pi f_C R}$$

where f_C is the critical frequency of the RC circuit consisting of the dropping resistor and the decoupling capacitor. *The Radiotron Designer's Handbook* suggests 16 Hz for f_C for an audio amplifier. It also says that a larger capacitor is needed for preamps. Using 1 Hz as f_C is a conservative practice. Then we find the coupling capacitor's value as:

$$C = \frac{1}{2\pi \times 22{,}000} = 7.2\ \mu\text{F}$$

We'll use a 10-µF/250-V decoupling capacitor. Since the power amp will use the ultralinear circuit, the 6550s' screen grids will be fed directly from taps on the output transformer, so no screen grid dropping resistor or decoupling capacitor is needed.

BIAS SUPPLY

Next, we need to design the supply to bias the 6550s. For 60-mA zero-signal plate current, we'll need a –48-V bias. This should be adjustable to compensate for variations among 6550s. Using a capacitor filter, if we choose a maximum bias voltage of –60 V, that means the peak voltage is 60 V, so the $V_{RMS} = 0.707 \times (60\ \text{V}) = 42.42$ V. We can use a half-wave rectifier since very little current is drawn from the bias supply.

To make the bias supply adjustable, we will use a potentiometer, which will be wired in series with a fixed limiting resistor. This arrangement prevents a technician from accidentally setting the bias too low, and possibly damaging the output tubes by excessive plate current. The bias supply topology is shown in **Figure 2**.

The bias filter capacitor can be chosen from the equation for the ripple from a capacitor filter (see the January 2013 *audioXpress* "Hollow State" article):

$$\gamma\% \cong \frac{1}{4\sqrt{3}\, f R_{LOAD} C} \times 100\%$$

where in this case, R_{LOAD} is the sum of the potentiometer and the limiting resistor. A typical value for the potentiometer is 50 kΩ (the value is not critical). If we limit the minimum bias voltage to –30 V (yielding a plate idling current of 210 mA), the limiting resistor should also be 50 kΩ. We will use a 47-kΩ resistor, as that is a standard 10%-tolerance value. The power dissipated by the potentiometer and limiting resistor is given by $P = V^2/R = 60^2/97{,}000 = 0.037$ W for both. We can use 0.5- or 0.25-W resistors,

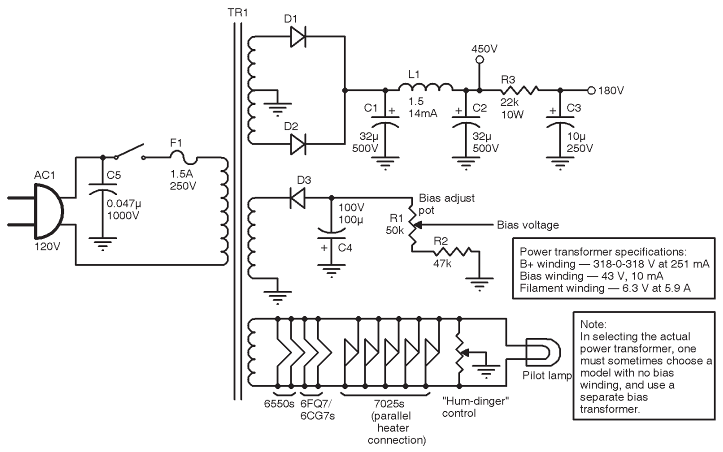


Figure 3: The complete power supply schematic shows the addition of an 1.5-A fuse.

whichever is less expensive. Solving the equation for C, remembering that the charging frequency for a half-wave rectifier is 60 Hz in the U.S., gives us a bias filter capacitor value of:

$$C = \frac{1}{4\sqrt{3} \times 60 \times 97{,}000 \times 0.000316} = 78\ \mu\text{F}$$

We will choose a 100-µF, 100-V capacitor. The 0.000316 value corresponds to the ripple being 70 dB below the bias voltage, as discussed earlier.

To keep the hum low, we will connect a 100-Ω, 2-W wire-wound potentiometer across the filament winding and ground the slider. During initial adjustment of the completed amplifier, we will measure the hum at the output while we adjust this "hum-dinger" potentiometer for minimum hum.

POWER SUPPLY

Figure 3 shows the complete power supply. Notice that we have added a 1.5-A fuse. The fuse's current rating is found by adding the product of the B+ voltage and current to the product of the filament voltage and current (which gave us about 134 W), then dividing this value by the AC line voltage (120 V in this case), and adding a bit more current for the "turn-on" surge. The calculations gave us 1.112 A, so we increased it to 1.5 A. Since we are using a solid-state rectifier, the turn-on surge will be significant, so we'll use a slow-blow fuse.

Also notice the 0.047-µF/1,000-V capacitor connected from the "hot" side of the power line to ground. This capacitor's purpose is to protect against noise and brief voltage spikes on the AC line.

In building a power supply for an audio circuit, you should keep the power transformer as far as possible from critical small-signal circuitry (preamps). Filament leads should be twisted together and routed as close as possible to the chassis, and as far as possible from leads carrying small signals. Ground leads should be kept as short as possible, and connected to the chassis at a single point, not bunched up on a turret or circuit board "land" and then run to the chassis through a long wire.

In next month's "Hollow State" article, we'll discuss troubleshooting power supplies. *aX*

MEMBER PROFILE

Member Name: Garth Wasson

Location: North Vancouver, British Columbia, Canada

Education: Garth has a BA and an LLB from the University of British Columbia and a certificate from King's College London University, London, England.

Occupation: Retired barrister and solicitor (a lawyer)

Member Status: He has subscribed to *audioXpress* since 2005.

Affiliations: He currently holds no official memberships with any audio organizations.

Audio Interests: Garth said he is most interested in classical music.

Most Recent Purchase: A digital multimeter that measures inductance and capacitance

Current Audio Projects: Garth said he has just completed two 100-W, push-pull El34 amplifiers with output transformers that Doc Hoyer designed and built.

Dream System: He said the system would include a couple 100-W amplifiers with a good preamp, a good turntable, and Klipsch horn enclosures. He would also add a quality FM tuner, digital inputs, and a large TV with good sound input. Garth said he doesn't care too much about surround sound, but he admitted that he does have a couple 50s he could use with his dream system. "Obviously, I would want two more Klipsch horns there, too," he added.

Garth Wasson

 By Vance Dickason (United States)

Motus Audio Dome Tweeter

A 1” high-frequency dome tweeter

Photo 1: The new Motus UH25CT1 tweeter

This month, I reviewed a new 1” dome tweeter, the Motus UH25CTi, from Motus Audio, a U.S. high-end driver OEM. Motus Audio is new to Test Bench. The company was founded in 2011 by Trevor Ryan who also started Ryan Acoustics. With his extensive experience in Asian sourcing and assembly of high-performance speakers, Ryan brings the high-end-manufacturing-in-China perspective to the OEM raw driver market.

In terms of features, the new Motus UH25CT1 tweeter employs a 26-mm (1”) diameter treated-cloth dome diaphragm, which is a proprietary design sourced from Japan, a 1” two-layer underhung voice coil with 0.3-mm X_{MAX} wound with copper-clad aluminum wire (CCAW) on an aluminum former (see **Photo 1**). This former features six vent holes and is damped with ferrofluid for cooling. Other features include a damped (fiberglass instead of foam) vented and an extended/undercut pole exhausting into a damped rear chamber (also fiberglass damped), a copper shorting ring in the gap area, and a second aluminum shorting ring mounted into the neodymium N48SH ring magnet. There are also eight vent holes beneath the tweeter surround to minimize back pressure applied to the surround. Keeping to the high-end motif, Motus uses Cardus Audio internal wire to go from the voice coil to the gold-plated terminals. Last, both the faceplate and rear cup are die-cast aluminum.

Testing commenced using the LinearX LMS analyzer to produce the 300-point impedance sweep shown in **Figure 1**. The magnetic fluid damped resonance occurs at a low 463 Hz. With a 4.61-Ω DCR, the minimum impedance for this tweeter is 5.15 Ω at 3.3 kHz.

Following the impedance test, I recess mounted the Motus tweeter in an enclosure that had a baffle area of 8” × 10” and measured the on- and off-axis frequency response with a 100-point gated sine wave sweep at 2.83 V/1 m.

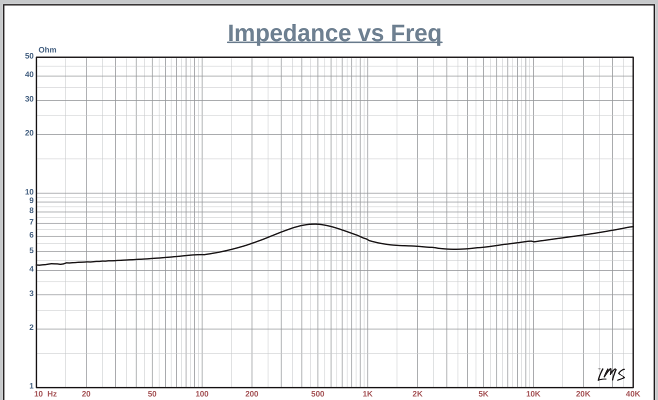


Figure 1: Motus UH25CT1 free-air impedance plot

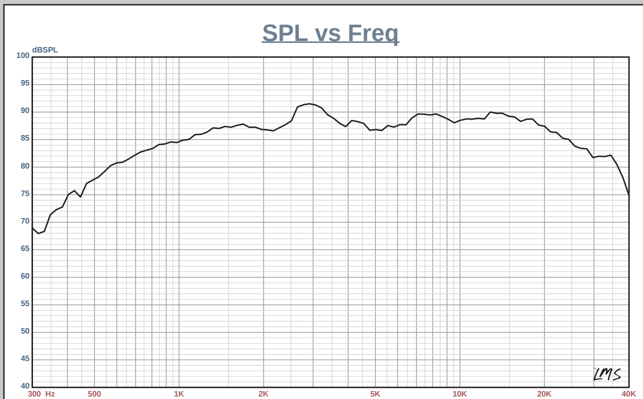


Figure 2: Motus UH25CT1 on-axis frequency response

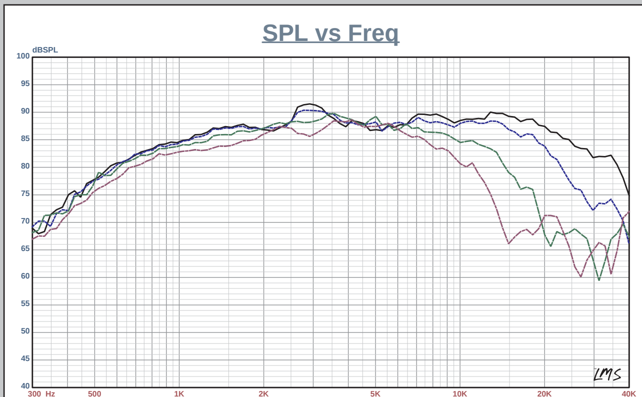


Figure 3: Motus UH25CT1 horizontal on-and off-axis frequency response (0° = solid black; 15° = blue dot; 30° = dash green; 45° = dash/dot purple)

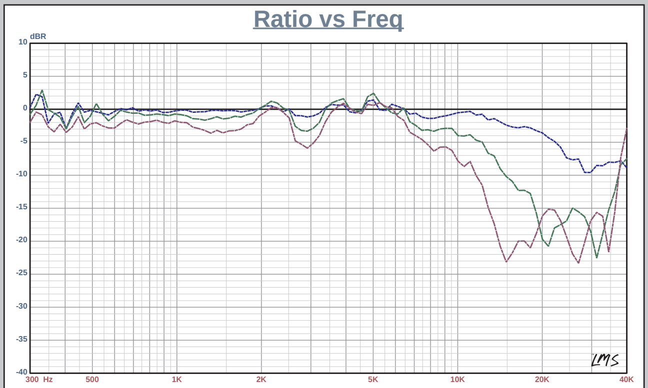


Figure 4: Motus UH25CT1 normalized on-and off-axis frequency response (0° = solid black; 15° = blue dot; 30° = dash green; 45° = dash/dot purple)

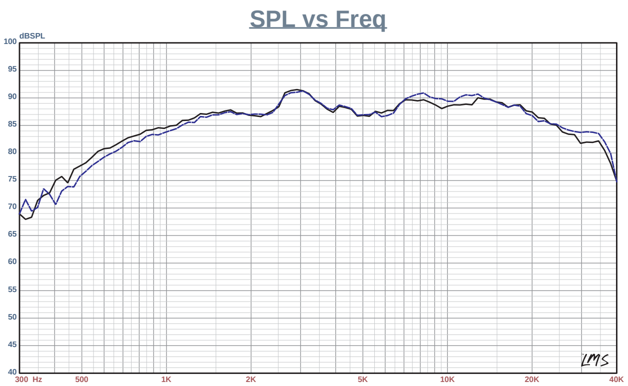


Figure 5: Motus UH25CT1 two-sample comparison

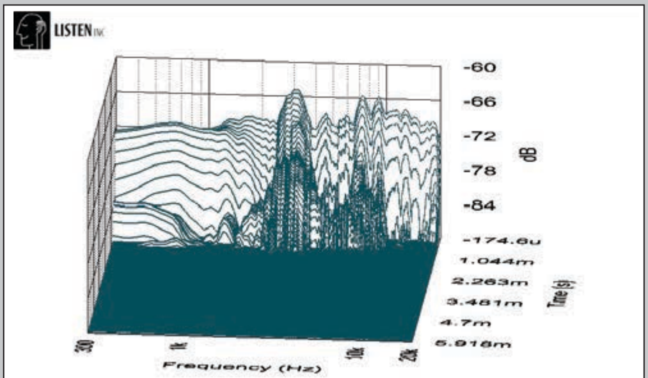


Figure 6: Motus UH25CT1 Soundcheck CSD waterfall plot

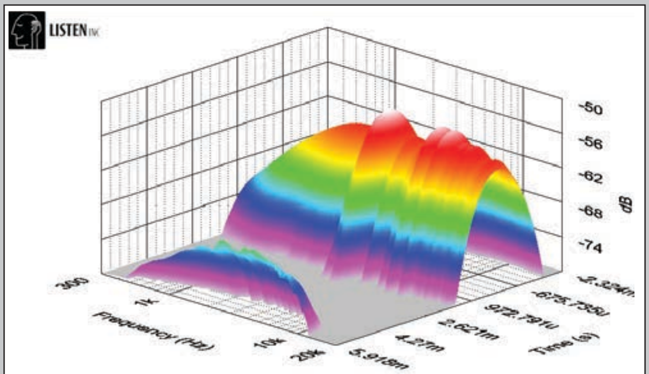


Figure 7: Motus UH25CT1 Soundcheck STFT surface intensity plot

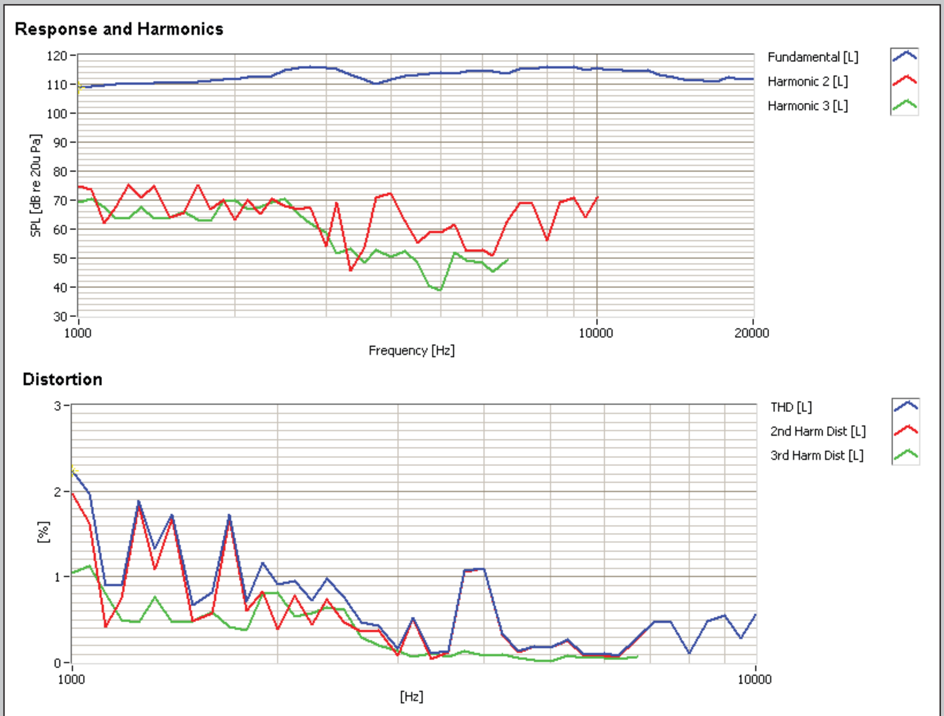


Figure 8: Motus UH25CT1 SoundCheck distortion plot

Figure 2 shows the on-axis response to be a rather flat ±2.2 dB from 3 to 21 kHz, and from 1 to 24 kHz, ±3.2 dB. **Figure 3** depicts UH25CT1's on- and off-axis response, with the off-axis curves normalized to the on-axis response in **Figure 4**. The two-sample SPL comparison is shown in **Figure 5**, indicating the two samples were closely matched, with a small 1-dB variation between 7 and 13.5 kHz.

Next, I initialized the Listen SoundCheck analyzer along with the Listen SCM 0.25" microphone (provided by Listen) to measure the impulse response with the tweeter recess mounted on the test baffle. Importing this data into the Listen SoundMap software produced the cumulative spectral decay plot (CSD) waterfall plot shown in **Figure 6**. **Figure 7** shows a short-time Fourier transform (STFT) displayed as a surface plot. For the final test procedure, I used a pink-noise stimulus to set the 1-m SPL to 94 dB (1.5 V), and measured the second- and third-harmonic distortion at 10 cm (**see Figure 8**). For more information on this and other Motus transducers, visit www.motusaudio.com. *aX*