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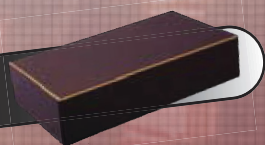


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Conventions, Shows, and Learning

Exhibitions are great ways to meet people, do business, and actually learn something. Instead of attending sessions full of people where someone spends an hour explaining a product or technology with PowerPoint, we can go to the company booth, receive the same explanation plus a working demo, and meet the person who happens to be the developer and tells you all the great things the company is planning for the next product/version.

Industry associations such as the Audio Engineering Society (AES), for instance, promote great sessions and debates during their conventions with knowledgeable professionals, not just professional "lecturers." Associations such as the AES have promoted some truly historical events with unique industry legends and personalities and its role cannot be praised enough. But the AES tends to regard exhibitions as a "necessary evil." In reality, AES exhibitions are great and the organization needs to promote a better integration between sessions and the show floor.

In contrast, the National Association of Music Merchants (NAMM) promotes one of the longest-running shows of any industry. Every year it does a great job of conveying a sense of "being in the right place, doing the right thing." Sessions and events mostly take place in the exhibit halls and special events happen early in the morning or after show hours. The NAMM Show is one of the best examples of a great trade show, combined with learning opportunities that try to enrich the event itself by avoiding diverting participants away from the show floor.

Typical "trade show" promoters, such as the German "Messess" and in certain ways the broadcast associations such as NAB and IBC, and the InfoComm, the Custom Electronic Design and Installation Association (CEDIA) or Integrated Systems Events (promoters of ISE in joint venture with CEDIA and InfoComm International), organize really great exhibitions that also happen to have a few "sessions," which are mostly social engagements to celebrate and network with a few keynote speeches for the daily show newspaper. Their contribution to the industry is insurmountable and should be cherished.

But none of these kinds of organizations can do the entire job and provide the much-needed learning materials, because that's not their specialty. Just look at the *AES Journal* or *The Journal of the Acoustical Society of America*. All the best content is there, but it's just a painful repository using 19th century printing templates, which no one is actually able to read. And that's why, they should partner with media companies for those things.

Successful industry events, such as recent AES US conventions, or the International Consumer Electronics Show (International CES), produced by the Consumer Electronics Association (CEA), are the result of the right combination of partnerships, retaining a role for independent media and publishers.

Industry associations need revenue to promote shows and the sessions. To correctly promote a convention and a series of conferences, you actually need sponsors and sustainable industry support. For that, you also need the right balance and the role of media partners. They should not try to absorb each other's roles, focus exclusively on what happens in conference sessions, or just sell square meters in shows.

All the best conferences, with a continuous program of keynotes, workshops, seminars, debates, and so forth provide better learning opportunities if there is also a magazine article reinforcing all the pertinent details. You could save time by just reading the articles, but you never would if it wasn't for the fact that the media has learned about it due to the event promotions.

When a trade show, which is supposedly promoted by an industry organization, promotes a few seminars and practical workshops and they own and publish a glossy monthly magazine with great photography but few actual articles you want to read, then you know it's actually a business disguised as an industry association, making money from the show and the ads they sell in their "pretend" publications. This is never a good thing. Most importantly, it removes much needed support for legitimate press and dedicated media, which fortunately still exists in the audio industry.

João Martins
 Editor-in-Chief



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COLUMNISTS

Vance Dickason has been working as a professional in the loudspeaker industry since 1974. He is the author of *Loudspeaker Design Cookbook*—which is now in its seventh edition and published in English, French, German, Dutch, Italian, Spanish, and Portuguese—and *The Loudspeaker Recipes*. Vance is the editor of *Voice Coil: The Periodical for the Loudspeaker Industry*, a monthly publication. Although he has been involved with publishing throughout his career, he still works as an engineering consultant for a number of loudspeaker manufacturers.

Dr. Richard Honeycutt fell in love with acoustics when his father brought home a copy of Leo Beranek's landmark text on the subject while Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical instruments and sound systems. He has been an active acoustics consultant since he received his PhD in electroacoustics from the Union Institute in 2004. Richard's work includes architectural acoustics, sound system design, and community noise analysis.

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Steve Tatarunis has been active in the loudspeaker industry since the late 1970s. His areas of interest include product development and test engineering. He is currently a support engineer at Listen, in Boston, MA, where he provides front-line technical support to the SoundCheck test system's global user base.

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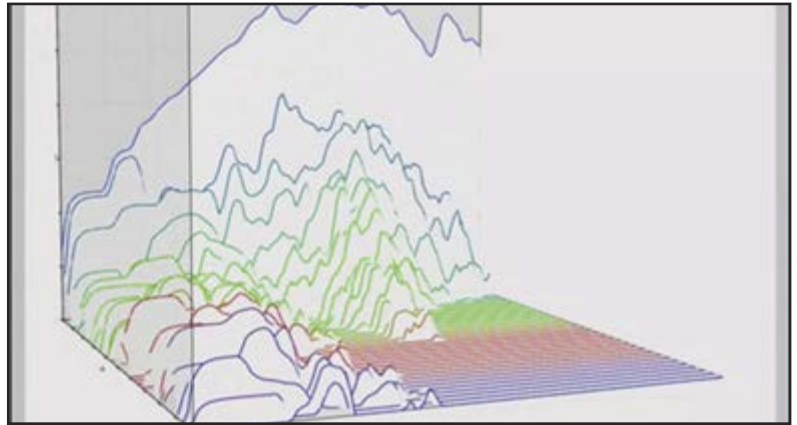
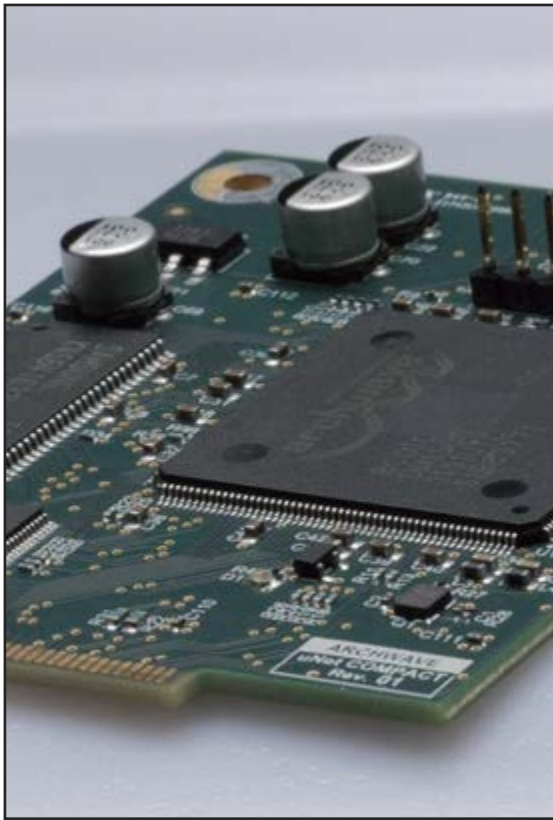
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Sound Cards for Data Acquisition in Audio Measurements (Part 7)

Pitting the CLIO Pocket Against a Sound Card

I admit to being intrigued by Joe D'Appolito's review of the CLIO Pocket—the "junior" version of Audiomatica's well-respected CLIO systems. Until now, my series on using sound cards for audio measurement primarily concentrated on electronics rather than loudspeaker or acoustic measurements, but the current generation of sound cards and software can provide a variety of loudspeaker measurements. So why my interest in the CLIO Pocket, given that I already have three sound cards and four software suites to support them? I was curious.

By
Stuart Yaniger
(United States)

To start, we need to determine the ideal measurement cycle for a loudspeaker. Typically, we set up the software, adjust the levels, hit "Go," and the measurement appears. That's nice and simple in theory, and simple in reality *if* you are a dedicated speaker guy who doesn't fuss much

with anything else and can leave things set up. I look with envy on the photos from folks such as Joe D'Appolito, who have dedicated speaker labs or those who can set up measurements in a living room. Instead, I operate out of my cellar electronics lab.

If you're not one of those fortunate people who can dedicate most of their time to speaker design and construction (and I'm not), then you have to hunt up the loudspeaker, the cables, the power amps, the jigs, an external sound card, a stand to put the speaker on, a test microphone, and a microphone preamp. Then dig out the various speaker-specific bits, set everything up, and verify the operation each time. Oh yes, and re-familiarize yourself with the acoustic end of the measurement software. (For example: When I did this six months ago, I remember using that command, where the devil is it? or I forgot how to load the microphone calibration files, let's read through the manual again, etc.)

This is sadly not trivial since the most versatile software is by definition bloated with rarely used features, which means lots of menus and dialog boxes to try to dig through to find what you want. On top of that, most of the preferred software options seem to originate from outside the US

Photo 1: This is an improvised microphone mount adapter and shock mount made from torn cloth strips, mounted on a flexible tripod. (Photo courtesy of Cynthia Wenslow)



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and England, so you have the triple-whammy of complex operation, manuals written in the turgid and opaque style of software engineers, and somewhat idiosyncratic English in the manual and in some of the menu commands. Warning: The first time performing a measurement (or the first time in a while) means a very steep learning curve looms ahead.

So the time breakdown for doing a

measurement might be several hours or more, with the measurement only taking a few seconds, but a lot of preparation and debugging prior to that. Again, if you do speaker work daily, it's only a one-time inconvenience.

This is where the CLIO Pocket comes in handy. It is for people who want to quickly and accurately take basic measurements, using a self-contained package and a minimum amount of sweat. Does it deliver on this promise? What has been sacrificed in functionality and versatility in the pursuit of convenience? How should someone choose between a CLIO Pocket and a sound card system?

The CLIO Pocket

As D'Appolito explained in his review, the CLIO Pocket includes just about everything you need—a compact USB-connected module containing most of the electronics, test leads with alligator clip termination for impedance measurement, and a dedicated test microphone. You'll still need a power amplifier for measurements other than impedance and near-field frequency response. I used a small LM1875-based chip amplifier, which has more than enough power. You'll also need microphone stands and a way of clipping the microphone to the stand. (Audiomatica sells a kit with the appropriate microphone mounting clip and a microphone calibration adapter.)

Ever the impecunious experimenter, I searched through my junk box and found that every microphone mount I had was too big for the slim-line microphone provided. So with my usual fine workmanship, I ended up improvising a shock mount from a standard-sized microphone mount and a strip of cotton (**see Photo 1**). It might be even better than the "official" microphone mount. For convenience, I used two different microphone stands—a standard boom stand for 1 m and far-field measurements and a small flexible tripod (shown in the **Photo 1**) for near-field measurement.

One note: Audiomatica recommends that the CLIO Pocket unit warm up for 15 to 20 minutes before calibration. They aren't kidding—you'll get "Calibration failed" warnings until things are good and warm, which can be disconcerting when someone (I'm not mentioning any names) is careless about reading the directions.

The Sound Card

Many of the good general measurement software packages have loudspeaker measurement capability. For my comparisons, I used ARTA (the registered version) along with an inexpensive USB

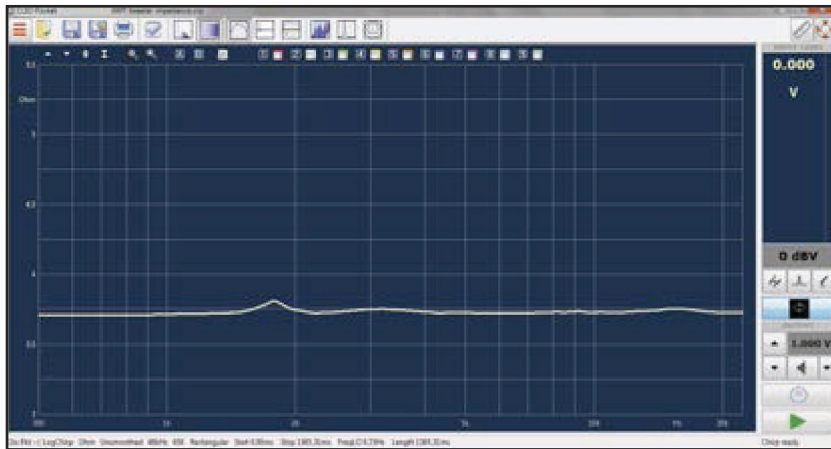


Figure 1: The screen shot of the impedance magnitude of AMT tweeter was acquired via the CLIO Pocket. This is a remarkably flat impedance curve.

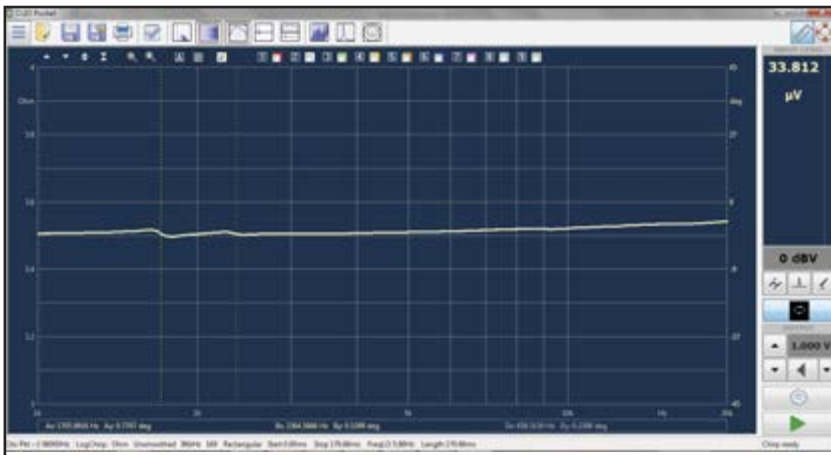


Figure 2: The screen shot shows the impedance phase of the tweeter measured in Figure 1 and simultaneously acquired.

About the Author

Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently works as a technical director for a large industrial company. His professional research interests have spanned theoretical physics, electronics, chemistry, spectroscopy, aerospace, biology, and sensory science. One day, he will figure out what he would like to be when he grows up.

sound card, the Focusrite Scarlett 2i2. The 2i2 has built-in microphone preamps and a phantom power option, which reduces the number of interconnected boxes needed. Its distortion characteristics are nothing to write home about, especially at high frequencies, but it's functional for loudspeaker work where we generally are not pushing parts-per-million distortion. The folks at ARTA feel that this particular card has inadequate dynamic range in two-channel mode, but that's not a huge limitation for most applications. Also, the examples I've seen from them show frequency response errors on the order of 0.1 dB. I'm not too worried. But if you aren't as dollar-constrained as I am, several more expensive cards will overcome this objection.

Because of the demands of the communications industry for inexpensive but high-quality compact microphones, electret microphone capsules have proliferated. Scott Wurcer demonstrated in a series of articles that with relatively simple interfaces, excellent noise performance and wide frequency response can be readily obtained (see Resources). As a result, speaker designers have a lot of good choices for inexpensive test microphones based on electret capsules.

Ideally, you would want a microphone with traceable calibration, since most software packages can accommodate calibration files to correct the measured frequency response. iSEMcon sent me a couple of excellent test microphones, and I ended up using the iSEMcon EMX7150 (which I will write more about in next month's article), with its supplied calibration file. There are also good inexpensive choices from, among others, Behringer, Dayton Audio, and MicW. If you have the money, Earthworks, Josephson/Gefell, Brüel & Kjær Sound & Vibration Measurement, and PCB Piezotronics offer first-rate laboratory-grade measurement microphones. The iSEMcon measurement microphones sit somewhere in between.

The Cost Difference

The CLIO Pocket sells for approximately \$600. To set up a sound-card-based system with similar capabilities, you'll need a USB sound card (approximately \$150), a test microphone with calibration (approximately \$300), and measurement software (approximately \$100). Figure another \$50 for cables and the price ends up being a wash. You can put together a significantly less expensive package by substituting a different microphone (e.g., the Behringer ECM8000 for approximately \$60) and freeware such as REW, but you're on your own as far as calibration goes. Low-level

measurements are more chancy because of higher microphone noise, and even surprisingly good freeware (e.g., REW) lack the versatility of proprietary packages such as ARTA or (as we will see) the flexibility and ease of use of CLIO's software.

Round One

My current speaker project (which I expect to publish on my website next year) is a mini-monitor, intended for use with a subwoofer. It is based on the excellent 4" woofer-midrange driver used by NHT in its venerable Super Zero design.

The original NHT was a low-cost giant-killer in its day, but had a few performance issues. The most obvious one (aside from the missing bass) was integration between the tweeter and the woofer. With normal program material, the speakers sounded smooth and a little laid back, but with a bit of extra zing at the very top end.

This was quite an achievement for a design that used no crossover on the woofer, relying on the mechanical properties of the cone and the surround to provide the necessary rolloff. The tweeter crossover is a single capacitor. Clearly, this is engineered to a budget, and I say that with admiration—getting a design this simple to sound as good as it does is a lot harder than engineering with a generous budget!

The clue to the areas that need improvement can be found with pink noise excitation. To my ear, the woofer and the tweeter sound completely separate rather than the noise integrating into a unified virtual location. This

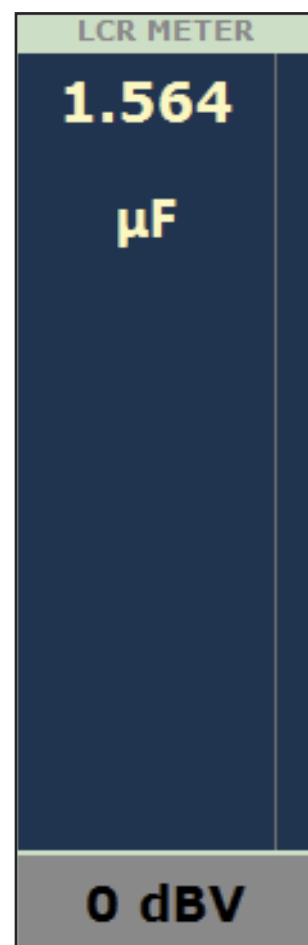


Figure 3: The CLIO Pocket's LCR meter measures a component's inductance (L), capacitance (C), and resistance (R).

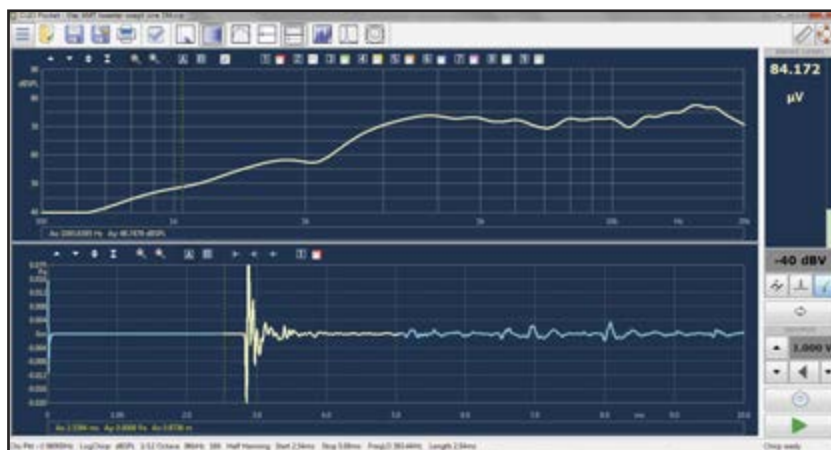


Figure 4: This is the Legacy AMT tweeter's 1 m on-axis frequency response, acquired with the CLIO Pocket, while the tweeter was mounted in an infinite baffle. The lower trace is the gated impulse response derived from the chirp excitation.

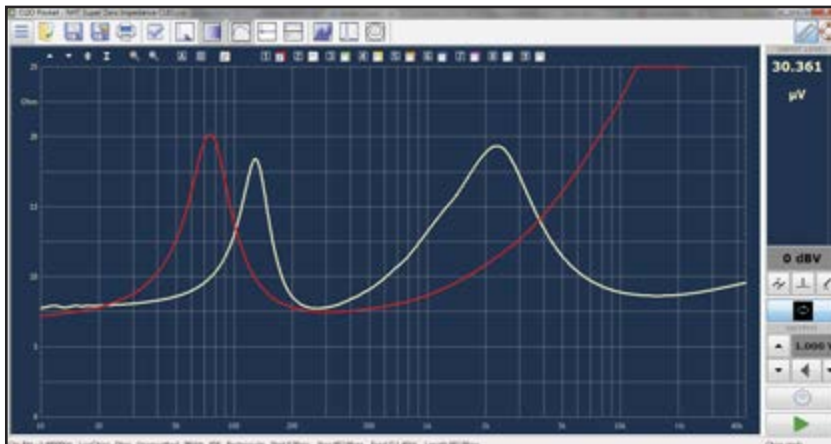


Figure 5: Here is a comparison of free air vs. closed-box impedance for the Super Zero woofer/midrange, acquired with the CLIO Pocket.

usually means that there's a frequency response hole near the crossover or a discontinuity in the polar pattern. So one of the goals in my eventual design is to make that transition as smooth as possible.

Although the woofer/midrange is a fine piece of design work, the tweeter I used was built to a price. This is also reflected in the rather diffuse quality (if I may use that term) in the treble, which contrasts with the microscopically detailed high-end of the Super Zero's big brother, the mighty M3.3. This will be changed in the new mini-monitors.

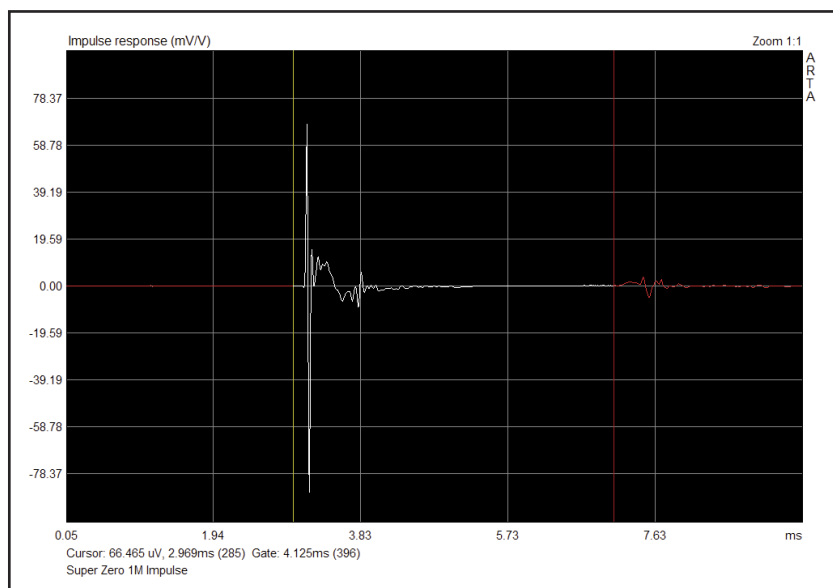


Figure 6: This is the 1 m on-axis impulse response of the Super Zero, acquired with a sound card, ARTA software, and an iSEMcon EMX7150 measurement microphone.

As a candidate for replacement, I recently acquired a surplus pair of AMT tweeters marked "Legacy Folded Ribbon," indicating the company to which they were OEMed. The origin is unknown, but they sure do look like ELAC JET tweeters.

Since these tweeters came with no documentation, it seemed logical to start by characterizing them. In last month's article, "Sound Cards for Data Acquisition in Audio Measurements (Part 6): Examples of Sound-Card-Based Measurements," I discussed how to use a sound card and an Audio-Tester to get an impedance. This time I am using the CLIO Pocket.

Once the unit was warmed up and calibrated (an automatic procedure only requiring a menu button press), impedance measurement setup and acquisition took less than a minute. I connected the cable with clips from the CLIO's output to the driver terminals, selected Log Chirp from the menu bar, set the Y-axis units to ohms, and fired away. Five seconds later, I got the impedance curve (see **Figure 1**). You can see that the impedance swing is extremely small, being within a few tenths of an ohm of 3R7. Likewise, the phase swing is quite small as well (see **Figure 2**). So, this driver can be treated as a constant resistive load for the purposes of crossover design.

CLIO Pocket enables you to export the numerical data for magnitude and phase so that with something such as Excel, you can produce Heyser plots. It would be nice to have that capability right in the CLIO software, but I admit there aren't many of us currently using Heyser plots (although there should be).

While we are discussing impedance, a useful feature is a built-in LCR meter (see **Figure 3**). It's very fast, convenient, and makes matching crossover parts a breeze.

I took frequency response measurements with the tweeter mounted on essentially an infinite baffle made from Styrofoam insulation board. This smooth, relatively dense, and inexpensive material is available in 4' x 8' sheets at most building supply outlets. It is easy to cut holes in, and sturdy and nonresonant enough to support all but the largest and heaviest drivers. For 1 m measurements, the external power amplifier is driven by the CLIO Pocket module. Because I'm a coward, I put a volume control at the input and turned it all the way down when starting an excitation from the CLIO's generator function, then I slowly advanced it while watching the sound pressure level (SPL) measurement until it reached 80 dB SPL.

I measured the frequency response using the Log Chirp option, which is basically a rapidly swept sine wave (see **Figure 4**). The bottom graph is the calculated impulse response, with the gated region (to eliminate the influence of room reflections) highlighted. The top graph is the SPL frequency response calculated via a Fast Fourier Transform (FFT) of the gated impulse response with a half-Hanning window. Note that the rolloff at the low end is consistent with the small impedance peak at 1.8 kHz. The frequency response is reasonably flat, fitting into a ± 2.5 dB window from 3 to 20 kHz.

Given that there's no correction for things such as microphone stand reflections, that's a very satisfactory result. Even more satisfying is the easy and intuitive way in which the measurement was set up and acquired. There was no fumbling around through a thick manual and cursing when I couldn't make the software do what I wanted through my own ignorance.

Likewise, CLIO Pocket makes it simple to acquire impedance and SPL data for the woofer/midrange. **Figure 5** is a comparison of the free-air impedance curve for this driver (in red) vs. the same driver in a Super Zero (in white), which has a volume of about 2 ltr. The driver's fundamental resonance moves from 75 Hz in free air to 128 Hz in the enclosure (the second, higher frequency peak is from the tweeter crossover). There's a provision in the menus to calculate Thiele-Small (T-S) parameters. Q_{TC} for this volume ended up being about 1, which was perfect for integration of a subwoofer by using a second-order high-pass filter with a Q of 0.5 on this driver/box combination, combined with a fourth-order Linkwitz-Riley low-pass filter on the subwoofer. I want to emphasize that the time from setup to useful data was less than 5 minutes for these measurements.

Round Two

It's always nice to start a design and modification project with some baselining. So to show what the sound-card combination can do, I conducted some more complete measurements on the Super Zero to see if I could pin down the cause of the minor sonic issues before I attempted to fix them with the redesign. Since D'Appolito discussed the step-by-step method to get quasi-anechoic data using the CLIO Pocket, I thought it would be useful to do the same thing with ARTA and a sound card. Briefly, a quasi-anechoic response is obtained by taking a near-field measurement for the bass, a 1 m measurement for the midrange and

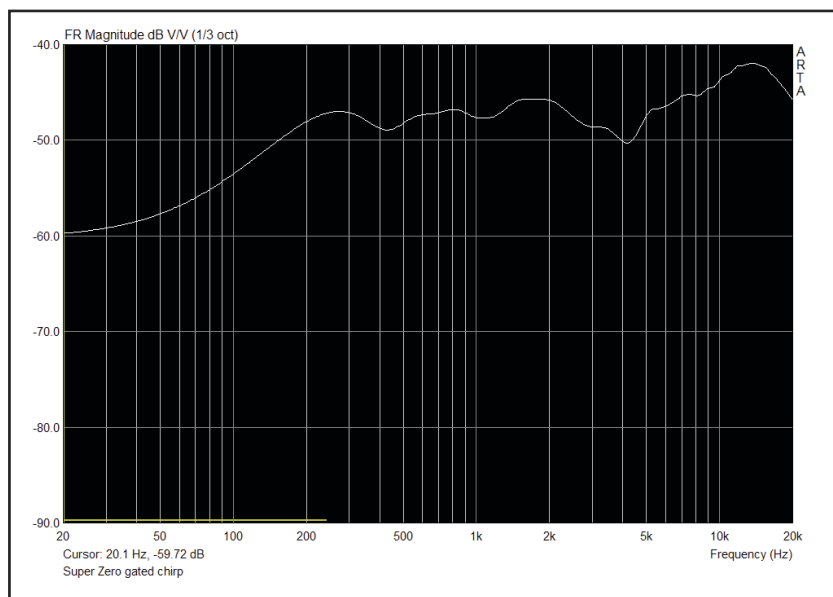


Figure 7: I used the transform of the impulse response shown in Figure 6 to get the frequency response.

treble, picking a point to merge the two responses, adjusting the displayed SPL of the responses to match, then trimming the two responses.

Before acquiring data, I loaded the microphone calibration file supplied with the iSEMcon EMX7150 into ARTA, using the frequency response compensation menu under Setup, a painless

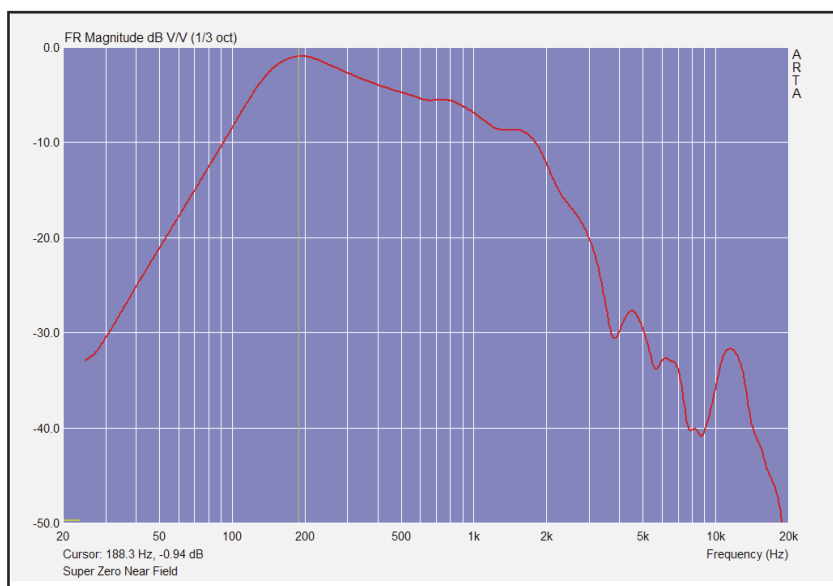


Figure 8: The near-field response of Super Zero was acquired with the sound-card system.

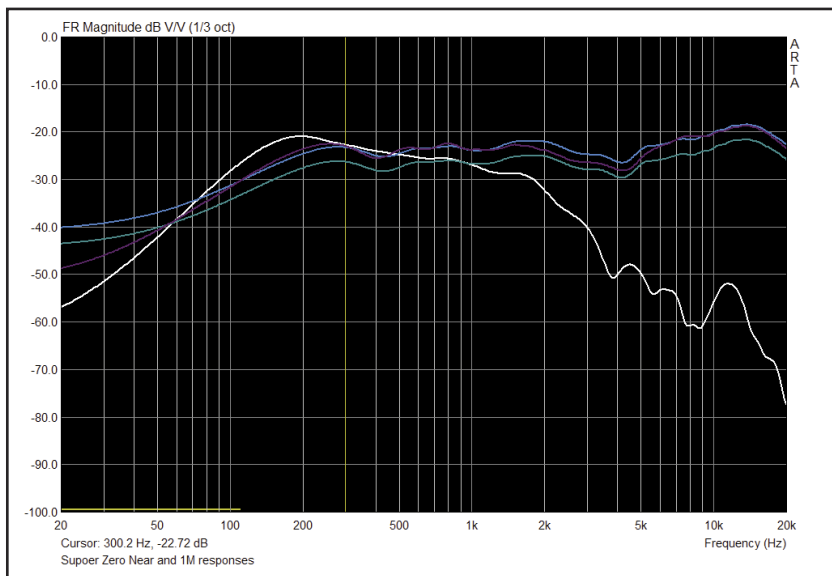


Figure 9: This shows the combined and amplitude-scaled near-field and far-field measurements. The near field curve is shown in white, and the 1 m curves are shown at three different points in blue, green, and magenta.

process. I did not go through the entire ARTA calibration procedure for SPL, so my measurements are relative. When I needed to determine specific SPL, I cheated and used one of two SPL meters that I keep on hand. Sheer laziness on my part, admittedly, but I have to admit that I missed the CLIO automated calibration here.

Figure 6 shows the 1 m on-axis impulse response. I chose gating to eliminate the influence of room reflections at the expense of the low

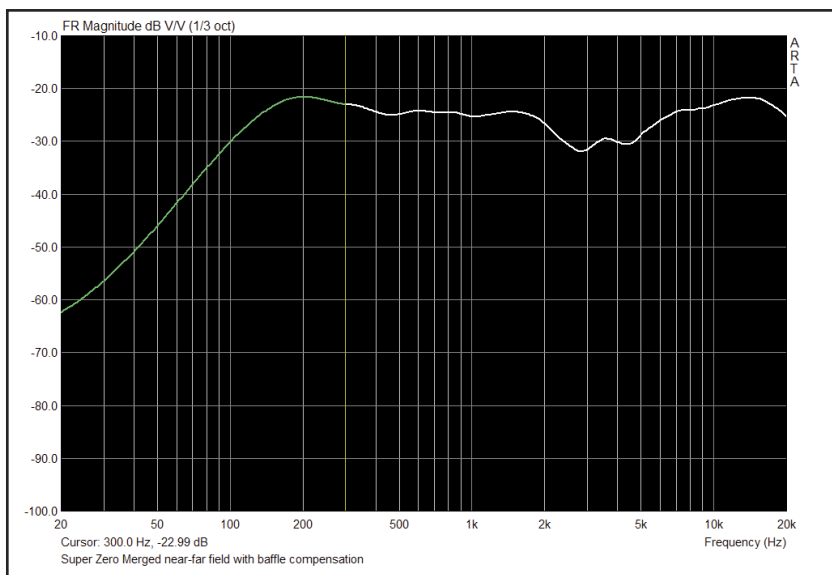


Figure 10: The quasi-anechoic frequency response was obtained from trimming the combined responses and using ARTA's baffle compensation feature.

frequency limit of the measurement. **Figure 7** shows the transformed frequency response.

Figure 8 shows the near-field measurement of the Super Zero. There is a nifty baffle compensation calculation feature, found (non-intuitively) in the Edit options. This function transforms infinite baffle or near field measurements into free space response for the actual baffle size. This useful feature is not found in the CLIO Pocket.

Next, the near-field and far-field measurements are combined using overlays (see **Figure 9**), then trimmed (see **Figure 10**) to yield the quasi-anechoic response. And it's pretty clear where that disconnected quality in the treble originates: the broad 5 to 6 dB dip centered at 4 kHz. This is most likely a tweeter defect, since it occurs about an octave above the crossover point.

So far, we can pretty much duplicate the CLIO Pocket's capabilities with a sound card, a test microphone, and software. But the versatility of general measurement software enables us to get quite a bit of additional information, if we're willing to expend the effort. There's a remarkable variety of esoteric measurements available in general software packages, but for the purpose of this article, I'll limit myself to just a few more common functions.

One interesting feature of ARTA is the ability to calculate group delay as a function of frequency (see **Figure 11**). The tweeter acoustic response is seen to lag the woofer by about 0.2 ms, which is well below what we can detect by ear. And, the group delay curves are satisfyingly flat. Perhaps this is one reason for the outstanding imaging of these speakers?

Another major contributor to the imaging of speakers (as well as their in-room tonal balance) is the polar pattern or directivity. The CLIO Pocket has a tantalizing button on the menu bar for polar plots, but it is not mentioned in any of the documentation, and the button appears to be inactive. No worries, we can do polar plots in ARTA.

My physical setup was rather crude—a kitchen Lazy Susan and a protractor. The speaker was measured on axis, and at $\pm 22^\circ$, $\pm 45^\circ$, and $\pm 90^\circ$. Because of this rather hastily assembled setup, my plots were not a model of perfect symmetry. More data points would have given me better resolution, but they still gave me the basic information I needed. Once the response data are acquired, the polar pattern can be graphed in several ways (e.g., waterfall, contour plots, or polar diagram). I chose the polar diagram, and the ARTA software enables me to select the frequencies for display.

Figure 12 shows this plot and gives another clue to the “disconnected” treble. Looking at the 4 kHz contour, corresponding to the dip in on-axis response, we can see that the dip fills in off axis. This sort of directional behavior can flatten the speaker’s power response, and thus, the overall perceived tonal balance, but can potentially cause audible issues.

Another nice non-exotic feature of the sound card system using ARTA is the ability to get detailed distortion measurements. The CLIO Pocket system can measure total harmonic distortion (THD) at spot frequencies and even give a distortion spectrum, but to get a plot of distortion vs. frequency or distortion vs. level, the data must be manipulated and plotted externally (e.g., using Excel). This feature is built into the STEPS module of ARTA.

Figure 13 shows a measurement of distortion vs. frequency, harmonic by harmonic, for the Super Zero at 1 m on axis at approximately 85 dB SPL. This is done in a single measurement. As expected, the distortion is reasonably low, and rises at lower

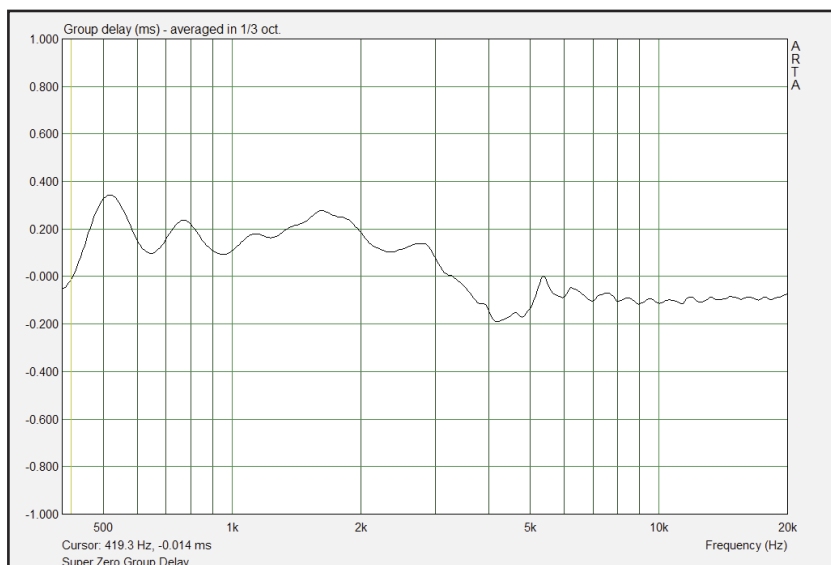


Figure 11: The group delay of the Super Zero was obtained from the gated 1 m on-axis impulse response.

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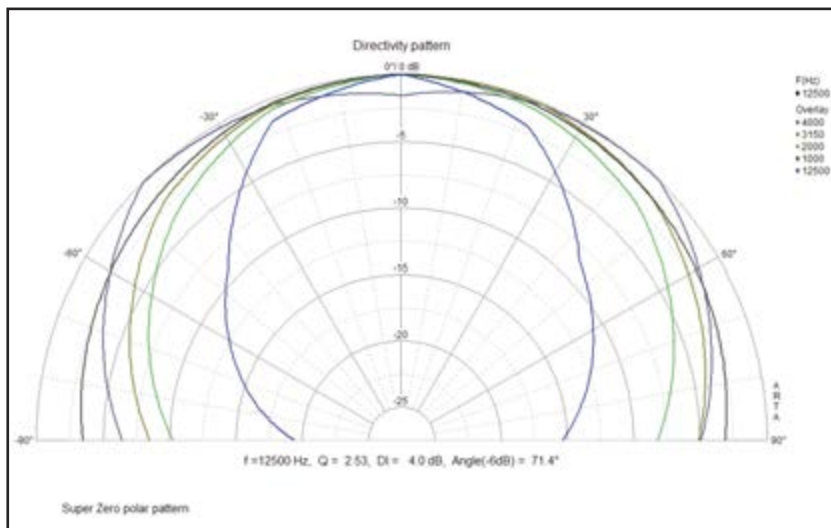


Figure 12: The polar plot of Super Zero directivity from -90° to $+90^\circ$ was derived from seven SPL measurements.

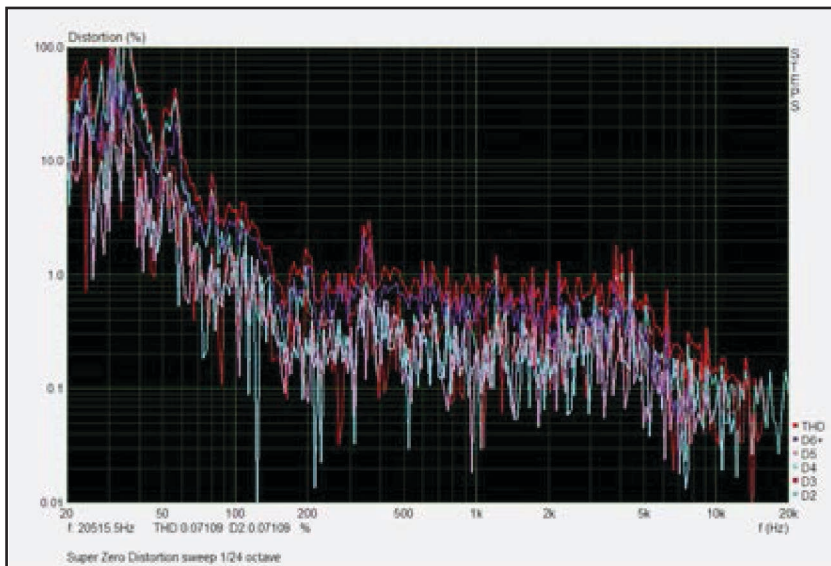


Figure 13: Here is the distortion vs. frequency for the Super Zero, 1 m on axis, 85 dB SPL, acquired by the sound card using ARTA's STEPS module.

Resources

J. D'Appolito, "Audiomatica's CLIO Pocket Electro-Acoustic Portable Measurement System," *audioXpress*, October 2015.

"ARTA User Manual," www.artalabs.hr/download/arta-user-manual.pdf.

"STEPS User Manual," www.artalabs.hr/download/steps-user-manual.pdf.

S. Wurcer, "DIY Low Noise Microphone Preamplifiers, Part II," *Linear Audio*, Volume 3, Winter 2012.

—, "Low Noise Microphone Preamps," *Linear Audio*, Volume 1, Spring 2011.

S. Yaniger, "Sound Cards for Data Acquisition in Audio Measurements (Part 6): Examples of Sound-Card-Based Measurements," *audioXpress*, November 2015.

frequencies where the woofer is being driven to greater excursions.

Since my eventual speaker will likely be crossed over near 120 Hz, I thought that it might be useful to see how distortion varies with level at that frequency. STEPS can perform this measurement with a couple of mouse clicks, and the results are shown in **Figure 14**.

Since I hadn't gone through the full calibration process, the numbers on the X-axis are again relative, but using an SPL meter, I determined that the highest level of the measurement was at 96 dB SPL at 1 m. The distortion there is still fairly low, especially considering the diminutive size of the Super Zero woofer. I attempted to take the measurement to a higher SPL, but the sound inevitably caused my rather sizable dog to start barking loudly, and that woofer was able to overcome the level of the Super Zero woofer, invalidating those measurements.

The Winner By Decision

The company where I work for my day job trains everyone on a set of business practices that it calls a Toolbox. The most important tool in the Toolbox—the primus inter pares (and the one that we are supposed to live and breathe)—is called 80/20. It's a simplification of Pareto's Law and says that for any process, 80% of results come from 20% of effort. The remaining 20% of results requires 80% of the effort.

The CLIO Pocket is a perfect example of 80/20 in action. For 20% of the effort and complication of using a versatile and sophisticated sound card system, you can get 80% of the results that you'd ever need. More importantly, that 80% includes all of the important factors needed for first-class speaker design and characterization: impedance, frequency response, both on-and-off axis, waterfall plots, crossover transfer functions, and THD. It is self-contained except for the power amplifier and is remarkably easy to set up, calibrate, and use. It is limited to a single channel, and tied to a specific test microphone for input (though another source can be used without the ability to calibrate).

For about the same amount of money, you can get an external sound card, a calibrated test microphone, and sophisticated software capable of doing everything the CLIO Pocket does, and a whole lot more (e.g., polar plots, multichannel capability, and distortion sweeps).

If I did speaker design and testing on a frequent basis and didn't usually need to reconfigure things for circuit measurements, this would absolutely be

the way I'd go. In exchange for a one-time steep learning curve, significant setup and calibration time and effort, the avid speaker builder can obtain a tremendous number of measurements, taking it much farther than I did. Once you've done something a few times and continue to practice, it quickly becomes second nature. The major difficulty for a short article was narrowing down the measurement possibilities, which I think is a pretty good problem to have.

On the other hand, for someone like me, who mostly designs and builds electronics but likes to get his hands dirty with loudspeakers on an occasional basis, the CLIO Pocket is a better option. The reduced functionality is of far less consequence than the ability to rapidly and effortlessly get the fundamental design and validation inputs with a high degree of confidence in the accuracy of the results.

In the best of all worlds, I'd have both, and the cost for that isn't terribly steep. 🚗

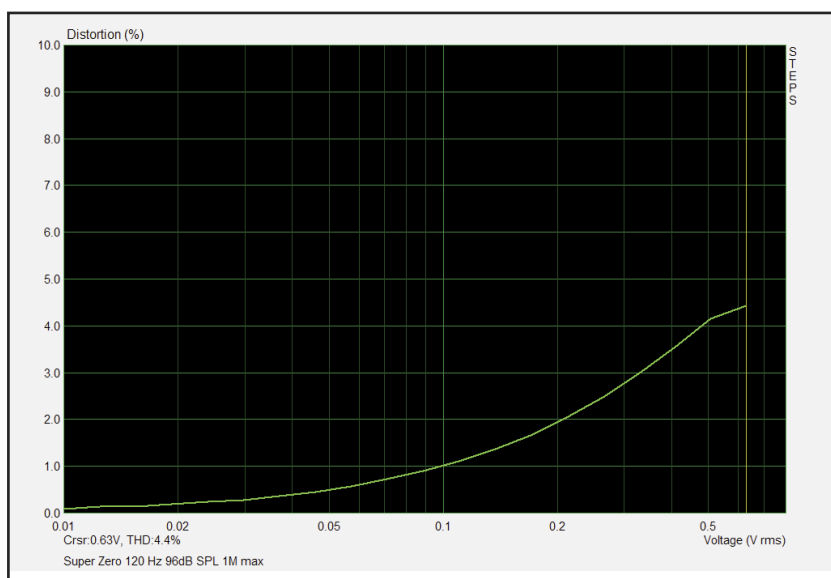


Figure 14: I used the STEPS program to show how distortion varies with the level at 120 Hz. The maximum SPL was 96 dB at 1 m.

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Immersive Sound

A Look at Auro-3D

Realistic sound reproduction doesn't stop with stereo or surround sound—it has evolved into immersive sound. Auro Technologies are the creators of the Auro-3D format; a development intended to move from 2-D surround sound (5.1/7.1) formats to add the missing third dimension of “height” around the listener. This article explores that missing third dimension.



By
Ron Tipton

(United States)

As the name suggests, immersive sound is an extension of surround sound, that is, you can think of your listening position as being located at the lower center of a hemisphere. Of course, your room is probably not a hemisphere but that makes no difference because any location on the hemisphere can be mapped to a corresponding place on your rectangular walls and ceiling. Thus, the sound can potentially come from any direction.



Photo 1: Wilfried Van Baelen (pictured during a presentation at the recent IBC 2015 Show in Amsterdam) is the founder and CEO of Galaxy Studios and Auro Technologies. He also designed Auro-3D's unique three-layered speaker array.

Immersive Sound Systems

There are primarily three competing “immersive sound” systems: Auro-3D, Dolby Atmos, and DTS:X. Each one has different levels of industry acceptance and different consumer setup costs. Auro-3D is channel-based and will be our primary focus for this article. The other two are object-based and embedded metadata directs the sound to the appropriate loudspeaker or speaker array. Dolby Atmos relies on fixed speaker locations while DTS:X is reportedly able to map the sound to any speaker arrangement from 5.1 to dozens of channels.

Immersive or 3-D audio is hitting the headlines with daily announcements relating to cinema, consumer, and broadcast, while the Society of Motion Picture and Television Engineers (SMPTE) continues to debate the merits of object-based sound and has created a working group to define a possible immersive audio standard.

Two of the proposals under discussion include the Dolby Atmos system and the Multi-Dimensional Audio (MDA) format, supported by the MDA Cinema Proponents Group, a consortium that includes DTS, Doremi Laboratories, Ultra-Stereo Laboratories, QSC, Barco, and Auro Technologies. While discussions around an object-based sound standard continues, Dolby Atmos and Auro-3D are being installed in theaters around the world, and they are also starting to make inroads in the home-theater market.

A few other object-based immersive audio proposals include the MPEG-H Audio Alliance TV System, which was jointly-developed by Technicolor and Fraunhofer IIS. It is currently promoted as a streaming media format and TV audio system, under consideration for the ATSC 3.0 standard in development.

DTS is currently promoting its MDA-based DTS:X immersive audio solution, which is mainly for production and new streaming platforms. It is also starting to appear in theatrical releases. Several receivers and processors from Onkyo and Denon are now able to support DTS:X.

Auro-3D

Auro Technologies is a spin-off of the Galaxy Studios Group in Belgium and owner of the Auro-3D Technology Suite. Wilfried Van Baelen, founder and CEO of Galaxy Studios and Auro Technologies (see **Photo 1**), designed the Auro-3D Concept and Listening Formats. The Auro-3D suite offers 3-D audio sound reproduction capabilities to the professional, automotive, broadcast, and consumer electronics markets (e.g., gaming, smartphones, multimedia PC, tablets, audio players, digital TV, media libraries, and packaged media).

Although Auro-3D supports multi-loudspeakers, it is still a channel-based system similar to mono, stereo, 5.1, and so forth. That is, the several audio tracks are in specific channels that are directed to specific loudspeakers or loudspeaker arrays.

Auro-3D uses a “top” and a “height” layer to add realism to surround sound (see **Figure 1**). Auro Technologies claims this adds the third dimension that is lacking in conventional 5.1 and 7.1 systems, which it calls two dimensional. The added layers are said to “make the sound more immersive.”

According to the company’s literature, the “top layer is situated directly above the listener. It can be an array of speakers or a single speaker. The top layer serves to enhance the realism of fly-over effects. The height layer is positioned approximately 40° above the lower layer and is the most important layer in the surround “sphere.” The height layer captures the ambient reflections normally experienced in this layer and enables Auro-3D to produce a more natural sound while improving sound source localization. The lower layer, which occurs between 0° and 20° produces the important 2-D horizontal plane where most ear-level sound sources are located in the mix.”

Auro-3D can be implemented to capture (record) 3-D sound, convert conventional 2-D formats to 3-D, and to play Auro-3D formatted movies and audio Blu-ray DVDs.

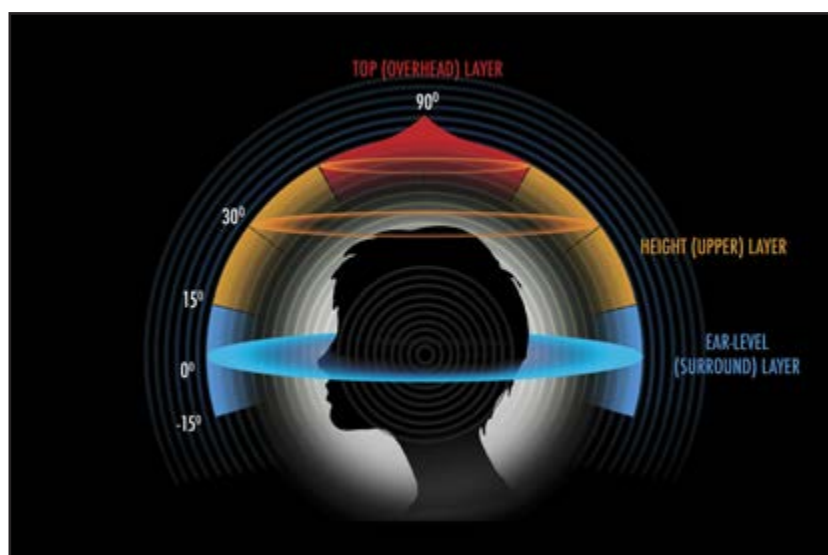


Figure 1: The three Auro-3D recording and playback layers are shown here with their elevation bands.

3-D Recording

Photo 2 clearly shows the three recording levels. In one of the tutorial videos on the Auro website (www.auro-3d.com/videos, “Working with the Auro-3D Format in Music Recording & Production”), Patrick Lemmens remarks that he likes to use one microphone for each playback loudspeaker. Therefore, the crew pictured in **Photo 3** is probably recording in 10.1 format: one top level, four height levels, and five lower levels with the subwoofer or low-frequency effects (LFE) channel being created from the other channels as needed.

The height level microphones appear to be arranged in a square array. This suggests that even



Photo 2: The Auro-3D Creative Tool Suite enables users to add immersive 3-D sound to any existing production workflow, while making the 3-D content available through a standard PCM single-file distribution. It is played back via existing standards such as DCP in digital cinema or via Blu-Ray for the home theater to reproduce in each channel high-resolution audio (HRA) to experience the sound as it was intended by the creators.



Photo 3: An Auro-3D crew recording in an outdoor setting. No information is included with the photo, but from the setting and microphone arrangement I suspect an aircraft flyover. The top, height, and lower recording levels are clearly shown.

though it's outdoors, they expect to be able to capture the full dimension of a fly over. From the angle of the camera position, we will get the information directly above captured with a top microphone, two front, and two rear microphones positioned on the height level and a center, left, right, and two rear surround microphones on the lower level (see **Photo 3**).

3-D Mastering

A suite of DAW plug-ins, Auro-3D Authoring Tools, is available for sale to work with 3-D sound (see **Photo 2** and **Photo 4**). There are five plug-ins and some of them are multi-functional. A basic work session consists of an Auro-Panner, Auro-Bus, Auro-Mixing Engine, and an Auro-Return plug-in. The Auro-Panner plug-in is the primary tool and several of them may be installed in the DAW, depending on the number of audio tracks. This is fully explained in the Auro-3D Authoring Tools Guide and a copy is included in the Supplementary Material.

Perhaps of more interest to us is the Auro-Matic Pro 3D plug-in available from both Auro and Minnetonka Audio. It supports the conversion of mono, stereo, and 5.1 into Auro-3D up to 11.1 and the conversion of 7.1 up to 13.1. At the time of this writing, the plug-in is available in only AAX format for the Mac Pro Tools DAWs. (Auro is working on AU and VST versions to become available in the near future.)



Photo 4: The Auro-Matic Pro 3D main screen lets you select the input and the output formats and variables including spatial size, the balance between front and rear, and the balance between the lower and height channels. Complete details are in the user guide.

Project Files

To download additional material and files, visit <http://audioxpress.com/page/audioXpress-Supplementary-Material.html>.

Resources

Denon, www.denon.com.

Merantz, us.merantz.com.

"Remote Galaxy," Flint Juventino Beppe, 2L Records, Norway.

Sources

Auro-3D

Auro Technologies | www.auro-3d.com

Pro Tools

Avid Technology, Inc. | www.avid.com/US/products/family/Pro-Tools

Dolby Atmos

Dolby Laboratories | www.dolby.com/us/en/technologies/dolby-atmos.html

Auro Technology

Minnetonka Audio | www.minnetonkaudioshop.com

About the Author

Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at the White Sands Missile Range. In 1957, he started Testronic Development Laboratory (now TDL Technology) to develop audio electronics. He is still the TDL president and principal designer.

Using Auro-Matic Pro 3D for a stereo or 5.1 conversion to a 9.1 output is straightforward when the plug-in is in stand-alone mode. A step-by-step guide is included in the Auro-Matic Pro 3D User Guide. A pdf copy is included in the Supplementary Material.

Listening to Auro-3D

Assume we have a conversion from 2-D or an audio or video Blu-ray disc, how do we set up a listening system? Unfortunately, no free or low-cost equipment or software is available. As of this writing, the most economical system can be built around a Denon AVR-X4100W player with an Auro-3D upgrade. New Denon players are available from around \$1,200. The 9.1 Auro-3D upgrade is \$199, and it can be done with an Internet connection to the AVR.

The suggested loudspeaker setup for 9.1 is shown in **Figure 2**. But here's the rub. This Denon player also decodes Dolby Atmos (which I'll discuss in a later article in this series). However, the loudspeaker arrangement is different, as **Figure 3** shows. In home-theater and theatrical applications, all "immersive" audio solutions, from Dolby, Auro or DTS have different speaker arrangements, even

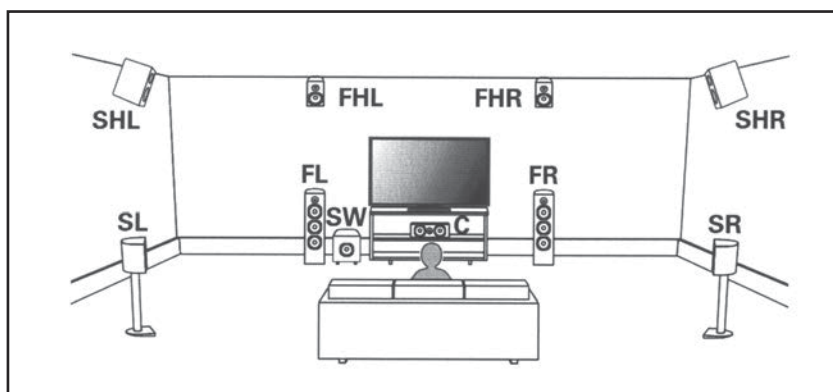


Figure 2: This is a suggested loudspeaker arrangement for an Auro-3D 9.1 system. Note that angle sizes and dimensions are not shown because they can be adjusted in the mastering software.

though Dolby and DTS claim their "object-oriented" approaches will be able to "render" the mix to any layout with or without ceiling speakers, something not endorsed by Auro Technologies.

Still, the single-file distribution and backward compatibility of the Auro-3D format enables 3-D sound to be easily delivered into the home via Blu-ray discs, with playback in Stereo, 5.1 / 7.1 Surround Sound, and Auro 9.1 (up to Auro 13.1). Additionally, the integrated Auro-Matic technology enables existing native Stereo and Surround Sound content to be

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up-mixed, converting all legacy film and music into a natural 3-D listening experience.

Apart from the more affordable Denon and Marantz AVR models, there are also more sophisticated home-theater offerings by Datasat

Digital Entertainment with its new Auro-3D Mensa and Crux AV Sound Processors, developed in partnership with Auro Technologies (see **Photo 5**). Auro Technologies also introduced its own Auro-3D Auriga AV Receiver in January 2014 (see **Photo 6**).

Auro Technologies is also working with selected manufacturers to introduce its immersive Auro-3D sound format to mobile devices (smartphones and tablets) and even headphones with the Auro Beautifyer app (<http://www.beautifyer.com>).

At this time, media availability is an issue and that's the reason for my emphasis on 2-D to 3-D conversion. However, there are several Pure Audio Blu-ray discs available. Check the current list at <http://www.auro-3d.com/consumer>. I have a copy of "Remote Galaxy." I haven't listened to the 9.1 content yet, but the stereo tracks are excellent and the DTS 5.1 and the 7.1 are awesome. My copy of the Mando Diao recording "Aelita" is on backorder and, as with the Remote Galaxy Blu-ray, it has tracks recorded in two-channel stereo, 5.1 DTS, and 9.1 Auro-3D at a 96 kHz sampling rate and 24-bit depth. These discs are music-only but quite a few movie theaters have Auro-3D sound capability. There are more than 100 movies encoded with Auro-3D. Although some of the movies were supposed to be released on Blu-ray, copies with Auro-3D soundtracks are not easily found.

3-D File Conversions

It was my intention to include some example stereo and 5.1 conversions to Auro 9.1 in the Supplementary Material. Unfortunately, that did not happen. Steve Severson arranged a 30-day trial license for Auro-Matic Pro 3D from Minnetonka Audio (thanks, Steve) but, of course, it's the Mac version because, at this time, there is no Windows version. Apparently, I was "asleep at the switch," as I had mentioned earlier that the plug-in was available only for the Mac. I did purchase a Windows edition of the Pro Tools DAW, which may be of use someday when a Windows version of Auro-Matic becomes available.

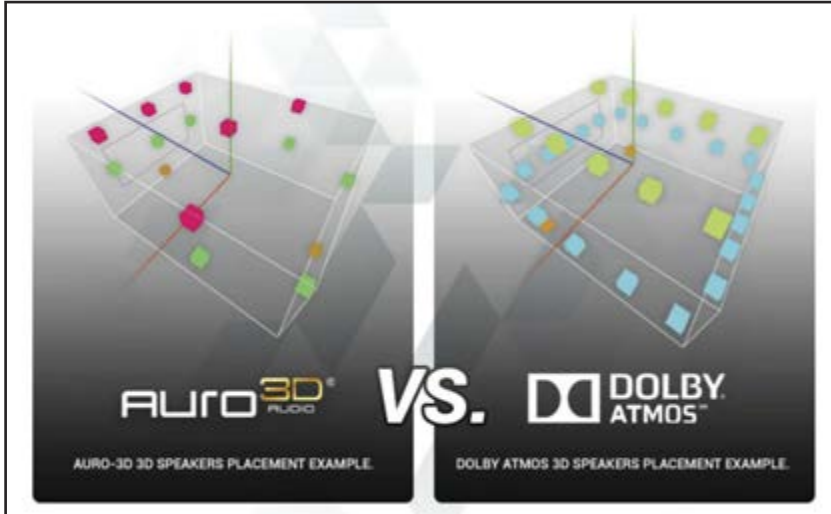


Figure 3: The Auro-3D and Dolby Atmos loudspeaker arrangements are different. Dolby says its Dolby Atmos could work with an Auro-3D setup. DTS says it supports both Auro-3D or Dolby Atmos-recommended speaker layouts for DTS:X.



Photo 5: Datasat's Auro-3D Crux is the first integrated 3-D sound home cinema processor for home cinema available at a lower price point.

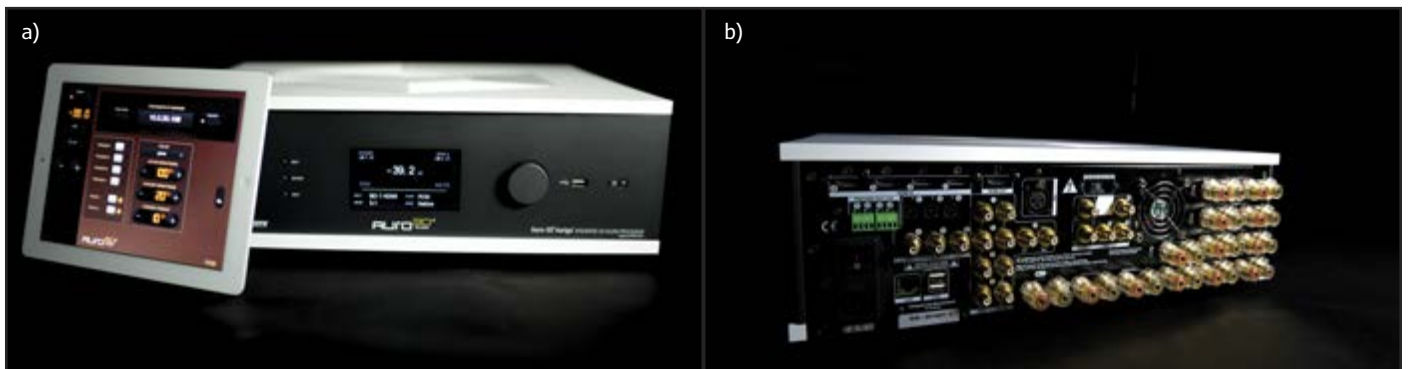


Photo 6: Auro Technologies also introduced its own Auro-3D Auriga AV Receiver in January 2014, see the front (a) and back view (b).

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Audio Network Development

Professional audio applications for live sound and recording lead the way for audio networking technologies with clear advantages. Companies such as Focusrite are reaping the benefits of such developments with state-of-the-art solutions (e.g., its RedNet range of modular Ethernet-networked audio interfaces).

An Introduction to Audio Networking

Your company is designing a next-generation audio product. Whatever application and market your new design targets, to make sure you are releasing something that stands out in today's competitive landscape, you should consider some sort of network connectivity. This article provides insights on the evolution of audio network connectivity, and available technologies and platforms.

By
João Martins
(Editor-in-Chief)

The first reason why any new audio project design should consider a network interface is not necessarily related to audio but rather to upgrades. The market is getting used to solutions that can keep up with market requirements and continuously improve, preferably with field upgrades. For that purpose, considering some sort of interface to connect an external computer (e.g., USB) is no longer enough. Systems need upgrade ability while deployed in the field, and without need of technical support intervention. For consumer electronics, that's one of the reasons why wireless LAN connectivity is becoming standard in every product category, which are also benefiting from the evolution in connected embedded solutions.

But apart from firmware and software upgrades, there is another major reason to consider network connectivity: the audio interface. Yes, even in applications where the input or output might still be analog audio, there's a compelling reason to choose a network interface and adopt an Internet Protocol (IP) compatible interface. Not only will you be able to ensure diagnostic and monitoring capabilities

through a direct or remote connection, you will also be able to implement field upgrades using the same interface. And, your system will be able to support remote control—another important requirement in today's market, especially in pro audio applications.

If your new audio design is targeted at simple consumer level applications and mobility is a key factor, you might rely on a wireless interface—preferably Wi-Fi—to fulfill those requirements. But if you are targeting professional audio level signals, several audio channels, or high-quality digital streaming, wireless might not be enough and an audio network interface will be required.

Why “Network”?

Yes, your design might only require a point-to-point data interface where, typically, a common communication protocol such as USB will seem sufficient. The Universal Serial Bus standard—developed in the mid-1990s—is also evolving. With the recent USB 3.1 Superspeed+ specification and the introduction of the new USB Type-C universal connector, it is clear that USB applications will

“Ethernet makes overwhelming sense.”

—Steve Church

eventually merge with next-generation interfaces (e.g., Intel’s Thunderbolt 3 high-speed dual-protocol I/O, with both data and video support on a single cable connection).

The reason for the convergence makes total sense in computer applications, providing a single universal daisy-chainable connector. Intel has decided to adopt the same Type-C reversible connector (replacing the original mini-DisplayPort connector). Thunderbolt 3 provides up to 40 Gbps bidirectional data rates, (USB 3.1 supports up to 10 Gbps), DisplayPort 1.2 (which supports multiple 4K and 5K displays), external graphics with four lanes of PCI Express Gen 3, built-in 10 Gigabit Ethernet (GbE) networking, and peer-to-peer connections at 10 GbE speeds.

Having 10 Gigabit Ethernet network support on the same interface—which currently benefits from the economies of scale provided by support from the computer, media, and telecom industries—is an important consideration for designs now in the development stages.

Specifically for audio applications, with the industry increasingly adopting common network and IT standards—rather than simply relying on the Ethernet physical layer—there’s every reason to consider the bigger picture, in which Audio-over-IP (AoIP) network connectivity is becoming something users will expect.

With important improvements on codecs and streaming audio formats, consumer applications for network audio streaming are no longer restricted to low fidelity. Also, audio transport over the Internet is continuously improving, to the point of it being possible to configure applications leveraging both higher performance local area network (LAN) solutions and remote distribution or contribution, depending on the latency requirements.

Also, using the Ethernet standards, it is possible to provide direct audio support to the network, replacing computer “sound cards” and even hardware audio interfaces (e.g., USB, IEEE 1394, etc.) with simple software drivers. This means avoiding problems such as noise and multiple conversions as much as possible—and that’s where audio technology is clearly headed.

Audio Networking Foundation

For about two decades, several companies have been investing in audio network transport protocols and streaming. Following early efforts that resulted in relatively successful implementations (e.g., CobraNet, AES50, and EtherSound, which tried to establish the first licensed standards) to proprietary solutions (e.g., Aviom A-Net, Riedel’s

Rocknet, Optocore, and several others), industry consortiums actively pursued a consensus. As a result, a few modern platforms have become “de-facto” standards, such as Audinate’s Dante, RAVENNA, Telos’ Livewire, and Audio Video Bridging (AVB)/Time Sensitive Networking (TSN). Most importantly, the first effort for high-performance streaming AoIP interoperability has come to fruition with the AES67 standard, published in 2013.

In our Standards Review column, *audioXpress* has published articles detailing some of those AoIP technologies, and we think now is the right time to resume the topic from a product development perspective. Because the good news for audio manufacturers and product designers is that the same companies, consortiums, and organizations that promote the technology also recommend implementers, deliver evaluation platforms, software development kits, application program interface (APIs), and even complete development platforms for direct product implementation that companies can adopt for building new projects.

Currently, we have four main competing technologies and a common interoperability standard—AES67. But those technologies are now addressing multiple applications and market



Steve Church, the founder of Telos Systems, believed computer networks and a single Ethernet cable could carry multiple audio signals. His work led to the development of Livewire AoIP technology in 2003. It is now deployed in thousands of facilities worldwide. In 2010, Church, together with Skip Pizzi, authored the book *Audio over IP: Building Pro AoIP Systems with Livewire*.



From home studios to broadcast, the benefits of audio networking technology are now clear, and high-profile companies, such as Solid State Logic, are betting heavily on AoIP implementation.

Kevin Gross, a pioneer in media networking, is now chairman of the Technical Work Group for the Media Networking Alliance (MNA). He was previously responsible for the CobraNet transport protocol, developed Q-LAN technology for QSC, participated in AVB development, and lead the group that produced the AES67 interoperability standard.



requirements and becoming exciting innovation enablers in all areas of audio—from home audio to recording, live sound, installation, and broadcasting.

Apart from providing an update of where those audio networking technologies are heading, this article series intends to address the question of who the companies are that support the product implementation efforts in regard to audio networking.

State of the Nation

Without fully revisiting the history of audio networking standards, we can summarize the current market evolving around four dominant proposals: Audinate's Dante, RAVENNA, AVB/TSN, and the first interoperability audio networking standard, AES67.

Other original audio networking technologies and pioneering efforts remain mostly "closed-gardens" or proprietary solutions, even though they were the foundation for the solutions currently available. One of the pioneering efforts in this domain was the CobraNet protocol, designed by Peak Audio to deliver uncompressed, multichannel, low-latency digital audio over a standard Ethernet network. Unfortunately, further development of CobraNet was effectively discontinued (even though it continues to be widely deployed) following the acquisition of Peak Audio by Cirrus Logic in 2001.

The team responsible for CobraNet, including Kevin Gross and Rich Zwiebel, also developed Q-LAN. Owned by QSC Audio, it is one of the most robust and efficient proprietary AoIP protocols. Q-LAN also essentially remained the audio network component for QSC's Q-Sys audio signal processing platform, which is still one of the most widely deployed solutions for permanent audio installations.

Another pioneering effort was Livewire, an audio-over-Ethernet system created by Telos/Axia Audio (The Telos Alliance group of companies). Following some pioneering efforts on audio streaming over the Internet, the late Telos Founder and President Steve Church focused on replacing the analog infrastructure in radio stations. Church used computer network technology and software as a foundation to fully replace the need for computer sound cards. Livewire was introduced in 2003 and quickly became the dominant AoIP technology in radio broadcasting, supporting everything from large-scale routing and distribution to simple stereo audio sources for remote contribution. There are more than 80 manufacturers adopting the technology and many thousand installations worldwide.

Both Q-LAN and Livewire remain widely deployed proprietary solutions, and both efforts

have contributed to the new AES67 standard for AoIP interoperability, now fully adopted by both companies. The Telos Alliance developed Livewire+, designed as a superset of functionality, utilizing common protocols and formats to bring the original Livewire protocol into AES67 compliance.

Telos worked with other manufacturers to develop the AES67 standard for AoIP interoperability, and The Telos Alliance is a founding member of the Media Networking Alliance (MNA), which promotes adoption of the AES67 standard on behalf of the pro audio industry.

Q-LAN remains a “closed-garden” audio network solution, exclusively implemented on QSC’s hardware products, even though the company provides support for integrators and provides basic development tools for remote control and monitoring interfaces as well as user interface customizations. Because QSC’s Q-Sys solutions run on standard Intel platforms, the company is able to continuously evolve the ecosystem with increasingly powerful processing features. QSC continues to provide hardware solutions to bridge existing AoIP standards. The fundamental change in this philosophy has been the May 2014 release of new software for the Q-Sys Network Audio platform, which now meets AES67 network clock and audio packet delivery standards. Since then, all updated Q-Sys solutions have become fully compliant with AES67 interoperability mode.

The importance of adding this new AES67 audio networking interoperability standard started with the Audio Engineering Society’s X192 task group, with determinant contributions from Telos/Axia Audio (Livewire), QSC (Q-Lan), and RAVENNA/ALC NetworX. Kevin Gross, who is also the mastermind behind the CobraNet transport protocol and Q-LAN, provided much of the work for the original AES67 open standard specification. Gross is now the chairman of the MNA Technical Working Group, a not-for-profit organization created to promote the adoption and future updates for AES67.

Gross is also the founder of AVA Networks and a leading independent consultant to AV equipment manufacturers and systems designers, providing support in media networking development. He has also contributed work to multiple standards bodies, including the Institute of Electrical and Electronics Engineers (IEEE), where he participated in AVB development and the Internet Engineering Task Force (IETF).

Considering AES67 Implementation

As stated by the MNA, AES67 is a standard to enable high-performance AoIP streaming, providing interoperability between the various IP-based audio-

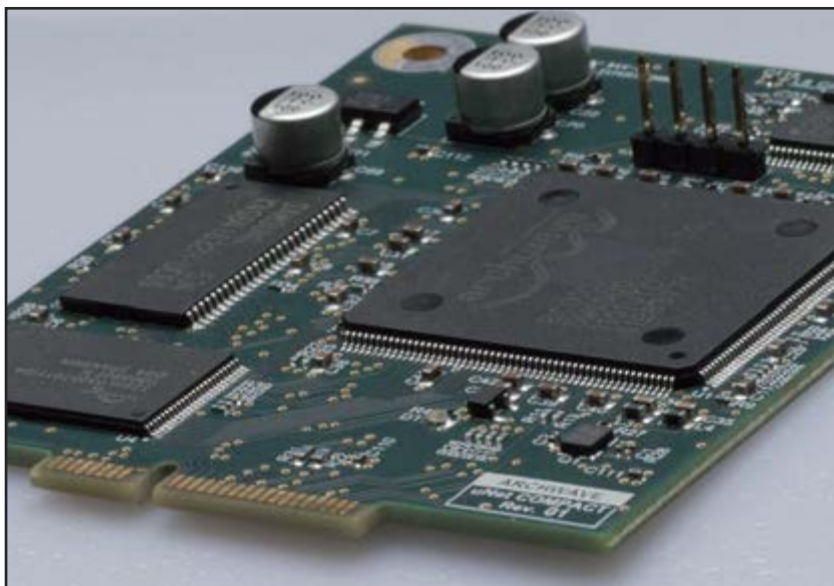


Audio-Technica’s ATND971 Network Microphone debuted at InfoComm 2014 and was the first microphone in the market to transmit audio and control data over Dante audio networks. The ATND971 is also powered by network PoE.

networking products currently available (e.g., Dante, Livewire, Q-LAN, and RAVENNA). It is not a new technology, but rather it is a bridging compliance mode common to all IP-networks—a mode to put a device into, on any participating network. AES67 operates over standard layer 3 Ethernet networks and, as such, is routable and fully scalable, similar to any common modern IT network. As a rule of thumb, any network infrastructure capable of handling VoIP should be able to handle AES67.



Coveloz’ AES67/Ravenna and Ethernet AVB audio networking and processing FPGA Development Kit offers a robust design platform built around the Altera Cyclone V System-on-Chip (SoC) FPGA. This kit is also host to the BACH-SOC firmware that provides FPGA and Linux firmware, enabling customers to quickly build AVnu-certified and AES67-capable products.



Archwave Technologies offers a series of modules in its AudioLan product family for streaming audio over AES67/RAVENNA networks. Archwave AudioLan is a completely integrated hardware and software package for professional-grade audio through an AES67-compliant network, supporting from 4 to 64x64 channel development platforms on its uNET series.



**Media
Networking
Alliance**
Promoting the adoption of AES67

During InfoComm 2015, the Media Networking Alliance promoted an “open” panel discussion on “How is AES67 Going to Change Our Industry?” moderated by Media Networking Alliance chairman Rich Zwiebel, VP of Systems Strategy at QSC, with participation from Johan Wadsten, Software Product Manager at Merging Technologies; Andreas Hildebrand, Senior Product Manager for the RAVENNA technology; Phil Wagner, President of Focusrite Novation; and Ethan Wetzell, platform strategist at Bosch Communications Systems.

This means companies that have already embraced existing audio networking technologies and want to provide audio interchange interoperability with other products should consider implementing AES67 support. And, several companies that provide AoIP development platforms and implementation services are already supporting AES67 development efforts.

But should a manufacturer rely solely on AES67 implementation for audio networking? As stated by the MNA founding members, AES67 is not intended to fully replace full-featured solutions such as Dante, RAVENNA, or AVB/TSN. These solutions offer different operating modes and expanded functionalities, and AES67 is basically a subset or an operating profile. AES67 does not include specific requirements for discovery and control functions, for instance. It simply describes interoperability modes, synchronization, encoding format, and QoS provisions for delivering audio data and connection management functions associated with audio delivery. It defines how to use existing protocols as a system in an interoperable manner. AES67 also uses existing standard protocols and technology from the IEEE, IETF, and other standards developing organizations.

As Andreas Hildebrand from ALC NetworX (the developers of RAVENNA) stated, “AES67 cannot be used on its own. It simply allows different audio protocols to talk to each other, so manufacturers who want to enjoy the interoperability of AES67 need to choose a protocol that is AES67-compliant to fit into their products.”

If the market requirements of a new audio product simply require audio data interchange, AES67 should be considered in the implementation. But, any approach to an engineering company that provides tools and services for audio networking development platforms will show one of the AES67-compatible solutions (e.g., Dante, RAVENNA, or Livewire+), depending on the market application, will offer a better return on investment.

The AVB IEEE 802.1 standards efforts have yet to confirm official AES67 support. The now renamed Time-Sensitive Networking Task Group is increasingly expanding the scope of its work while leaving many questions unanswered, including what to do with interoperability of existing AVB implementations.

According to the information provided by the MNA, “AVB standards define improvements to Ethernet providing synchronization, QoS and an admission control behavior that prevents the network from becoming overloaded by media traffic. AES67 is compatible with and receives performance

benefits from these capabilities on an AVB infrastructure."

The problem seems to be that "AVB defines two types of transport protocol for media data. IEEE 1722 is an Ethernet-only transport and is incompatible with the defined scope of AES67. IEEE 1733 defines an IP transport for AVB based on Real-time Transport Protocol (RTP).

This "flavor" of AVB is compatible with the scope and the goals of AES67. Interoperability with this type of AVB is described in annexes C and D of the standard. AES67 enables AVB implementations to reach beyond Ethernet into wider-area applications."

It's also important to understand that work on the AES67 standard will continue. In October 2015, a maintenance revision to the original AES67-2013 standard was issued.

The revision, called AES67-2015, clarifies some of the interoperability requirements, based on general implementation and "plugfest" experience, and mainly attests

to its robustness.

Commenting for the MNA, Gross, stated: "The limited scope of these revisions is testimony to the robust and stable nature of AES67 as originally published and successfully implemented by numerous audio manufacturers. As proven in the plugfest testing, AES67 functional compatibility performs as described and these revisions further complement that functionality."

The MNA continues to develop future AES67 specifications and other documents that augment, enhance, or extend the primary AES67 specification, while providing developer support for AES67 products offering an AES67 System Development Kit and AES67 Test Tools.

Recently, Audinate also announced the delivery of its AES67-enabled firmware to OEM partners, offering AES67 compatibility with its Dante protocol.

In a follow-up article, we will detail the development of Dante, RAVENNA, Livewire+, and AVB/TSN platforms. 

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Audio Tools from Studio Six Digital



As was evident from last month's Sound Control article, "Smartphone Apps for Sound Measurement," many mobile apps have been developed for making electroacoustical and acoustical measurements. However, in researching these apps, I kept finding references to one company that is responsible for the most comprehensive and professional such products—Studio Six Digital.

By
Richard Honeycutt
(United States)

Studio Six Digital was founded by Andrew Smith. During his 30-plus years' experience in consulting, software design, and development, Andrew also founded TerraSonde, which created the Audio Toolbox, a pace-setting multi-function audio test device that inspired several competitors. The Audio Toolbox is now marketed by Sencore as the SoundPro. However, Studio Six did, and continues to do, all the R&D for this product line.

Audio Tools

The Studio Six suite of mobile apps for sound measurement is called Audio Tools, and is presently

available only for iOS devices, although there are ongoing efforts to port as much of the Audio Tools functionality as possible to the Android platform. Many of the Audio Tools functions are also available as stand-alone apps.

Included in the base price of the Audio Tools app are the following measurement functions:

- **Sound Pressure Level (SPL) Meter**—An analog-look sound level meter, which, when used with the Studio Six iAudioInterface2 and the internal microphone in iOS 5, can measure up to 120 dB SPL. It includes A & C weightings, and Fast and Slow modes. Needle ballistics accurately represent the Fast and Slow response of an analog meter. Filters and ballistics meet ANSI/ISO Type 1 specifications, when used with a suitable microphone.
- **Real-Time Analysis (RTA)**—Provides octave and 1/3 octave spectral analysis. Includes cursor readout of decibel bands, maximum and minimum lines, noise curve overlays, a pink noise generator, and optional sound transmission class (STC) transmission loss. It meets all the ANSI and the ISO Class 1 standards for filter-based RTA measurements. It also has a built-in NC readout.
- **Fast Fourier Transform (FFT)**—FFT analysis is offered in two ranges, 20 to 20 kHz, and 5 Hz to 3,100 Hz. Up to 32k point FFT, smoothing, decay, peak frequency tracking, and overlays.
- **Delay Finder**—Sends a signal to a driver, then measures the time it takes for that signal to return. Shows time in milliseconds, as well as distance.
- **Level/Frequency**—Measures the decibel level and frequency of the iAudioInterface2 line input or

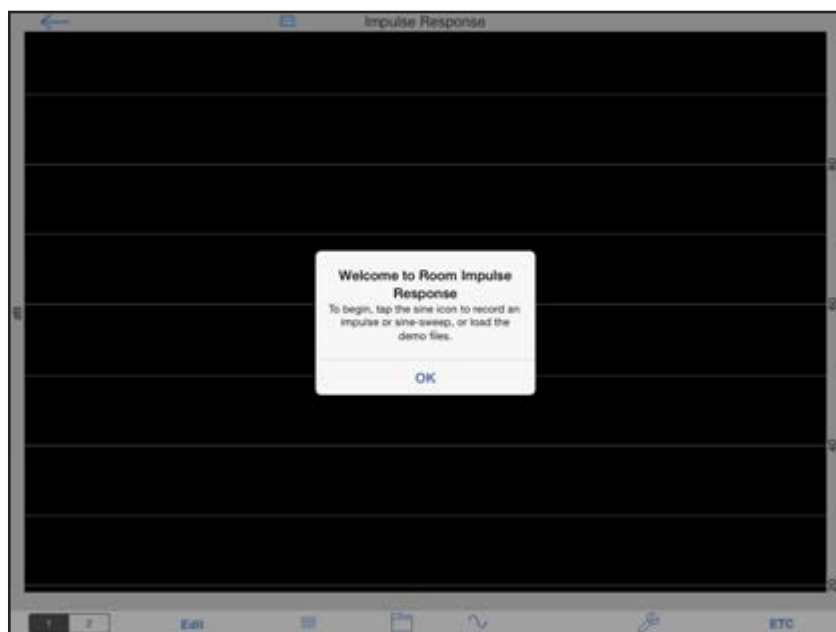


Photo 1: To select the Impulse Response mode, the user clicks the sine-wave icon on the opening screen.

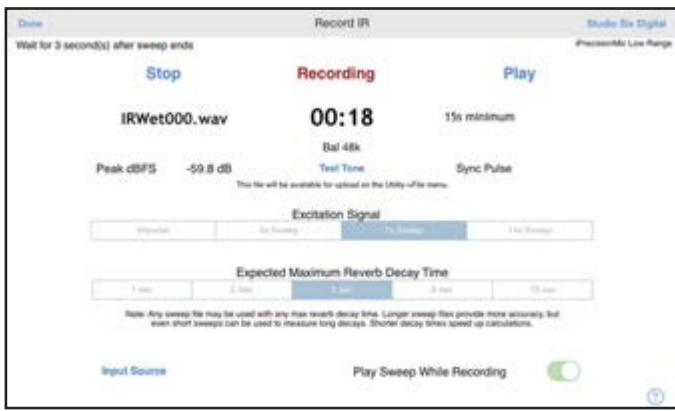


Photo 2: This setup screen permits the user to specify Impulse Response test parameters.

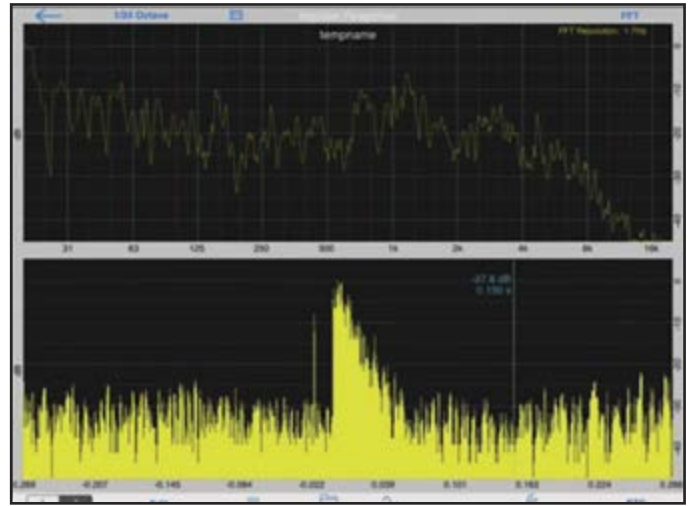


Photo 3: The unedited Impulse Response is shown in this screen.

line output (also works with any dock input audio source). Units of dBu, dBV, Vrms, Vpp, and Vavg may be selected.

- **Audio Scope**—A dual-trace audio-band oscilloscope that includes auto- and single-trigger modes, variable sweep rate, and up to 128x gain. Includes X-Y and M-S modes, and cursor readout of frequency.
- **Generator**—Audio signal generator that creates sine waves, square waves, white noise, and pink noise. The noise signals may be octave-band limited. The user can set sweep start and stop frequencies, and sweep time in seconds. The sweep, once set up, can be run from the generator screen, and from the FFT and SmaartTools screens.

More Modules Available

Additional modules are available as in-app purchases. Some of the modules include:

- **SPL Pro**—A professional full-featured digital SPL

meter. ANSI/ISO Slow, Fast, Impulse, and Peak modes. Includes peak hold, LEQ with pause/run, octave band filters, and more. All filters and exponential decay modes meet or exceed ANSI and ISO Type 0 standards. It includes reference mode (seat-to-seat), and peak hold, and bright and dark screen options.

- **SPL Graph**—Plot SPL over time (up to 24 hours) and optionally record the audio. Supports A, C, and octave-band weightings. Includes remote logging of SPL over 3G or Wi-Fi, Ln measurements, auto-save and restart, playback of audio events in the app.
- **SPL Traffic Light**—Set three SPL decibel levels to trigger the green, yellow, and red lights. Fast or Slow decay, or Leq from 1 minute to 60 minutes. Remote logging available.
- **Energy Time Curve (ETC)**—Plots the decay of an impulse over time. Also computes RT60 from the curve. A, C, and octave band filters are available. Speaker delay may also be measured.
- **Impulse Response (IR)**—Record an IR

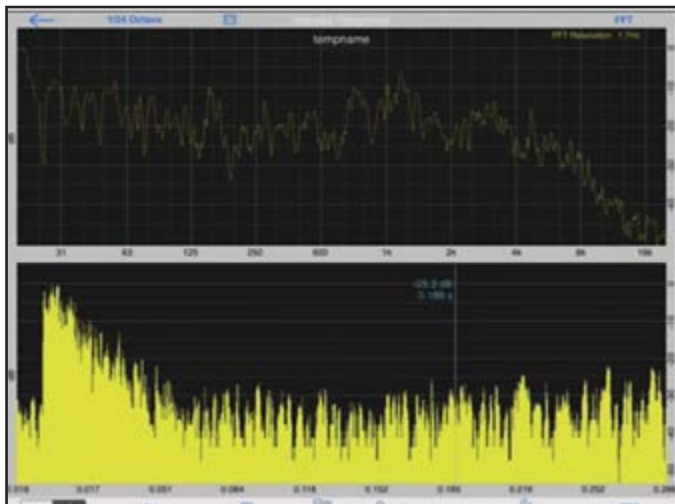


Photo 4: In this display, we see the impulse after extraneous noise has been removed at the beginning and the end.

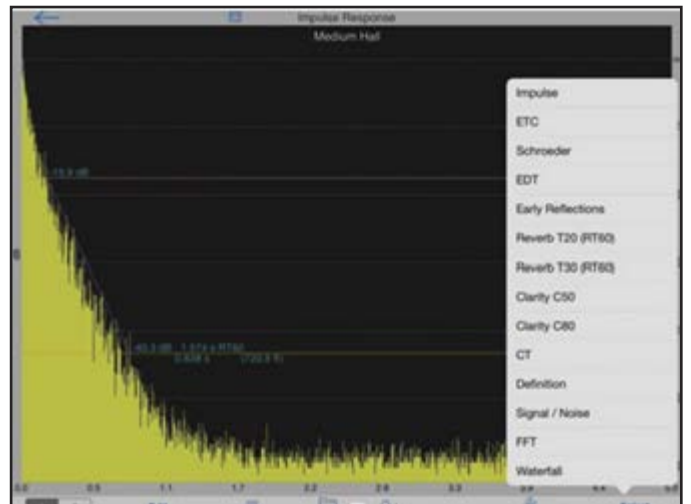


Photo 5: This menu shows the analysis/display options for Impulse Response measurements.

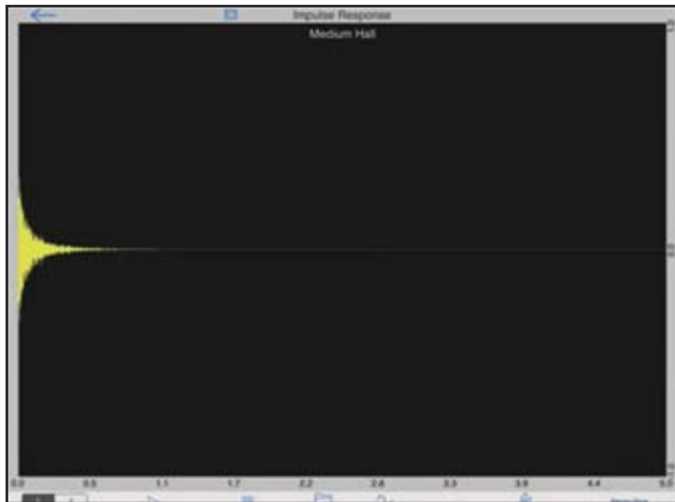


Photo 6: The recorded impulse is shown here.

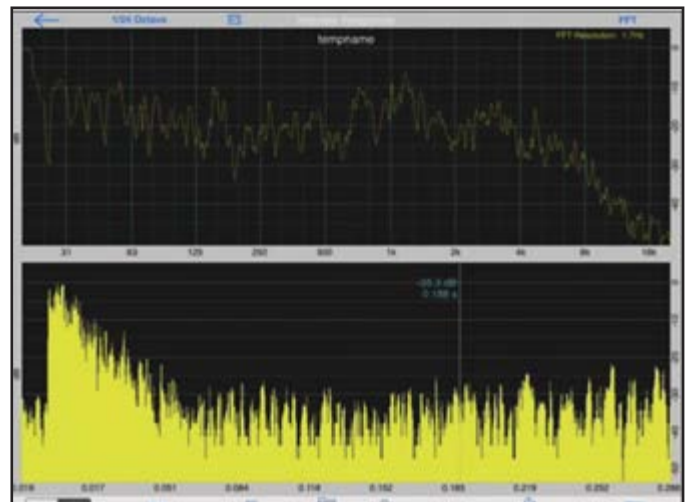


Photo 7: A split-screen display can be used on an iPad.

audio file, and then analyze the IR for commonly used metrics, including 1/3 octave RT60, clarity, definition, and EDT. Both the data and the IR can be uploaded to a PC or a Mac. Includes 3-D waterfall plots and windowing.

- **Speaker Polarity**—Determines speaker polarity by analyzing a proprietary signal.
- **Total Harmonic Distortion plus Noise (THD+N)**—Measure the THD+N of the line input signal. Requires iAudioInterface.
- **Amplitude Sweep**—Measure the amplitude-response linearity of equipment. Requires iAudioInterface.
- **Impedance Meter**—Measure the impedance from 0.5 Ω to 5,000 Ω $\pm 1\%$ or 1 Ω . Also computes power requirements in watts for a given voltage speaker system. Results are shown numerically, or may be plotted. Requires iAudioInterface 1 or 2.
- **Smaart Tools I**—Single-channel spectrum

measurements ported from Rational Acoustics Smaart 7. Includes octave to 1/48th octave spectrum line and bar plots, and the spectrograph.

- **Transfer Function**—A true two-channel transfer function, complete with magnitude, phase, and coherence plots, coherence masking, and a sample-accurate impulse response-based delay finder. Requires a two-channel audio interface, such as iAudioInterface2.
- **STIPA Basic**—Speech intelligibility module that meets all of the requirements of the international STIPA standard.
- **STIPA Pro**—Speech intelligibility module. Industry-standard test for speech intelligibility, based on version 4 of the international standard. Complies with all current fire and building codes. Includes advanced features, including averaging of multiple tests, save/recall of results, and the ability to email detailed results, or to transfer them to

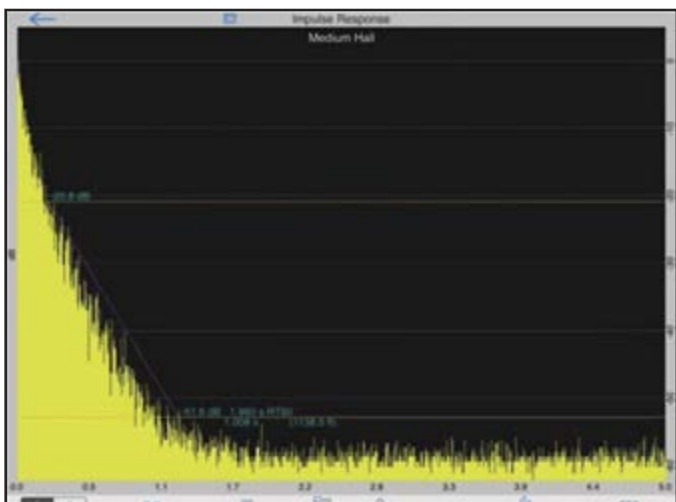


Photo 8: The Reverberation Time (RT) can be calculated and displayed based upon manual selection of start and end times.

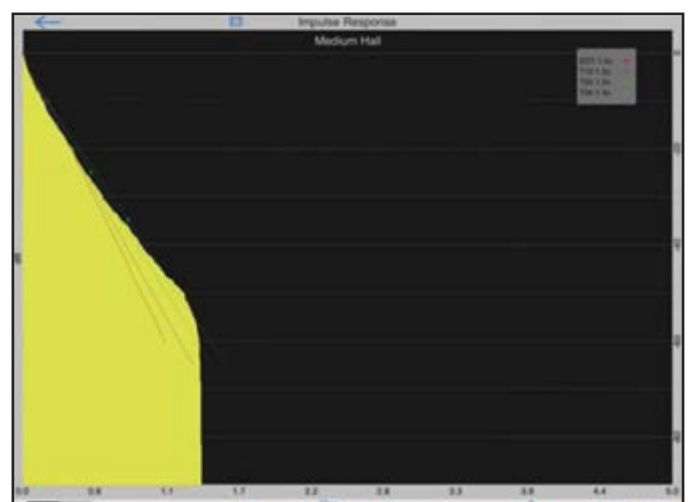


Photo 9: Schroeder plots and calculations can be performed for T10, T20, and T30.

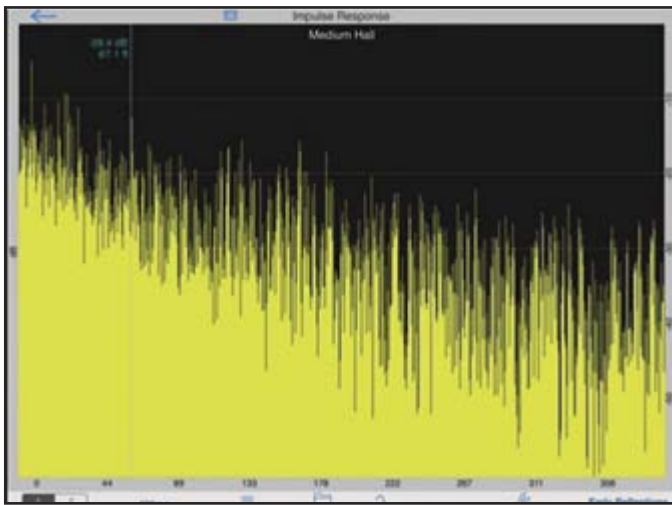


Photo 10: Early reflections can be studied in detail.

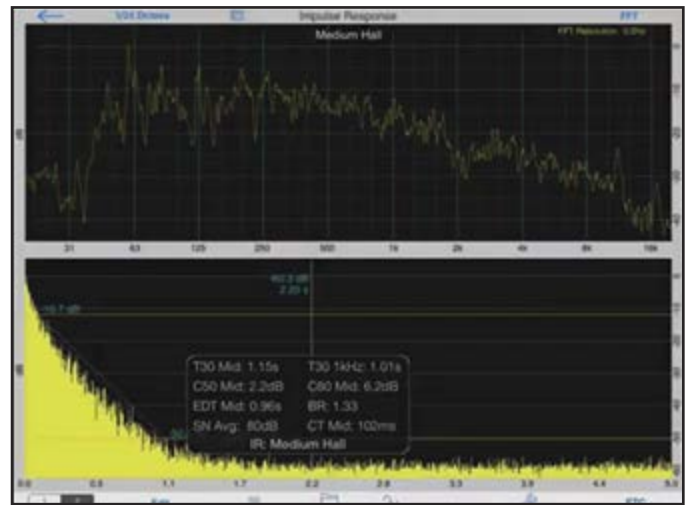


Photo 11: Other acoustical parameters are summarized on this screen.

your computer for reporting.

Product Evaluation

To evaluate Audio Tools as an acoustics measurement platform, we will examine the IR and Speech Transmission Index (STI) capabilities in more detail. Audio Tools can perform an IR measurement using either an impulsive sound or a sweep generated within the app. To enter the IR mode, the user taps the sine-wave icon, which can be seen at the bottom of **Photo 1**. Then the IR setup screen shown in **Photo 2** appears. From this screen, the input source and excitation type are selected, as is the maximum IR length (specified as the maximum expected RT of the room).

After the Record button is tapped, the impulse is recorded. Once the sweep has ended, and an interval equal to about one second longer than the expected RT has elapsed, the user taps Stop, and the impulse recording concludes.

Tapping Done brings up a display of the recorded IR (see **Photo 3**). The recording may be

edited by using a two-finger motion to bring the beginning of the impulse to the left-hand side of the screen, and to eliminate most of the noise after the conclusion of the decay. **Photo 4** shows the edited impulse. It can be saved by tapping Save File on a small dialog box that appears on the screen.

After the file is saved, a new dialog box appears. The user then taps Deconvolve to create the IR. Tapping the wrench icon at the bottom of the main IR screen brings up the menu shown in **Photo 5**. Tapping Impulse causes the IR to be displayed (see **Photo 6**). The ETC, which is the magnitude of the IR, can be shown as the bottom half of the split display on an iPad (see **Photo 7**), with the frequency response in the upper half. Only a single-screen display is available on other iOS devices.

In addition to recording impulse responses, Audio Tools can import IR files that have been recorded as 16-bit mono files with a 48-kHz sampling rate.

The room's RT can be calculated and displayed in several ways. Vertical cursors can be set at desired parts of the decay on the ETC using a two-finger

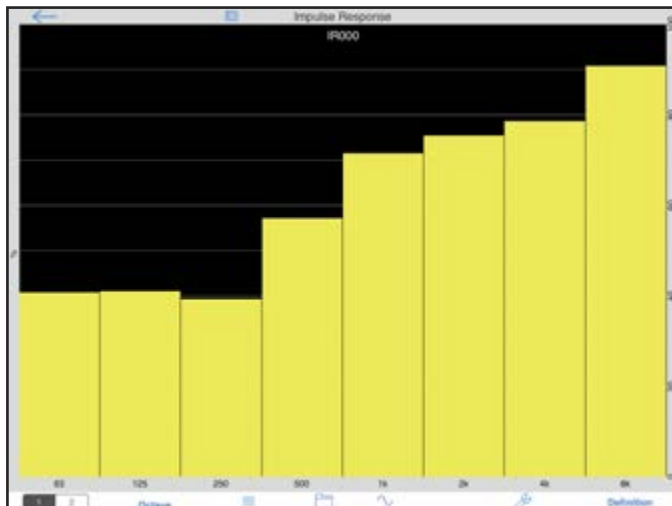


Photo 12: Many acoustical parameters can be displayed in octave-band averages.

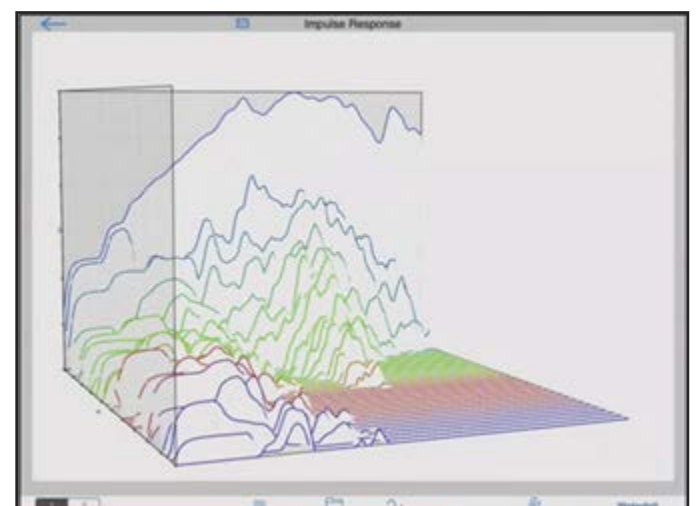


Photo 13: Waterfall plots can be useful in analyzing the behavior of decay vs. frequency.



Photo 14: This is the STIPA opening screen.



Photo 15: When the STIPA measurement is in progress, the screen changes.



Photo 16: The octave-band details of the Speech Transmission Index (STI) measurement can be viewed once the measurement is complete.

gesture, and RT will be calculated and displayed based upon these (see **Photo 8**).

If Schroeder is tapped in the menu dialog box, broadband Schroeder plots will be created for T10, T20, and T30, and the corresponding numerical values will be displayed (see **Photo 9**). The Early Reflections option on the menu expands the early portion of the ETC, enabling the user to study Early Reflections. The time span displayed is user adjustable. A cursor is available to provide identification of the Early Reflection's SPL and the time or distance from the speaker to the reflecting surface to the microphone (see **Photo 10**).

Other IR-derived acoustical parameters can be displayed in summary form (see **Photo 11**). The "mid" values correspond to the average of the 500-Hz and 1-kHz values. In addition to the RT-based and early-to-late-ratio parameters, this screen displays Bass Ratio (BR), Center Time (CT), and the signal-to-noise ratio (SNR). The latter is helpful in judging the reliability of the other results, as a minimum SNR of 60 dB is preferred, although values in the 50 dB range may be acceptable.

The 1-octave- or 1/3-octave-band averages of many acoustical parameters can be displayed (see **Photo 12**). These include EDT, T20, T30, C50, C80, BR, CT, and definition (D50). In calculating the values to be displayed, if insufficient SNR is found for any

band, the bar for that band will not be displayed. If the SNR is questionable, the bar will be displayed in red rather than yellow.

A waterfall plot is a 3-D graph of energy, frequency, and time. Such a plot assists in analyzing the decay vs. frequency, which can help pinpoint certain objectionable acoustical anomalies in a room. **Photo 13** shows an Audio Tools waterfall plot.

Measurement of speech intelligibility is performed using either the STIPA Basic or the STIPA Pro module. When the module is first opened, the screen appears as shown in **Photo 14**. The user taps Run, and the screen changes to the one shown in **Photo 15**.

Once the measurement is complete, the detailed octave-band results are shown (see **Photo 16**). STIPA Pro operates similarly to STIPA Basic, but also provides averaging, more measurement options, and the ability to save (using the Files utility in Audio Tools), and to export in the Excel format.

In addition to the Audio Tools app, Studio Six Digital also sells several test microphones and the iAudioInterface2, a balanced line/microphone preamp made for iOS devices. This audio interface provides 48 V phantom power for a microphone, very low noise, and precision A/D conversion, as well as an internal Li-ion battery, balanced line output, plus a Toslink output.

Demo videos for many of the Studio Six apps are available at www.studiosixdigital.com and on YouTube.

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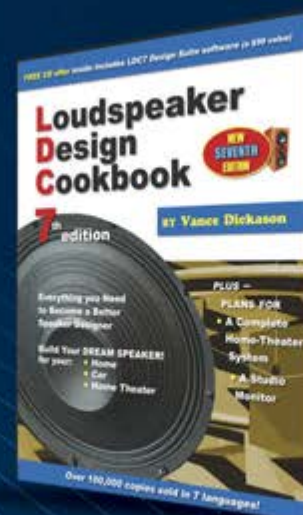
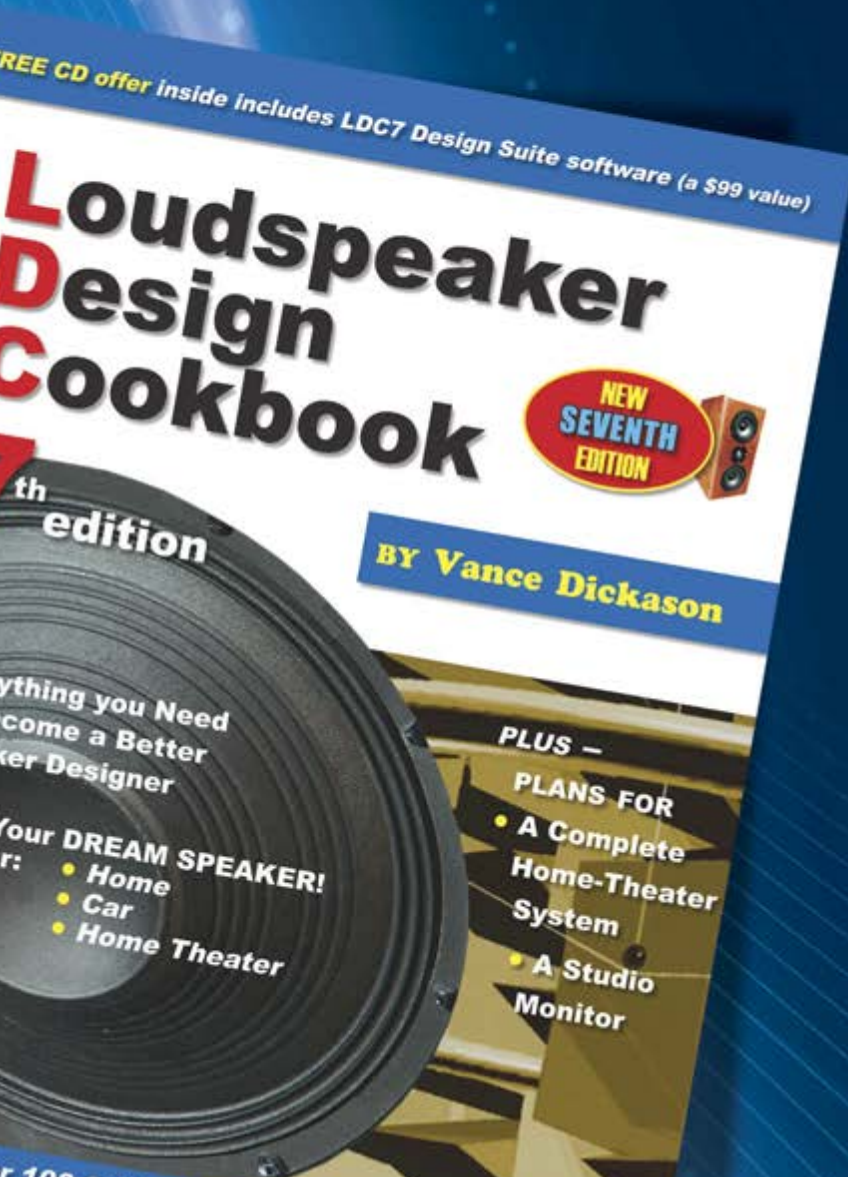
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You Can DIY!

The SRX Plus Hybrid Amplifier (Part 2)

Construct a Modified Tube Amplifier

I designed this tube amplifier with solid-state support, to use two different pairs of electrostatic headphones. In the first part of this article, I discussed the design. In the second-part, I detail the power supply, the construction, and the test results.

Photo 1: I designed the SRX Plus Hybrid Amplifier to drive my Stax electrostatic headphones.

By
James Lin
(United States)

The power supply provides the raw material the amplifier uses to convert the input signal into the output. The headphone diaphragm sits an inch or less from our ears, so a quiet power supply is necessary. The AC power line is far from pristine. A transformer with electrostatic shielding minimizes the amount of crud entering from the AC line. Look for a transformer diagram that has a dotted or solid line between the primary and secondary windings that is connected to ground.

With this design, I used a hybrid bridge rectifier with a rectifier tube for the positive voltages and solid-state tubes for the negative voltages (see **Figure 1**). Damper diode tubes produce the least amount of noise, have a relatively low voltage drop, and are inexpensive. HEXFRED soft recovery diodes also generate significantly less noise than normal silicon diodes. The tube rectifier provides some high voltage time delay. However, solid-state diodes immediately turn-on, so I added a time delay circuit from K&K Audio on the negative line with a 14-second delay and a 2-second rise time.

Choke input filters provide a smooth current flow from the power transformer without sharp spikes in the current flow as with capacitor input filters.

This results in less broadband noise for the filter sections. Also, a choke filter's high-frequency noise is 12 dB/octave high frequency rolloff vs. 6 dB/octave for a RC filter. The disadvantages of a choke are the cost, bulk, and weight. To calculate the choke value needed to have current flow over the entire AC cycle, use the following equation:

$$L > RL/1130 = 0.88 \times V/I \text{ (mA) for a 60 Hz AC}$$

where: RL = power supply voltage (volts)/power supply current (amps).

The first choke section requires at least 10 H in each leg. A second choke/capacitor section minimizes noise and ripple. The raw power supply puts out about 780 V rail-to-tail.

High-Voltage Regulators

Besides the use of current sources, some modern headphone amplifiers also have a regulated power supply, which stabilizes the amplifier circuit. We are direct coupling the amplifier output to the headphones, so variations in the circuit voltages can lead to undesirable drift in the output voltage. Since our active circuit uses current sources in both

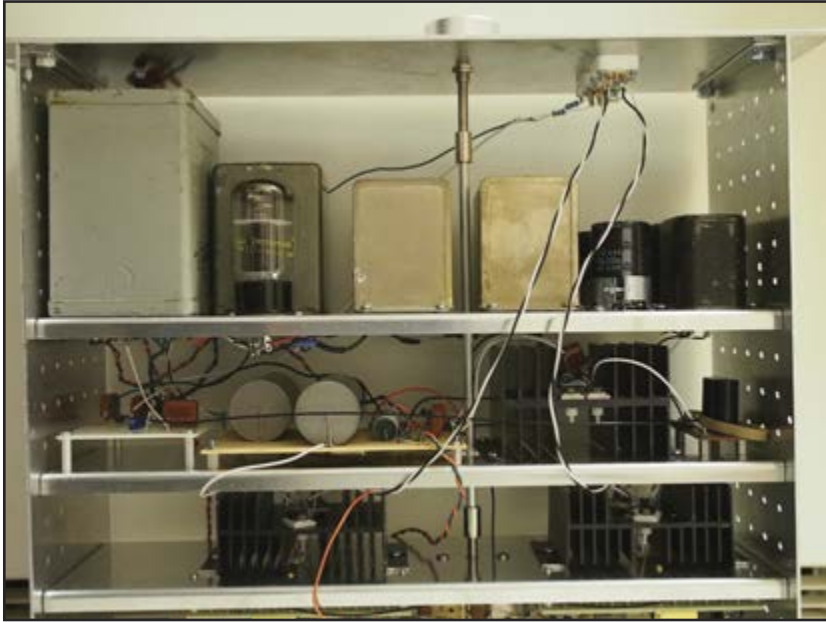


Photo 2: Top view of the amplifier interior shows the front of the amplifier at the top. The front shelf holds the raw power supply, the middle shelf holds the power supply current source, the shunt regulator heatsink, the regulator circuit board, and the headphone bias supply. The output heatsinks are mounted on the shelf closest to the rear of the amplifier (bottom of photo).

capacitor filters regulator noise and stabilizes the SPX431 against oscillation. B+ noise and other signal variations are directly fed into the lower MOSFET gate by a 0.20 μF capacitor for cancellation. The headroom between the raw power supply and the regulated voltage is about 120 V. This voltage headroom is needed to accommodate low line voltage conditions.

A current source isolates the regulator from the raw power supply, and prevents any signal-related currents from contaminating the power supply lines. It also serves as a current limiter in the event the amp load disappears. In that case, the entire current goes through the shunt regulator, which would dissipate about 35 W, so a large heatsink is strongly recommended to prevent it from blowing up. All the active devices except the SPX431 are mounted on the heatsink.

There are some gyrations in the output voltage at turn-on and turn-off, but because the amplifier circuit is fully differential, the plus and minus output voltages remained well matched, limiting turn on and turn off thumps.

This is not the lowest impedance supply. Since the SPX431 is only reacting to a small proportion of the overall voltage, the DC impedance is around 40 Ω . However, the amplifier circuit has a pretty constant DC draw so this is not a major drawback. Over much of the audible spectrum the dynamic impedance is

dominated by the on resistance of the 19NM50N, which is 0.25 Ω maximum.

At the highest audio frequencies and above the dynamic impedance is dominated by the impedance of the metallized polypropylene power supply output capacitors. Since the amplifier circuit is very high impedance, this simple regulator still looks pretty close to an ideal voltage source.

In most amplifier circuits, the bulk of the current variation occurs in the output stage. However, in this circuit, the output stage is well isolated by current sources and sinks. This leaves only the input differential amps, so a few extra milliamps in the regulator beyond the requirements of the active circuit should suffice. The normal power dissipation in the regulator is about 5 to 6 W. The broadband noise between B+ and B- is less than 2 mV RMS, which is adequate, if not spectacular.

Filament Supplies

The filament supply for the input stages should be able to supply at least 1.2 A (1.3 A if you use 5751 tubes for the top of the cascode) and sits at about +60 V. The output stages need at least 1.2 A sitting at -310 V. You *must* use separate filament supplies for the input and output stage tubes to avoid exceeding the maximum filament to cathode ratings for the tubes. Exceeding this limit can significantly shorten tube life. I used a separate filament transformer for the rectifier tube. This filament supply is tied to the positive HV line.

Bias Supply

Electrostatic headphones need a bias supply. A simple voltage doubler supply with small 0.1 μF filtering capacitors is more than sufficient. The bias supply is a copy of Kevin Gilmore's design (see **Figure 1**). Don't use large capacitors in the bias supply, and *do not* use a large capacitor after the isolating resistors. This could fry the headphone diaphragms if it discharges into them. I used two 2.2 M Ω , 0.5 W isolating resistors in series to achieve adequate voltage ratings, or you can use a 5 M Ω , 600 V rated resistor.

Parts

I used 0.25 W metal film resistors for the current sources, 0.5 W, 350 V rated metal film resistors for the active circuit, and Panasonic 1,000 V film capacitors for the headphone bias supply, mostly obtained from Mouser. I chose Audiocap Theta polypropylene coupling capacitors because of their sonic reputation and compact size. Whatever coupling capacitors you choose should be at least 600 V rated, and 800 V or higher is safest, in the event of

About the Author

James Lin became interested in DIY audio by reading *The Audio Amateur*. He has written modification articles for *The Audio Amateur*, *Speaker Builder*, *Glass Audio*, and *audioXpress*. In real life, he is a doctor at the Veterans Administration in Albuquerque, NM.

a catastrophic tube short. Each channel was built using point-to-point wiring on terminal boards from Welborne Labs. I bought the shielded transformers and chokes online.

The output current loads used heatsinks made for solid-state relays (SSRs). I also used small finned heatsinks for the output cathode current sinks. For the power supply, I used a heatsink and electrolytic capacitors from my parts box, and Panasonic and Solen film capacitors in the high voltage regulator. The volume control was a Daven 100 k Ω dual-stepped attenuator, which I bought online some years ago. DACT also makes stepped attenuators. I purchased the Teflon output sockets from HeadAmp.

Tubes

Although new manufacture tubes can be used, good new-old-stock (NOS) tubes are still available at reasonable prices. A potential advantage of NOS tubes is long-term reliability, since they were made in the days when tubes were the only amplification devices. Many tube aficionados also prefer the sound of NOS tubes to new newly manufactured ones.

The 1985 vintage Phillips 6SN7WGTA tubes used during circuit debugging sometimes saw DC cathode-to-plate voltages over 600 V for a minute or more at a time on several occasions without any problems. *Do not use 5692 tubes or 6SN7 tubes without the A or B suffix.*

Construction

The SRX Plus Hybrid Amplifier's construction is detailed in **Photos 2–4**. I placed the shelf for the raw power supply at the front of the chassis. Then I soldered the HEXFRED diodes directly to the 6BY5GA octal socket. I used a second shelf for the power supply current source and regulator MOSFETs heatsink, the regulator circuit board, and the headphone bias supply (see **Photo 2**).

I mounted the amplifier boards on a third shelf with the input volume control (see **Photo 3**). The constant current sources and loads were built as mini circuits (see **Photo 4**). I used a heatsink output channel's current loads.

I built the power supply first (see **Photo 2**). To test the high voltage section, I soldered four 20 k Ω , 20 W resistors two each in parallel, and then in series for a total power dissipation of 80 W as a dummy load.

Once I built and tested the raw voltage supply, I connected it to the high voltage regulator supply to test the regulator, using the same dummy load that I used for the raw supply. Next, I adjusted the trimmer pot on the current source to 41–42 mA (0.41–0.42 V across the 10 Ω sense resistor).

Before connecting the boards to the high power

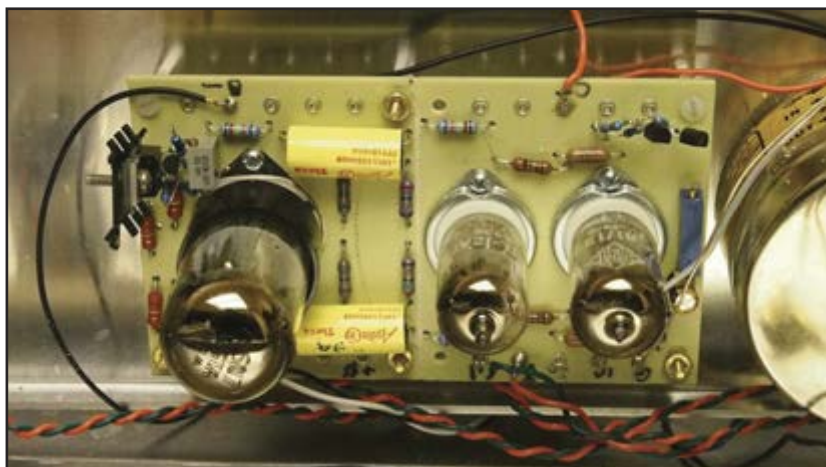


Photo 3: Here is the amplifier board and the volume control.

supply, I connected a low voltage source to each current source on the board (one or two 9 V batteries in series will do), and adjusted each source to the correct current using a digital voltmeter across the 100 Ω sense resistors. The input stage tail current sink should be set to 1.3 mA (0.13 V), the output plate source current loads should be set to 7 mA (0.7 V), and the output tail current sink should be set to 17 mA (1.7 V).

Testing, Debugging, and Adjustment

There is always a moment of anxiety when first powering up. Most electronic DIY articles are written as if everything always works perfectly. In fact, about half the things I've built worked perfectly when I

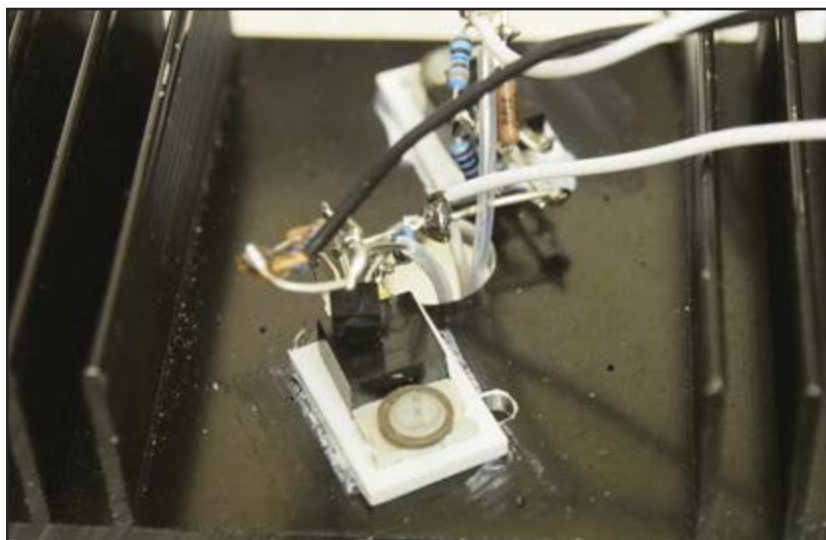


Photo 4: Detail of the amplifier heatsink with current sources is shown.



first flipped the switch. It is helpful to use a variac to power up initially—that way if there is a problem, you should catch it before things get too bad. At a minimum, you will need a voltmeter with clip and probe leads and an insulated screwdriver.

Before powering up, set the first stage balance pots in the middle and set the bias pot to minimum. The first step is to recheck the high voltage positive and negative supplies to insure that they are correct.

After confirming the power supply voltages, I adjusted the amplifier boards. The output section should be adjusted first. Clip the negative voltmeter lead to chassis or ground and insert the positive probe into one of the sockets in the output jack with the power off. Then, turn on the amplifier and adjust the output tail current sink.

If one output plate voltage is around +300 V and the other output plate voltage is -300 V, or if both voltages are negative, increase the pot resistance. If both voltages are positive, decrease the pot resistance. The amplifier took 30 minutes for the output voltages to settle down, starting at +50 V and drifting toward zero as it warmed up. After the first 20 minutes, all the plate voltages were less than 10 V above ground.

Amplifier (per channel)

Note: all wattage and voltage ratings are minimum

Quantity Part

Capacitors (4)

- 2 0.1 μ f/630 V
- 2 100 μ f/6 V, electrolytic

Resistors (35)

- 9 100 Ω , 0.25 W gate stoppers for DN2540, and sense resistors
- 1 102 Ω , 0.25 W (source resistor for output cathode sink)
- 2 250 Ω , 0.25 W (source resistor for output current load)
- 2 500 Ω , 0.5 W
- 3 1 k Ω , 0.25 W gate stopper for 10M90S
- 1 1.43 k Ω , 0.25 W (source resistor for input current sink)
- 2 2 k Ω , 0.5 W
- 2 5.1 k Ω , 1 W/500 V
- 1 100 k Ω , 0.5 W
- 4 124 k Ω , 0.5 W, 350 V rated
- 2 250 k Ω , 0.5 W, 350 V rated
- 4 220 k Ω , 2 W, 500 V rated
- 2 510 k Ω , 0.5 W, 350 V rated

Solid State (8)

- 5 Supertex DN2540 N-Channel depletion mode MOSFET (TO92 or TO220)
- 3 IXYS IXCP10M90S current source (TO220)

Trimms (5)

- 1 20 Ω multiturn trimmer for output current sink—set current to 17 mA
- 2 100 Ω multiturn trimmer for output current load—set current to 7 mA
- 1 500 Ω multiturn trimmer for input stage current sink—set current to 1.33 mA
- 1 5 k Ω , 0.75 W multi-turn trimmer for input stage balance

Tubes (3)

- 2 12AT7, may also use 12AX7 or 5751 for top tube only
- 1 6SN7GTA or 6SN7GTB

(Author's Note: Kevin Gilmore has designed a circuit board for the amplifier with two changes—a bipolar 100 μ f capacitor across the 5 k Ω balance adjustment pot for the input stage, and a balanced input. It is available at his website: <http://gilmore.chemo.northeastern.edu/boards>. The board file is under srx6/zip. This file can be sent to a circuit board maker to make the board.)

Resources

J. Broskie, "Shunt Regulators, *The TubeCAD Journal*, June 2007, www.tubecad.com.

—, "Tube Shunt Regulators," *The TubeCAD Journal*, July 1999, www.tubecad.com/july99/page12.html.

F. Langford-Smith, *Radiotron Handbook*, Chapter 30: "Rectification with a Choke Input Filter," RCA Victor Division, Radio Corporation of America; 4th edition, 1953.

Bias Supply

Quantity Part

Capacitors

- 4 0.1 μ f/1,000 V

Resistors

- 1 1 k Ω , 0.25 W
- 1 3 k Ω , 0.25 W
- 2 2.2 M Ω , 0.5 W/600V

Solid State

- 2 1,000 V/1 A diode
- 1 100 V Zener diode
- 1 130 V Zener diode
- 2 180 V Zener diode
- 1 IXYS IXCP10M90S current source

Raw Power Supply

Quantity Part

Capacitors

- 2 47 μ f/450 V, electrolytic
- 2 330 μ f/450 V, electrolytic

Chokes (minimum)

- 2 10 H or higher/50 mA
- 2 3H or higher/50 mA

Fuse

- 1 A, slow-blow

Solid State

- 2 1,200 PIV/1 A or better HexFRED diodes

Transformer (minimum)

- 1 850 V/50 mA
- 1 6.3 V/1.2 A
- 1 6.3 V/1.3 A
- 1 6.3 V/1.6 A
- 1 5 V/1 A

Tube

- 1 6BY5GA

High-Voltage Regulator

Quantity

Part

Capacitors

1	2.2 μ f/50 V, polyester
2	0.1 μ f/1,000 V, polypropylene
1	100 μ f/16 V, electrolytic capacitor
2	22 μ f/630 V, metallized polypropylene

Resistors (0.5 W unless otherwise specified)

1	10 Ω sense resistor
1	30 Ω source resistor for DN2540
1	100 Ω , 0.25 W gate stopper
1	200 Ω , 1 W resistor
3	1 k Ω , 0.25 W gate stopper
1	10 k Ω
1	15.4 k Ω
1	82 k Ω
2	110 k Ω /1 W
2	365 k Ω /2 W
1	590 k Ω /2 W
8	1 M Ω

Solid State

1	IXYS IXCP10M90S current source (TO220)
1	DN2540N5-G MOSFET (TO220)
2	STP19NM50N MOSFET (TO220)
1	Exar SPX431 shunt regulator
1	15 V/3 W Zener diode

Trimmer

1	10 Ω trimpot for current source (adjust to give 0.41 to 0.42 V across the 10 Ω sense resistor)
---	--

Re-adjust the output section after the amplifier has been on for about 30 minutes, then check it again at one and two hours. I found that there was a few volts variation over several hours, but the output plate voltages stayed well within 10 to 15 V of zero.

Next, set the balance of the input stage by attaching the positive and negative voltmeter test leads to the input ends of the coupling capacitors, and adjust the input stage balance pot to 0 V. I like to run in a circuit for around 100 hours to stabilize it, with several on-off cycles in that period of time.

Measurements

Using a Fluke 189 digital multi-meter (DMM), the frequency response at 20 V RMS was flat from 20 Hz to 20 kHz, and 1 dB down at 50 kHz in both channels. With a 100 pf polypropylene cap dummy load, the frequency response at 100 V RMS was 3 dB down at 46 kHz. At 200 V RMS, it was about 0.7 dB down at 20 kHz and 1.5 dB down at 30 kHz. At 300 V RMS the frequency response was about 1.5 dB down at 20 kHz and almost 5 dB down at 30 kHz. The unweighted hum/noise was around 3 to 5 mV in one channel and 3 to 4 mV in the other. I did not hear any noise or hum with either of my headphones. The power consumption was about 66 W.

For this project, a 1 A slow blow fuse is adequate. After several hours running time, I used an infrared thermometer to measure the output stage current sinks. The reading was at about 75°C to 80°C. The output stage current source heatsinks measured about 50°C. The power supply heatsink's temperature was about 45°C.

Sonics

The original SRX has a solid reputation for good sonics. Given the objective improvements in performance with the modifications described, there is no reason why it shouldn't sound even better than the original. However, I have not compared it with the original version or with other electrostatic headphone amplifiers. It does sound very good to me, but then a baby is always beautiful to its own parents.

When I started in audio, I was a member of the Boston Audio Society, a group that an audiophile magazine once sneered, "wants audio nirvana for \$79.95."

Yup, that's me. I'm not a Class-A type guy, I go for Class B, "Close to the best at a reasonable price." I believe that the SRX Plus fits that category. 

Author's Acknowledgement: I want to thank Kevin Gilmore for valuable input and for designing a circuit board for this amplifier.



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A Low-Cost High-Performance IC Audio Preamplifier



I really like valve audio equipment and all my usual components are valve based. However, I am not adverse to designing solid-state equipment. The preamplifier described in this article was borne out of the need for something compact that did not generate a lot of heat.

Photo 1: This is the exterior of the slightly larger deluxe version of my high-performance, integrated circuit (IC) audio preamplifier.

By
Bruce Heran
(United States)

My electronics design area is small and tube gear heats it up far too rapidly. But, I wanted to listen to records in my shop on headphones. Since the turntable already took up a lot of space, the preamplifier had to be fairly compact. In the end, I made two versions. One is a nice looking wood-cased version (see **Photo 1**). The other is a compact, but more-limited-in-features mini version (see **Photo 2**).

The mini version is a bit crowded. If I had to do it again, I would rearrange the components so everything would fit on one board. However, there are some areas in which I refuse to compromise. Audio equipment must be easy to operate, have a superior signal-to-noise ratio (SNR), and sound great. Considering the preamplifier's simplicity, I believe it accomplishes these things. Using all new components, I would expect a DIYer to be able to make the mini version for approximately \$100. The "deluxe" version would likely cost about \$75 more, largely depending on the cost of the case and added features.

The Design

The design is rather straight forward as anyone familiar with integrated circuits (ICs) will discover. However, there are some design features that, in my opinion, separate it from similar designs. First and most obvious is the use of modular DC to DC converters. Both versions use a Mean Well DCW03A-12 to obtain clean symmetrical well regulated positive and negative 12 V for the amplifier ICs. These converters allow an input voltage of between 9 and 18 VDC. It need not be regulated and doesn't really have to be all that clean. A wall wart will do fine, and that is what I used in the mini version. The deluxe version uses a second Mean Well converter (a PM-05-12) to convert 120 to 240 VAC into 12 VDC (see **Photo 3**). It is more compact and more versatile than pretty much anything else that could be used. Both converters are inexpensive and available from several sources. Using the converters greatly simplifies the design and the configuration of the preamplifiers.

The Circuit

The actual circuit uses two stages. The first stage is configured to have two gain settings. They are approximately 40 and 55 dB. It also is designed to have two equalizations. One is for vinyl reproduction with the Recording Industry Association of America (RIAA) curve and one for tape head playback with the National Association of Broadcasters (NAB) curve. In the case of the RIAA equalization, the values computed by common computer programs were slightly different than the final ones determined through testing on the bench. The schematic shown in **Figure 1** details the bench determined values.

I used Burr Brown OPA2134 high-performance audio ICs. Other ICs may be used, but not all of them will perform satisfactorily. Three factors need to be considered. First, the property of the IC known as gain bandwidth product must be sufficiently high to allow the rather large amount of gain and high frequency response desired. The second factor is stability. With a high gain bandwidth capable IC, it is possible to reach unity or lower gain through the capacitors in the feedback loop at some extreme frequency above the audio band, particularly when using the high gain setting. To compensate for this, I included the 100- Ω resistor in the loop to



Photo 2: The top view of the mini version is shown here.



Photo 3: The rear view of the deluxe version shows the inputs and the outputs.

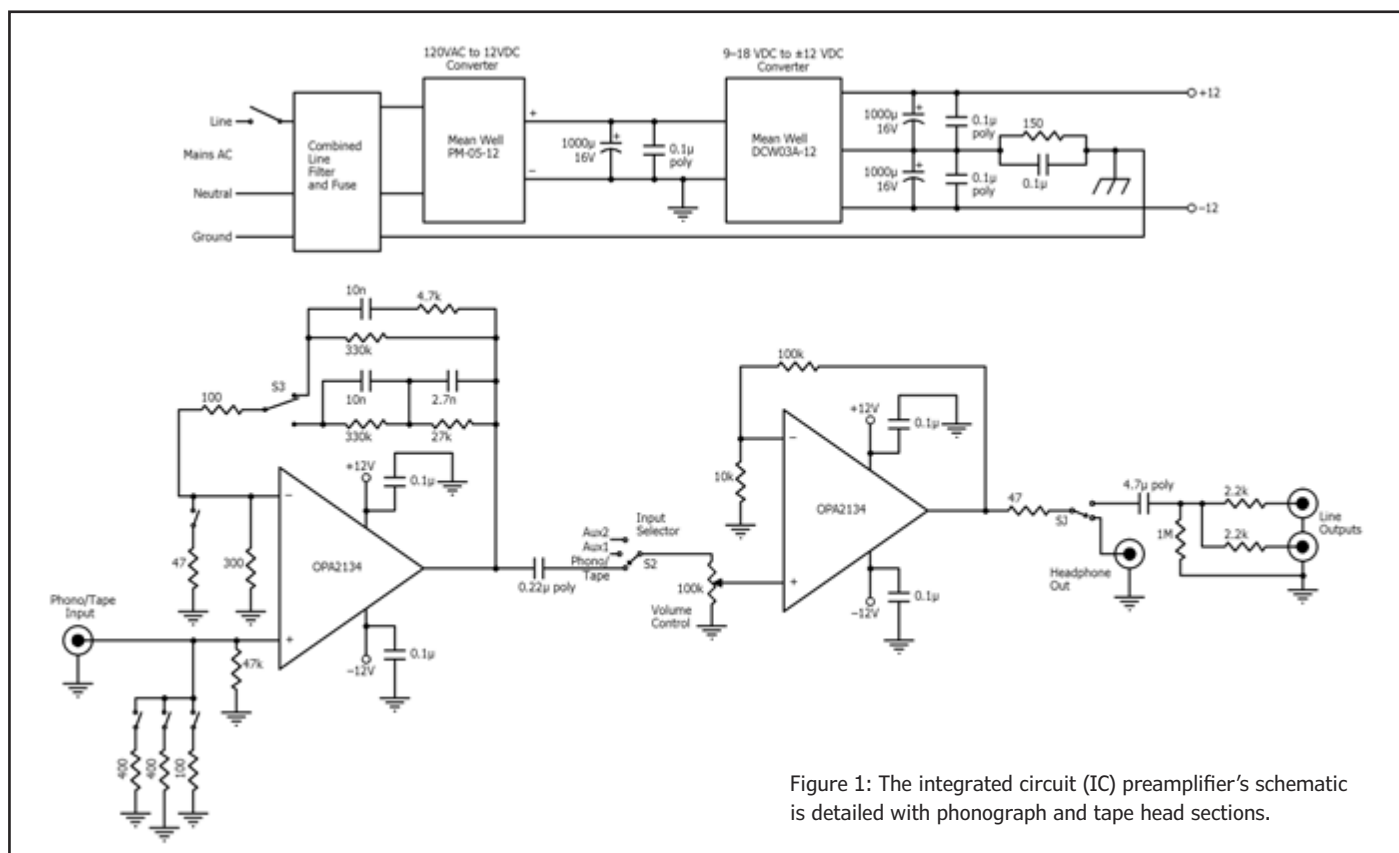


Figure 1: The integrated circuit (IC) preamplifier's schematic is detailed with phonograph and tape head sections.



Photo 4: The interior design of the deluxe version is rather straight forward.

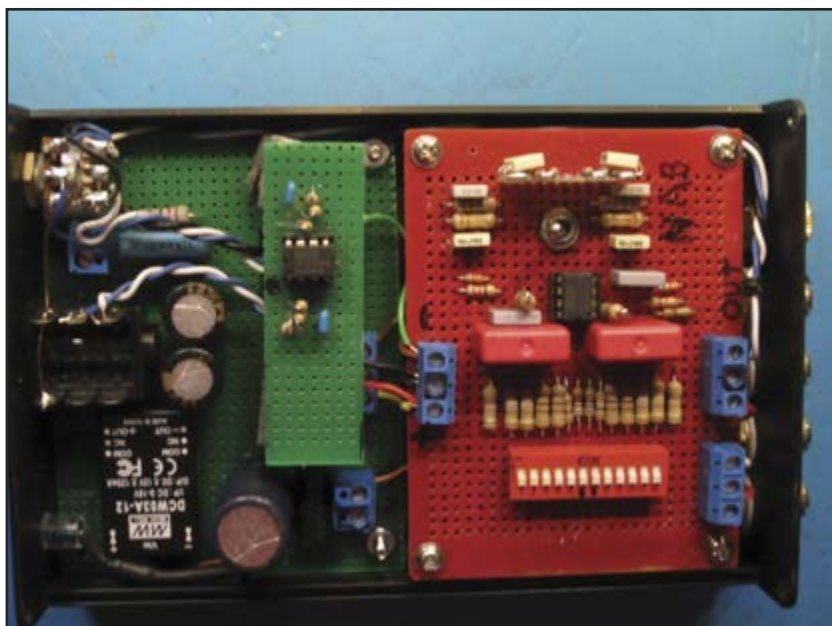


Photo 5: The interior design of the mini version is a bit more crowded.

About the Author

Bruce Heran lives in Sierra Vista, AZ, and has been involved in electronics for 53 years. He built his first project—a one-transistor amplifier—when he was 13. He has a BS in Biology and studied electrical engineering for two years. Electronics has always been his hobby. In May 2011, he retired from his work as a project manager of a logistics contract with the federal government. Now he is co-owner of Oddwatt Audio, an electronics company that specializes in vacuum-tube audio components. He is vice president of design and support, and many of the company's products are kits and assembled audio equipment based on his personal designs.

pretty much insure that the condition cannot exist. In many cases, it can be omitted and or vary in size up to about 1,000 Ω , with negligible effect on performance. The third and final consideration is the IC's ability to deliver some current to the load. While the OPA2134 is no powerhouse, it can deliver sufficient current for use by many headphones. The mini preamp is used just for that purpose, and it does a great job on my Sennheiser and Koss headphones. One advantage of the low-output impedance and high-current capability is that the preamplifiers can easily drive virtually any power amplifiers.

Preamplifier Construction

Building either preamplifier is fairly easy. With the exception of the AC power input section in the deluxe model, all voltages are low. I used perf boards with solder pads. Normal consideration should be given to signal grounding, wire routing, and such, but it is generally not critical. If a shielded case is not used, then any of the input signal bearing conductors longer than about 3 cm should be shielded.

I like to have all the signal grounds and the shields—but not the chassis itself—connected at a single point. This is essentially a “star” arrangement. The point I usually choose is at the ground side of the input jacks. Be sure that they do not contact the chassis as a ground loop hum will occur. I have found that for effective shielding and the prevention of ground-loop-based hum, the chassis should be separated from the signal ground. There are several ways to connect the chassis to the signal ground. I prefer the use of a resistor in the 120-to-150- Ω range with a parallel 0.1 to 0.22 μF type X2 capacitor. This is important in the deluxe version as the AC line filter ground is connected to the chassis as a safety feature (see **Photo 4**). The smaller interior of the mini version is a bit more crowded (see **Photo 5**).

Each IC needs to have the positive and the negative voltage terminals bypassed right at the IC. I use low voltage 0.1 μF ceramic or poly capacitors there. All the electrolytic capacitors should also be similarly bypassed. In the deluxe version, I put the 120 VAC to DC converter, a line filter with a fuse, an International Electrotechnical Commission (IEC) power connector, and the AC power switch inside a metal shield. This was partly for electromagnetic interference (EMI) considerations and partly for safety reasons (see **Photo 4**).

I used quality coupling capacitors and components throughout the preamps. The ones in the negative feedback loop on the first stage need

Parts List

Most components are available from all of the larger parts houses

Part	Part Number	Source
Meanwell AC to DC converter	PM-05-12 or IRM-05-12	(Jameco, \$9.95 each)
Meanwell DC to DC converter	DCW03A-12	(Jameco, \$9.95 each)
Integrated circuits	OPA2134	
IEC Line filter and fuse	Delta Electronics 04BEEG3S	(Jameco, \$1.29 each)
Capacitors are by WIMA and Vishay for "poly" types		
Capacitors are generic for electrolytic types		
The X2 capacitor		Vishay
Resistors are carbon film 0.25 and 0.5 W, 5%		
Volume control in the deluxe model		Alps
Volume control in the mini version		RadioShack
DIP switches		C&K Components

to be fairly close tolerance. Five percent tolerance or better is needed to insure the equalization is correct. I bought a batch of the resistors and capacitors and hand tested them for tolerance and selected the ones that most closely matched the need. Dual in-line package (DIP) switches with various resistors provide for several values of input loading. This is important for low output moving coil cartridges. A standard 47 k Ω load is included for moving magnet cartridges. When used as a tape play back preamplifier the 47 k Ω value is suitable for many tape heads however you may have to change it to accommodate the ones you have.

Performance Results

So how well does it perform? Several listeners have confirmed my own evaluation. It is far better sounding than you would expect from its modest design. With hand-selected components, the phonograph response was within ± 0.1 dBV from 20 Hz to 30 kHz (yes, I know the standard officially stops at 20 kHz) and only down by 1 dBV at 10 Hz.

The line stage was better and flat to almost 100 kHz. The SNR was over 90 dBV in any gain configuration. Distortion was below my ability (0.15%) to measure. The

deluxe version was able to reproduce rather clean 50 kHz square waves at the 8 V RMS level into 50 k Ω .

I conducted some listening tests with the deluxe version in my main system. To do the tests, I used a custom Dual 701 turntable that has an Origin Live arm and AT33PTG/II cartridge for the low output moving coil listening. I used a Sota turntable with an Origin Live arm with a Grado Reference Platinum I cartridge for the moving magnet setting (yes, I know it is a moving iron cartridge), and an Akai 215D reel-to-reel tape deck modified for direct output from the tape heads for the NAB listening. The remainder of the system consists of KT120 Class A monoblocs, MartinLogan electrostatic main speakers, and MartinLogan powered subwoofers.

In each case, the sound was clean, well detailed, and had a wide sound stage. The background noise was inaudible. I would characterize it as a rather easy-to-listen-to all-around nice sounding preamplifier. It was particularly good on the Akai. And, it was far quieter than the built-in amplifiers.

Is it a giant killer that will outperform my custom all-valve preamp? No, but for a modest system, you will have to pay a lot more money to reach the same level of performance. 



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Sources

DCW03A-12 and PM-05-12 Output converters
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Burr Brown OPA2134 IC
Texas Instruments, Inc. | www.ti.com

Creativity, Expression, and Innovation

An Interview with Tlacacl and Tenoch Esparza, Sunhouse Founders

By
Shannon Becker
(United States)



Tlacacl Esparza learned how to program and do things with digital audio signals and decided to go into business with his brother Tenoch.

Tenoch Esparza started learning about audio when the two brothers began their music tech company, Sunhouse (<http://sunhou.se>).

Sunhouse's first product, Sensory Percussion, was successfully funded on Kickstarter and managed to create genuine excitement within the drumming and percussion community. The solution combines the expressive nature of drumming and the essence of performance with samples, synthesizers, and digital audio effects to create totally new music styles, simply by playing the drums.

SHANNON BECKER: Tell us a little bit about your background.

TLACAEL ESPARZA: I grew up in eastern Los Angeles, CA, and studied music from a young age —first piano, then the drums. I went to the LA County High School for the Arts where I studied jazz drums and first became serious about playing as a profession.

After that, I moved to New York to play jazz and get a degree. I got BA degrees from Columbia University in music and math and then just played in indie rock bands and toured for a few years. In 2011, I decided to go back to school for my master's degree at New York University's Music Tech program. That's where I learned how to program and do things with digital audio signals.

TENOCH ESPARZA: Like Tlacacl, I started studying music at a young age, though I stuck with piano and ended up studying at the Moscow State Conservatory in Russia after high school. After some musical twists

and turns, I ended up changing careers, getting a BA in literature from Stanford University, working at Google for a while, and then earning an MBA from Columbia University in New York.

SHANNON: How did you become interested in audio?

TLACAEL: I became interested in audio purely as a performer. Recording in a nice studio with my first band Frances was my first exposure to working with audio and it just grew from there. I started adding electronics to my drums and making my own recordings. Once I learned how to program, I started building my own systems for controlling electronics.

TENOCH: I really started learning about audio when Tlacacl and I began working together on Sunhouse. As a startup, everything is DIY. So one of the responsibilities that fell in my lap was shooting and editing video for demos. I got a serious crash course in how to record, edit, and use good audio.

SHANNON: What made you decide to go into business together?

TENOCH: We both finished graduate school at the same time in New York and were looking for what to do next. I always wanted to start a company and Tlacaél didn't want to have a desk job. He had this idea for a completely new system for electronic drums that sounded like a real challenge to make—not only would it require some serious engineering research on the software side, but it also had a hardware component.

We talked about it for awhile and decided it was the right time for both of us, and a meaty enough challenge to pursue. As for working together as brothers, it never seemed like a negative.

SHANNON: Where did you get the idea for your company, Sunhouse (<http://sunhou.se>)?

TLACAEEL and TENOCH: Our company was initially called Sensory Percussion, Inc., just like the product. But we quickly realized that we'd like to extend the technology beyond drums in the future, so we looked for an umbrella name that would cover all kinds of products. And Sunhouse was born! The name is a reference to the Aztec myth of how music came to the earth.

SHANNON: Tell us about your Sensory Percussion product. What was the inspiration behind it?

TLACAEEL and TENOCH: Sensory Percussion is a new system that creates a bridge between acoustic drum performance and control of electronic sound. It's a hardware/software system where a small sensor attaches to the side of your drum and feeds our software which then can pick apart the sound of your drum to tell where and how you're hitting it. This enables you to map different sounds and effects to follow your playing in real time.

TLACAEEL: I was inspired to create this system after becoming frustrated with the limited capabilities of the available systems for electronic drums on the market and the complete lack of progress in the field over that last couple decades. Trigger systems, drum pads, and e-drums are mostly designed with a really simplistic view of what it means to play the drums as an instrument.

Though you can play beats on a sample pad or trigger a sample from your snare, the kinds of music that electronic musicians are making with sequencers, midi-controllers, and software has completely



outpaced what real drummers can do. So Sensory Percussion was really born out of my own desire to do more with the electronics behind the drums.

SHANNON: How does Sensory Percussion work?

TLACAEEL and TENOCH: Sensory Percussion uses a custom hardware sensor that attaches to any drum in a standard drum kit (snare, toms, and kick) and picks up on the vibrations that result from playing the drum. It's essentially a microphone, but dialed in to get a very clean, isolated sound from your drum. The sensor feeds the software, which then analyzes the sound of each hit and enables you to map electronic sounds and effects to follow your playing.

There is a quick calibration process where you hit

The Sensory Percussion is a new system that creates a bridge between acoustic drum performance and control of electronic sound.



This is a side view of the Sensory Percussion system, which was successfully funded on Kickstarter, raising \$94,245 with 328 backers.



Sensory Percussion is designed to understand where and how you hit the drum. It not only lets you map different parts of the drum to any sound desired, it also lets you control those sounds.

your drum in various ways—this lets the software learn the sonic range of your drum. From there, you simply drag and drop samples, synths, and effects to be controlled by different sounds of your drum through our software's user interface. It's easy to use, but lets you control electronics in a powerful and intuitive way that was previously not possible. Bringing up the reverb on the entire kit could be as simple as playing closer to the edge of the drum. Though we will ship the device with great presets for different sounds and effects, it's entirely programmable to function as you want it to.



Sensory Percussion is easy to use, but lets you control electronics in a powerful and intuitive way.

SHANNON: Can you share some of the challenges involved with the design?

TLACAE: There have been many. It's a fairly novel idea in the professional audio software world, so its functionality, user interface, and feature set really had to be designed from the ground up. We've gone through many iterations—the software and the user flow is completely different from even six months ago and it's still changing as we incorporate new ideas and user feedback.

One of the biggest challenges has been to create a graphical representation of the different timbres of a drum. You can hit a drum in countless ways but we've worked to distill different drum strokes and ways of hitting the drum into a standard, tractable list. Also, the sounds that you can make on a drum are not just about hitting the drum in different places. You can create two distinct sounds by hitting the drum in the very center—for instance stick shots, damped hits, and normal strokes. So just a picture of a drum doesn't quite cut it. We had to figure out a way to visually represent these sounds in a way that would make sense to drummers. Still working on it.

SHANNON: Why did you go the crowdfunding route and fund it through Kickstarter?

TENOCH: A lot of big companies use Kickstarter as a marketing tool, or a way to gauge customer interest before investing a lot of money in development. We actually needed the money from our Kickstarter to fund our manufacturing. We're a bootstrapped startup and really liked the idea of being funded by drummers, our customers, rather than investors in suits. Plus, we wanted to tap into a group of enthusiastic early adopters. This core group is going to help us make sure we deliver an awesome product. So far, the process has been really great.

SHANNON: What's next for your company?

TLACAE and TENOCH: We're completely focused on finishing development of Sensory Percussion and shipping to our Kickstarter backers and new customers in January, 2016. It's important to us that we ship a great product that drummers will love, so that's our focus until it's finished. But we're planning on building Sunhouse into a company with a range of products for all kinds of musicians, and have some great ideas on how to extend the technology behind Sensory Percussion in new directions. But we're not quite ready to share those plans just yet! 

Testing Audio ADCs and DACs



Learn more about ADCs and DACs, and the methods engineers use to procure high-quality results by correctly evaluating those two critical stages, with conventional and more evolved measurements used in audio tests.



In 2015, there's not much question about audio storage, transmission, or streaming: it's digital. Apart from rare sightings of vinyl or open-reel tapes in boutique sales or creative enclaves, audio is digital. Done right, digital audio is flexible, robust, and of very high quality. Pulse code modulation (PCM) recording, lossless surround formats, and even lossy compression (at least at high data rates) provide the soundtracks for our lives.

But, of course, sound in air is not digital. The pressure waves created by a human voice or a musical instrument are recorded after exciting a transducer of some sort, and the transducer responds with an electrical voltage that is an analog of the pressure wave. Likewise, at the end of the chain the digitized audio signal must eventually move air, using a voltage that is the analog of the original sound wave to drive a transducer that creates a pressure wave.

Near the beginning of a digital chain, we must use an ADC (analog-to-digital converter) to transform the analog electrical signal to a digital representation of that signal. Near the end of the chain, we must use a digital-to-analog converter (DAC) to transform the digital audio signal back into an analog electrical signal. Along with the transducers, these two links in the chain (the ADC and the DAC) are key in determining the overall

quality of the sound presented to the listener.

Testing Audio Converters

The conventional measurements used in audio tests can also be used to evaluate ADCs and DACs. These measurements include frequency response, signal-to-noise ratio (SNR), interchannel phase, crosstalk, distortion, group delay, polarity, and others. But conversion between the continuous and sampled domains brings several new mechanisms for nonlinearity, particularly for low-level signals. This article looks at problems seen in audio analog-

By
David Mathew
(United States)

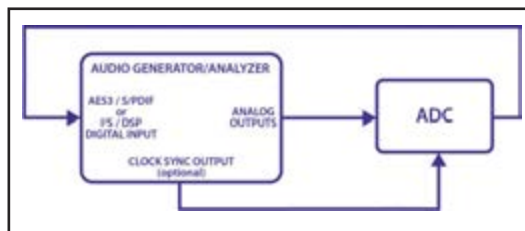


Figure 1: This is a typical ADC test block diagram.

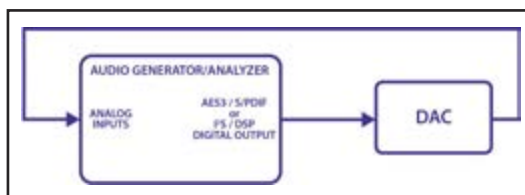


Figure 2: Here is a typical DAC test block diagram.

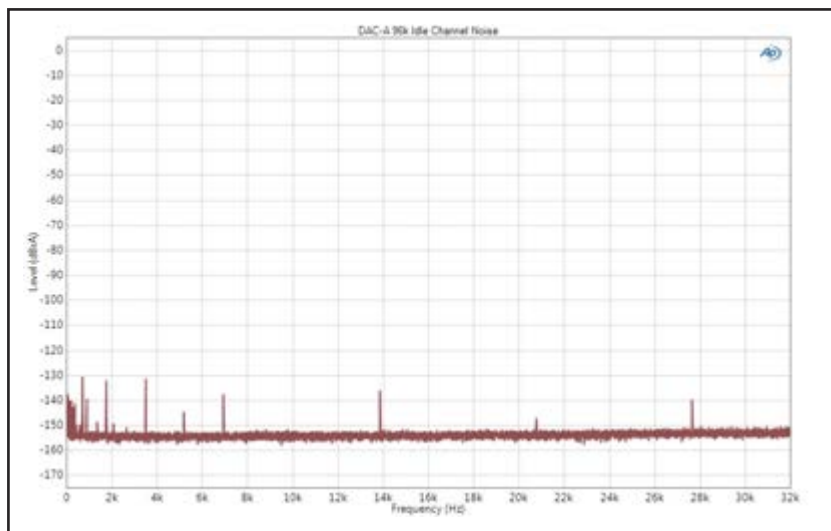


Figure 3: Here is the Fast Fourier Transform (FFT) idle channel noise, DAC "A."

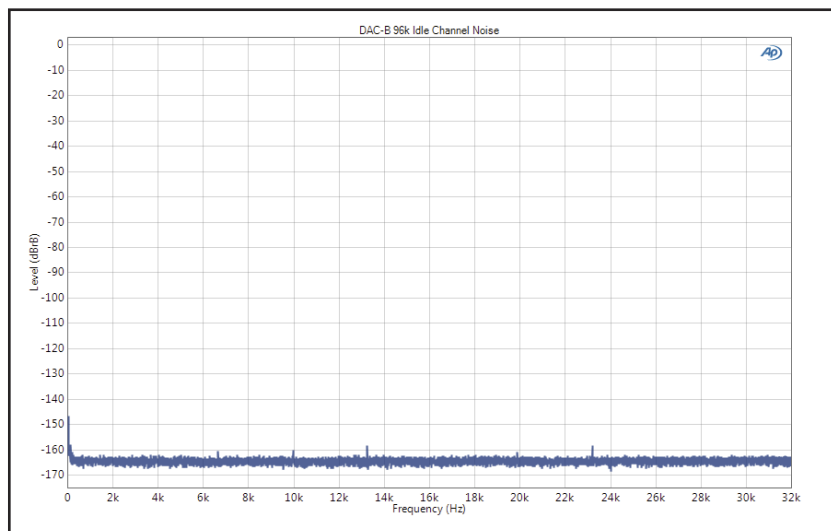


Figure 4: This is the Fast Fourier Transform (FFT) idle channel noise, DAC "B."

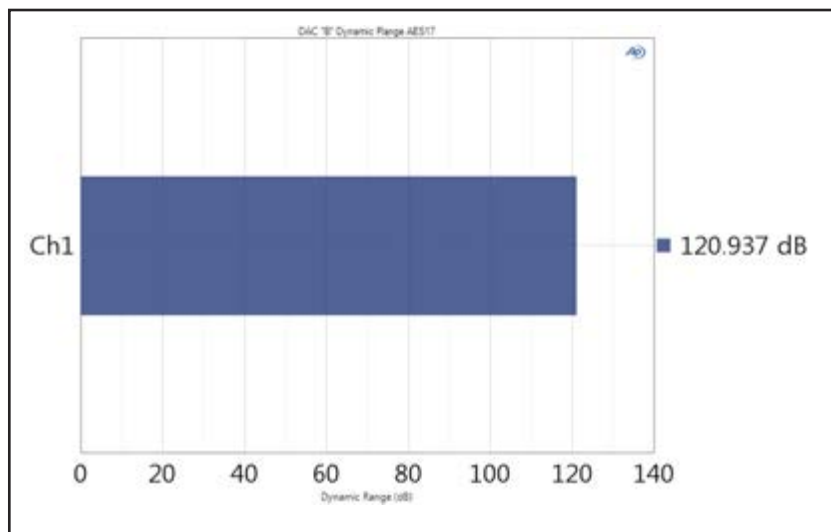


Figure 5: The signal-noise ratio (SNR) for DAC "B" is shown here.

to-digital and digital-to-analog conversion and some of the methods that have evolved to address these issues.

Of course, ADCs and DACs are used in a great number of non-audio applications, often operating at much higher sampling rates than audio converters. Very good oscilloscopes might have bandwidths of 33 GHz and sampling rates up to 100 GS/s, with prices comparable to a Lamborghini. Although audio converters don't sample at anywhere near that rate, they are required to cover a much larger dynamic range, with high-performance ADCs digitizing at 24 bits and having SNRs over 120 dB.

Even a high-end oscilloscope typically uses only an 8-bit digitizer. A 24-bit conversion pushes the measurement of noise and other small-signal performance characteristics to the bleeding edge. Consequently, measurements of such converters require an analyzer of extraordinary analog performance.

Test Setups

The typical test setups are straightforward. For ADC testing, the analyzer must provide extremely pure stimulus signals at the drive levels appropriate for the converter input. For converter ICs, the analyzer must have a digital input in a format and protocol to match the IC output, such as I²S, DSP, or a custom format.

For a commercial converter device, the digital format is typically an AES3-S/PDIF-compatible stream. For devices that can sync to an external clock, the analyzer should provide a clock sync output.

For DAC testing, the analyzer must have a digital output in the appropriate format, and analog inputs of very high performance. The graphs in this article were created by testing commercial converters, using the AES3 digital transport. The analyzer is the Audio Precision APx555.

As previously mentioned, ADCs and DACs exhibit behaviors unique to converters. The Audio Engineering Society (AES) has recommended methods to measure many converter behaviors in the AES17 standard. The following examples examine and compare several characteristic converter issues.

Idle Tones

Common audio converter architectures, such as delta-sigma devices, are prone to have an idling behavior that produces low-level tones. These "idle tones" can be modulated in frequency by the applied signal and by DC offset, which means they are difficult to identify if a signal is present. A Fast

Interpretation of Noise in the Fast Fourier Transform Power Spectra

By David Mathew

In audio systems, noise figures are generally measured and reported as:

-103 dB rms noise, 20 Hz to 20 kHz BW, A-weighted

The noise signal is measured with an RMS detector across a specified bandwidth. The 0 dB reference for the noise measurement might be a device's nominal operating level, but more typically it is the maximum operating level, which produces a more impressive number. Similarly, weighting filters usually produce better noise figures and are often used in marketing specifications. A weighting filter attempts to approximate the response that we humans perceive.

So let's measure the noise of DAC B using these methods. We'll make an RMS measurement first across the full signal bandwidth, then across a limited bandwidth, and with an added weighting filter.

-120.3 dB RMS noise, DC to 48 kHz.

-123.9 dB RMS noise, 20 Hz to 20 kHz.

-126.2 dB RMS noise, 20 Hz to 20 kHz, A-weighted.

But if you look at the Fast Fourier Transform (FFT) spectrum (see **Figure 4** in the main article), the average noise floor appears to be about -163 dB or so. That's a big difference.

Conventionally, an audio FFT amplitude spectrum is displayed by scaling the vertical axis so that a bin peak indicates a value that corresponds to the amplitude of any discrete frequency components within the bin. This calibration is not appropriate for measuring broadband signals (e.g., such as noise power) without applying a conversion factor that depends on the bin width and on the FFT window used.

In this case, each bin is 0.375 Hz wide (the sample rate of 96 kS/s divided by an FFT length of 256K points). The window spreads the energy from the signal component at any discrete frequency, and the Y-axis calibration takes this windowing into account.

For the Audio Precision Equiripple window used here, the calibration compensates for the power being spread over a bandwidth of 2.63 bins.

This can be converted to the power in a 1 Hz bandwidth, or the power density, by adding a scaling factor in decibels that can be calculated as follows:

$$\begin{aligned}\text{scaling factor} &= 10\log(1 / \text{Window Scaling} \cdot \text{Bin Width}) \\ &= 10\log(1/2.63 \cdot 0.375) \\ &= 4.2 \text{ dB}\end{aligned}$$

To estimate the noise from a device based on an FFT spectrum, you can integrate the power density over the

frequency range of interest. For an approximately flat total noise (where the noise power density is roughly constant) it is possible to estimate the sum of the power in each bin within reasonable accuracy, by estimating the average noise power density and multiplying by the bandwidth.

Figure 4, for example has a noise floor that is approximately in line with about -163 dB on the Y-axis. The noise power density is the apparent noise floor minus the window conversion factor or:

$$-163 \text{ dB} - 4.2 \text{ dB} = -167.2 \text{ dB per Hz}$$

The integration to figure the total noise over a given bandwidth is simple, if the noise is spectrally flat. Multiply the noise power density by the bandwidth, which in this case is 20 kHz. For decibel power (dB = 10logX), this is the same as adding 43 dB = 10log(20000), as follows:

$$-167.2 + 43 \text{ dB} = -124.2 \text{ dB}$$

This calculation provides a result only 0.5 dB different from the 20 Hz to 20 kHz unweighted measurement cited earlier.

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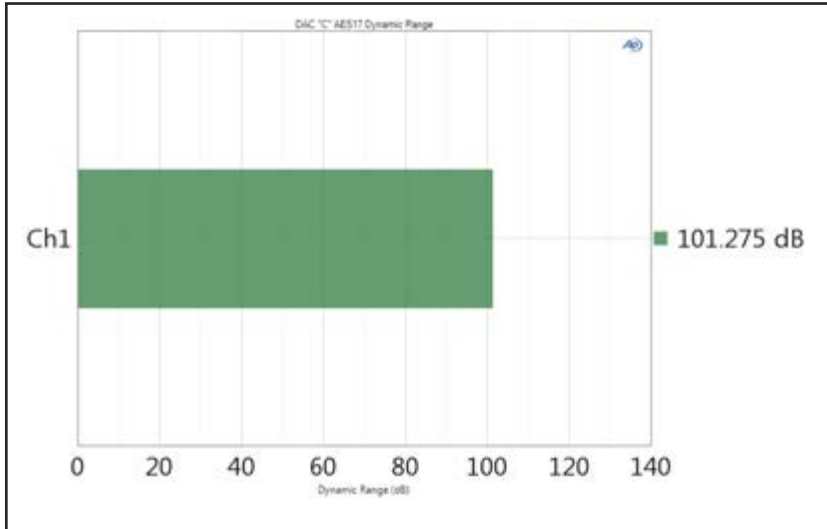


Figure 6: This is the signal-noise ratio (SNR) for DAC "C."

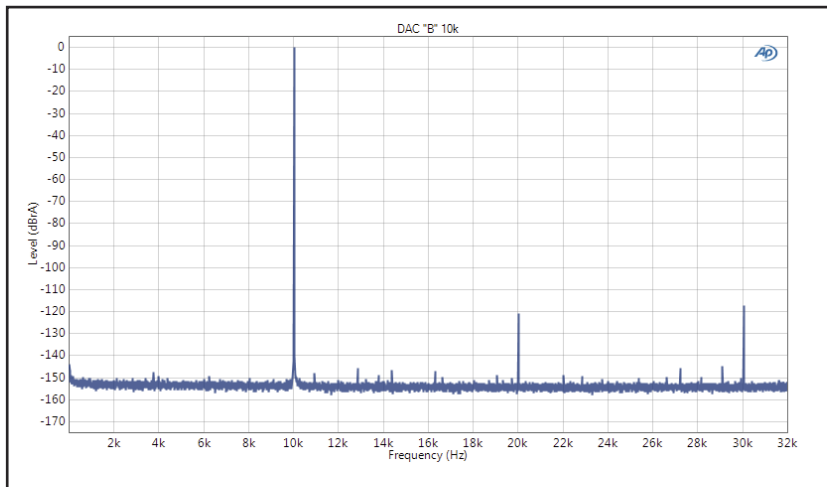


Figure 7: DAC "B" shows no jitter sidebands at 10 kHz.

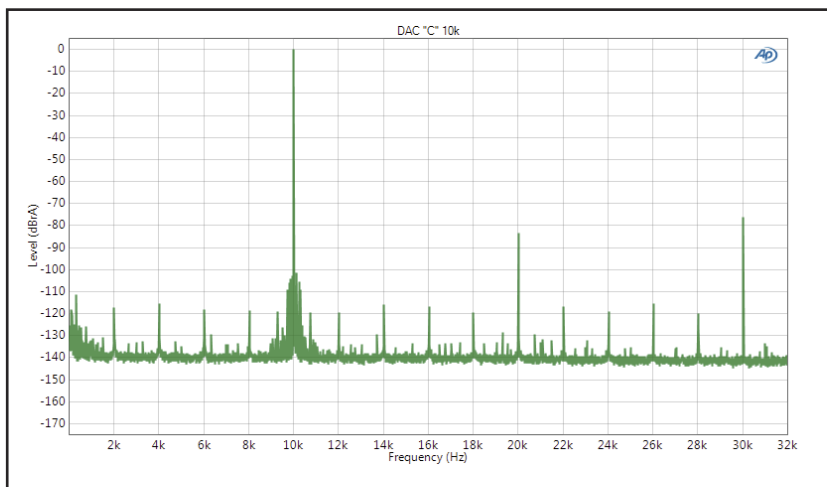


Figure 8: DAC "C" shows strong jitter sidebands at 10 kHz.

Fourier Transform (FFT) of the idle channel test output can be used to identify these tones.

The DAC shown in **Figure 3** has several idle tones, some with levels as high as -130 dB. The idle tones (and the noise floor) of the DAC shown in **Figure 4** are much lower.

Signal-to-Noise Ratio (Dynamic Range)

For analog audio devices, a SNR measurement involves finding the device maximum output and the bandwidth-limited RMS noise floor, and reporting the difference between the two in decibels.

With audio converters, the maximum level is usually defined as that level where the peaks of a sine wave just touch the maximum and the minimum sample values. This is called "full scale" (1 FS), which can be expressed logarithmically as 0 dBFS. The RMS noise floor is a little tricky to measure because of low-level idle tones and, in some converters, muting that is applied when the signal input is zero. AES17 recommends that a -60 dB tone be applied to defeat any muting and to allow the converter to linearly operate. The distortion products of this tone are so low they fall below the noise floor, and the tone itself is notched out during the noise measurement. The IEC 61606 standard recommends a similar method, but calls the measurement dynamic range.

Figure 5 and **Figure 6** show a comparison of the signal-to-noise measurements of two 24-bit DACs operating at 96 kS/s, using this method. As can be seen, some converter designs are much more effective than others.

Jitter

For ADCs, clock jitter can occur within the converter, and synchronization jitter can be contributed through an external clock sync input. For DACs receiving a signal with an embedded clock (e.g., AES3 or S/PDIF), interface jitter on the incoming signal must be attenuated.

Sinusoidal jitter primarily affects the audio signal by creating modulation sidebands—frequencies above and below the original audio signal. More complex or broadband jitter will raise the converter noise floor. A common measurement that reveals jitter susceptibility is to use a high-frequency sinusoidal stimulus and examine an FFT of the converter output for jitter sidebands, which are symmetrical around the stimulus tone. DAC "B" shows no sidebands (see **Figure 7**). However, DAC "C" (see **Figure 8**) shows strong sidebands. Note that the strong tones at 20 kHz and 30 kHz are products of harmonic distortion and are not jitter sidebands.

Jitter Tolerance Template

AES3 describes a jitter tolerance test, where the capability of a receiver to tolerate defined levels of interface jitter on its input is examined. A digital audio signal is applied to the input. The signal is jittered with sinusoidal jitter, swept from 100 Hz to 100 kHz. As the jitter is swept, its level is varied according to the AES3 jitter tolerance template. Jitter is set at a high level up to 200 Hz, then reduced to a lower level by 8 kHz, where it is maintained until the end of the sweep.

An interface data receiver should correctly decode an incoming data stream with any sinusoidal jitter defined by the jitter tolerance template shown in **Figure 9**. The template requires a jitter tolerance of 0.25 UI peak-to-peak at high frequencies, increasing with the inverse of frequency below 8 kHz to level off at 10 UI peak-to-peak below 200 Hz.

In this case, jitter is set to about 9.775 Unit Intervals (UI) at the lower jitter frequencies, and drops to about 0.25 UI at the higher frequencies. The blue trace is the THD+N ratio (distortion products of the 3 kHz audio tone), which remains constant across the jitter sweep, indicating good jitter tolerance in this DUT. As jitter level rises, poor tolerance will cause a receiver to decode the signal incorrectly, and then fail to decode the signal, occasionally muting or sometimes losing lock altogether.

Filter Effects

Figure 10 shows the response of the anti-alias filter in ADC “C.” A tone at the input of the ADC is swept across the out-of-band (OOB) range of interest (in this case, from 40 kHz to 200 kHz) and the level of the signal reflected into the passband is plotted against the stimulus frequency. A second trace shows the converter noise floor as a reference.

For **Figure 11**, spectrally flat random noise is presented to the DAC input. The analog output is plotted (with many averages) to show the response of the DAC’s anti-image filter. In this case, a second trace showing a 1 kHz tone and the DAC noise floor is plotted, scaled so that the sine peak corresponds to the noise peak.

Polarity

Audio circuits (including converters) often use differential (balanced) architectures. This opens the door for polarity faults. An impulse response (IR) stimulus provides a clear observation of normal or reversed polarity (see **Figure 12**).

Summary

Tests for the high-level nonlinear behavior of an ADC are similar to those for nonlinearities in analog

electronics, using standardized tests for harmonic distortion and intermodulation distortion. But audio converters bring new mechanisms for non-linearity, particularly for low-level signals. AES17 and Audio Precision’s Technote 124 describe effective testing methods for audio converter measurements. 

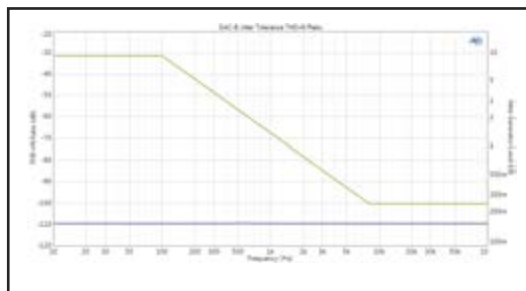


Figure 9: This is the total harmonic distortion plus noise (THD+N) for DAC “B” during the jitter tolerance sweep.

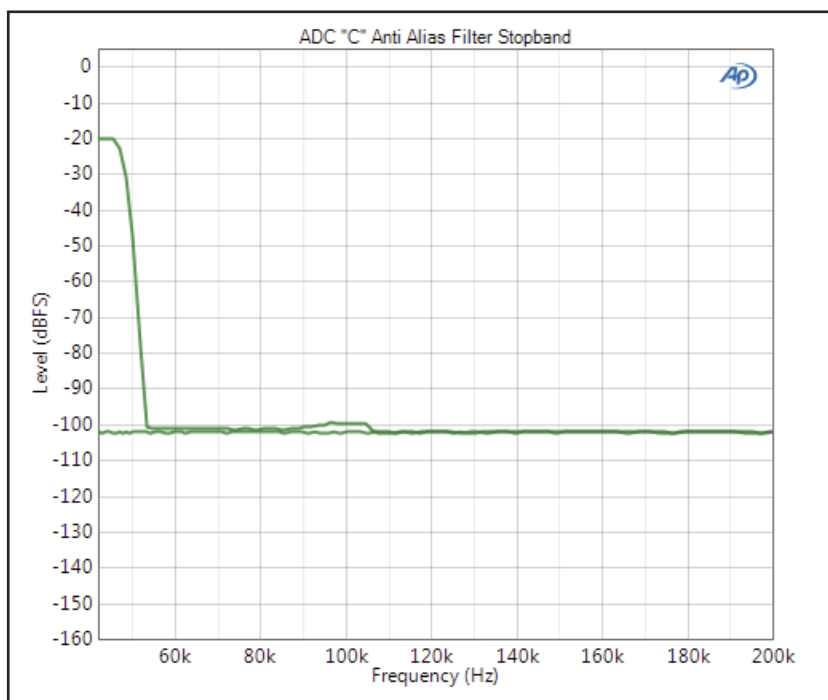


Figure 10: Here is the response of the anti-alias filter in ADC “C.”

Resources

AES3, AES standard for digital audio engineering—Serial transmission format for two-channel linearly represented digital audio data, 2009 (rev. 2014), www.aes.org.

AES17, AES standard method for digital audio engineering—Measurement of digital audio equipment, 1998, rev. 2009, (Revision of AES17-1991), www.aes.org.

J. Dunn, “Application Note #5: Measurement Techniques for Digital Audio,” Audio Precision, 2004, www.ap.com.

S. Peterson, “Technote 124, Measuring A-to-D and D-to-A Converters with APx555,” Audio Precision, 2015, www.ap.com.

Glossary

AES3, S/PDIF—In the consumer and professional audio field, digital audio is typically carried from point to point as a bi-phase coded signal, commonly referred to as AES3, AES/EBU or S/PDIF. There are electrical and bitstream protocol differences among the variations of bi-phase coded digital audio, but the various signals are largely compatible. Variations are defined in the standards AES3, IEC 60958, and SMPTE 276M.

anti-alias filter—In sampled systems, the bandwidth of the input has to be limited to the folding frequency to avoid aliasing. Modern audio ADCs normally have this anti-alias filter implemented with a combination of a sharp-cutoff finite impulse response (FIR) digital filter and a simple low order analog filter. The digital filter operates on a version of the signal after conversion at an oversampled rate, and the analog filter is required to attenuate signals that are close to the oversampling frequency. This analog filter can have a relaxed response, since the oversampling frequency is often many octaves above the passband.

anti-image filter (reconstruction filter)—Digital audio signals can only represent a selected bandwidth. When constructing an analog signal from a digital audio data stream, a direct conversion of sample data values to analog voltages will produce images of the audio band spectrum at multiples of the sampling frequency. Normally, these images are removed by an anti-imaging filter. This filter has a stopband that starts at half of the sampling frequency—the folding frequency.

Modern audio DACs usually have this anti-imaging filter implemented with a combination of two filters: a sharp cut-off digital finite impulse response (FIR) filter, followed by a relatively simple low-order analog filter. The digital filter is operating on an oversampled version of the input signal, and the analog filter is required to attenuate signals that are close to the oversampling frequency.

unit interval (UI)—The UI is a measure of time that scales with the interface data rate, and is often a convenient term for interface jitter discussions. The UI is defined as the shortest nominal time interval in the coding scheme. For an AES3 signal, there are 32 bits per subframe and 64 bits per frame, giving a nominal 128 pulses per frame in the channel after bi-phase mark encoding is applied. So, in our case of a sampling rate of 96 kHz:

$$1 \text{ UI} = 1 / (128 \times 96000) = 81.4 \text{ ns}$$

The UI is used for several of the jitter specifications in AES3.

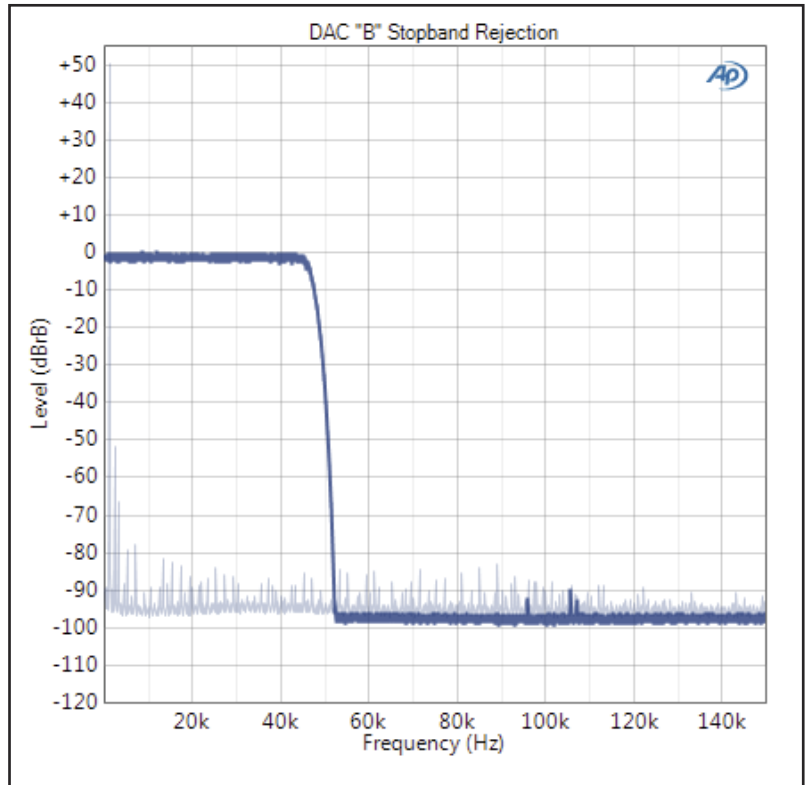


Figure 11: Spectrally flat random noise is presented to the DAC input. This figure shows the DAC "B" anti-image filter out-of-band rejection.

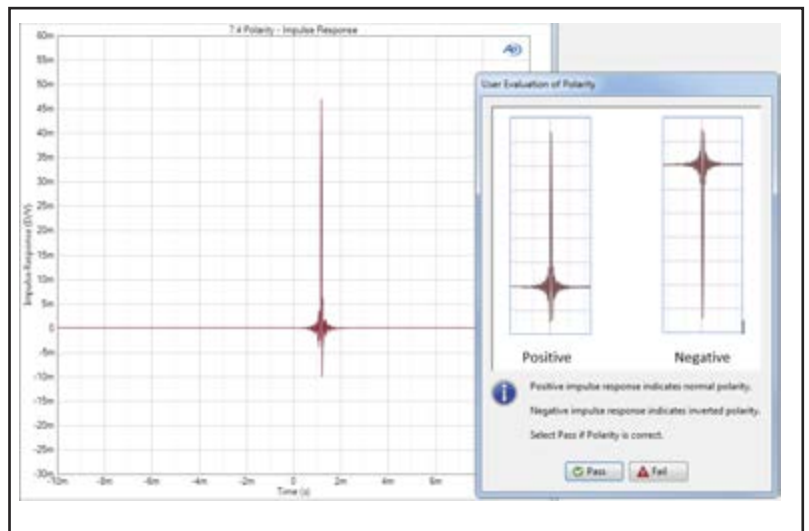


Figure 12: An impulse response (IR) is used to check polarity.

About the Author

David Mathew is Technical Publications Manager and a senior technical writer at Audio Precision in Beaverton, OR. He has worked as both a mixing engineer and as a technical engineer in the recording and filmmaking industries, and was awarded an Emmy for his sound work in 1988.

The APx555 and Benchmark's AHB2 One Perspective

By Eric Hodges

(United States)

It was July 30, 2014, and I'd been with Audio Precision for a whopping nine days. A co-worker had just forwarded me an article on Benchmark's AHB2 power amplifier, highlighting a section in the piece that described the challenges of measuring the amplifier's performance with commercially available instrumentation.

A little more than a year later, this still makes me smile. The entire Audio Precision team was focused on final preparations for the introduction of our APx555, slated for early September 2014. Here was a well-respected audio organization validating one of the key elements of the APx555's design—the need for analog test performance that surpassed that of the long-standing leader, our own 2700. The only frustrating part was the need to remain silent on the point until September. I made a mental note for a follow up and returned to the task of preparing for our product launch.

Fast forward a few months to a Thursday evening in October, and I'm in Los Angeles, CA, at the 137th International Audio Engineering Society (AES) Convention supervising the finishing touches in the booth set-up process. Imagine my surprise—seriously, the mental note I'd made in July was a bit dusty—when a trio from Benchmark and THX (a patent holder for amplifier and error-correction technologies used in the AHB2) arrived at our booth wondering if we'd like to introduce the APx555 and the AHB2. In the ensuing flurry of activity to set up a test of the AHB2 with the APx555—and effectively a test of the APx555's analog performance as indicated by its total harmonic distortion plus noise (THD+N) measurement capability—I'm not sure which engineers were more excited by the trial unfolding in booth #1212. And, as a singular group, this was a noteworthy collection of engineering talent, comprised as it was of Laurie Fincham and Jayant Datta (THX), John Siau (Benchmark), and Tom Kite (Audio Precision).

During this time, the corner of the booth occupied by the AHB2



Audio Precision was one of the exhibitors at the 139th International Audio Engineering Society (AES) Convention in New York. The APx555 analyzer was demonstrated with the Benchmark AHB2 amplifier. Here Benchmark's John Siau takes the opportunity to show the measurements to Bob Cordell.

and the APx555 radiated a palpable sense of anticipation from all involved. While I'm not sure how long the experiment set-up process actually took, focused as I was on my own show-prep duties, it seemed the test was ready in no time at all. With all eyes focused on the THD+N meter in the APx555's Bench Mode display, Tom enabled the instrument's generator and stepped back. As the generator kicked in, the bright green digits of the meter quickly stabilized at -118 dB and the resultant grins spreading across each participant's face appeared to include equal measures of pride and joy with a bit of relief thrown in as well. Just as they'd said, the AHB2 had market-leading performance. Likewise, the APx555 delivered, validating its own capabilities as it did the same for the AHB2.

For a spontaneous, completely unscripted event, it worked out quite well. Almost as though it had been engineered that way. For more information on the Benchmark AHB2 amplifier, the complete *audioXpress* review is available at <http://audioxpress.com/article/Fresh-from-the-Bench-Benchmark-AHB2-Stereo-Power-Amplifier.html>.

More information about Audio Precision's APx555 audio analyzer is available at www.ap.com/products/apx555.

About the Author

Eric Hodges is the Marketing Manager for Audio Precision in Beaverton, OR. His 18 years as a marketing professional have been exclusively in the high-tech industry, across applications as varied as test and measurement, microelectronics, broadcast, post-production, and audio.



The Benchmark AHB2 amplifier at the 137th AES convention in Los Angeles, CA.

A Single-Ended Type-A Monobloc Amplifier

Obtain 25 W With Two KT 88 Vacuum Tubes

In contrast to all other types of power amplifiers, single-ended Type-A power amplifiers offer the unbeatable advantage of not splitting the audio signal into two half-waves. In general, this provides huge sound advantages since there are no crossover distortions during the composition of the half-waves in the output stage or output transformer. The disadvantage is low efficiency. To obtain optimum results, the speakers must feature the highest degree of efficiency possible.



Photo 1: This classic design single-ended Type-A power amplifier offers the unbeatable advantage of not splitting the audio signal into two half-waves.

By
Gerhard Haas

(Germany)
Original article published in
German in Elektor's *Roehren*
Sonderheft 1, December 2005

The operation of high efficiency speaker systems is the domain of single-ended Type-A power amplifiers. For efficiencies from 100 dB/1 W/1 m and above, power amplifiers with a power output of about 10 W are sufficient. For a long time, there has been a trend toward horn loudspeakers with efficiencies of around 100 dB or slightly less. However, the demand for a clean sounding and noiseless power amplifier still exists. Also, there is a greater need for more power to get tighter basses and more dynamics. That is why I designed this single-ended Type-A power amplifier. I paid particular attention to making the full undistorted amplifier power available up to 20 Hz!

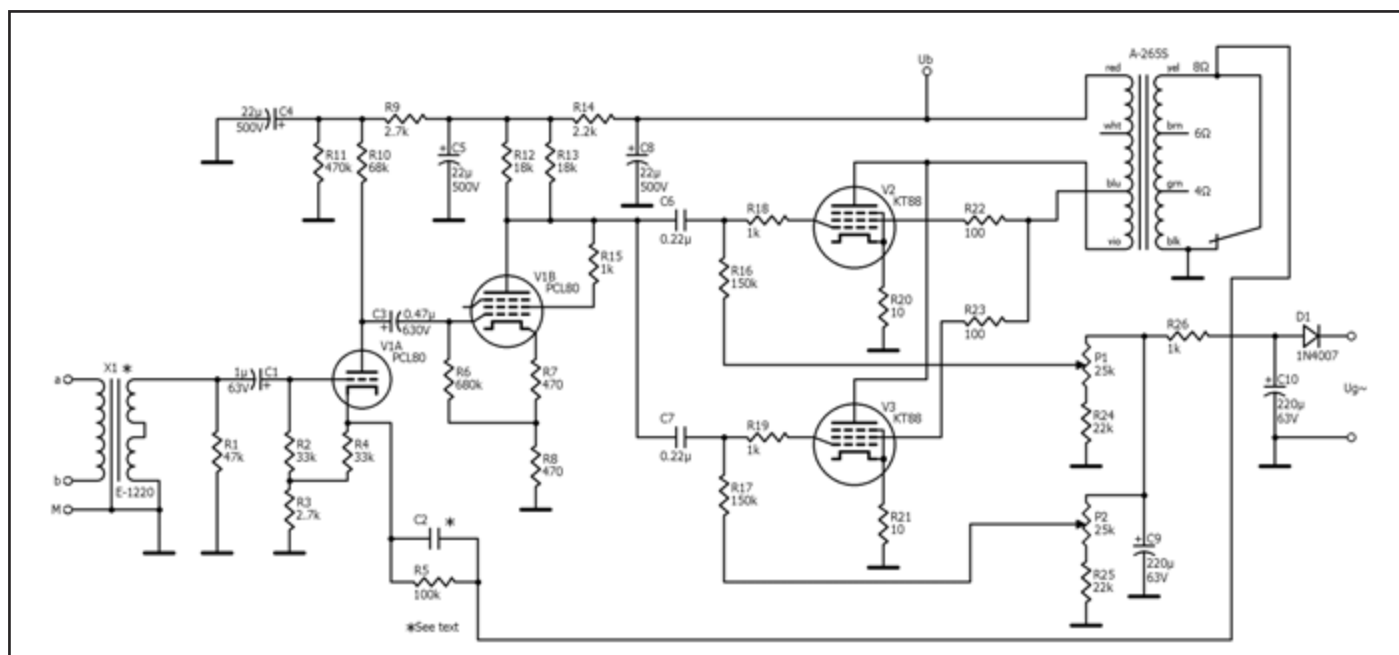
Whether an amplifier has an upper limit frequency of 50 kHz or 70 kHz is not as significant. However, it is important that there is no significant drop at 20 kHz. If an amplifier reaches 20 Hz at full power instead of 40 Hz, from a musical point of view, this is equivalent to a whole octave! Although the monobloc design is more expensive, its channel

separation permits a higher signal resolution and the best 3-D playback. With 25 W standard speakers, higher efficiency can be obtained (see **Photo 1**).

The Design

Figure 1 shows the power amplifier schematics. To obtain a higher output power, I connected two KT 88 power tubes in parallel. Alternatively, you can use the 6550 A without any changes to the circuit because both tube types feature similar specifications and are interchangeable. This means double the current is available with the same operating voltage, which results in double the output power in comparison to using a single tube.

The transformer features a screen grid tapping on the primary side (see **Photo 2**). In this case, the negative screen grid feedback dampens the odd and the unmusical distortions generated by the pentodes. In this power amplifier, I wanted to have decreasing distortion values. If the second harmonic is dominant, then it is also called the typical triode



sound, which comes from the beam-power tetrodes used in this circuit. In comparison with the triodes, it has the advantage of being able to intervene via the screen grids and to favorably shape the distortion path over the modulation. Thanks to the indirect heating of the tubes, it is possible to easily heat with direct current and completely eliminate hum from this area. This is an important factor for achieving excellent signal-to-noise values from this power amplifier.

The idle currents of the power tubes are individually set by means of potentiometers. In this way, you can get a good sense of the tube tolerances and can do without selected tubes. The PCL 86 tube (= 14 GW 8) is used for the driver stage, the triode section corresponds to half an ECC 83, which ensures a high open-loop gain. The pentode section provides additional amplification and offers a relatively high output current. With this tube type, it is possible to easily set up an operational amplifier based on tube technology: high open-loop gain, high input resistance, and high output current. The pentode's screen grid is connected to the anode via R15. As a result, this part of the circuit features a quasi-triode circuit.

To avoid hum when the monoblocs are connected with other grounded equipment, an input transformer is useful. But, I did not place this transformer on the PC board. You can also use balanced cables with XLR connectors instead of cinch connectors. The transformer type E-1220 is completely shielded and was developed for professional studio equipment. It is made for print mounting and can be soldered on a small PC test board.

The Power Supply

I chose a double-choke filter for the high-voltage power supply shown in **Figure 2**. Since an operating

voltage of 480 V is applied here, the chokes have to be substantially more robust than a power supply stabilized by semiconductors, especially during voltage ramp up. I also selected BY 299 type fast recovery diodes as rectifiers because they have less commutation difficulties than normal rectifier diodes.

The rectifier is followed by a C-L-C filter, after which another LC element is installed. The first two electrolytic capacitors are smaller, which reduces the gassing voltage peaks from the mains transformer and keeps the stray field and the current spikes low. Furthermore, the surge current loads to the rectifier diodes are lower, which benefits the operational safety.

The last electrolytic capacitor is very large. This

Figure 1: To obtain a higher output power, I connected two KT 88 power tubes in parallel. This schematic shows my power amplifier design.

Technical Data

Measured at 8 Ω load

Output Power (below clipping) 25 W at THD = 4% and 1 kHz

Frequency Response 16 Hz–20 kHz (–0.2 dB)
30 kHz (–1 dB)

Input sensitivity (full output power) . . . 1.5 V

Input resistance 33 kΩ

Noise unweighted < 200 μV (–74 dBV, 20 Hz–20 kHz)

Noise A-weighted < 70 μV (–83 dBV)

Idle current per output tube. 80 mA

Spectrum of distortion at 1 kHz and 1 dB below full output power

THD 2.8%

K2 2.5%

K3 0.6%

K4 0.25%

K5 0.15%

Spectrum of distortion at 1 kHz and 4 W

THD 1%

K2 0.89%

K3 0.06%

K4 0.024%

K5 0.022%

Resistors metal film 0.6 W load-bearing capacity, 1% tolerance, resistors 2 W metal oxide types, 2% tolerance, and resistors 4 W, metal oxide types, 5% tolerance.

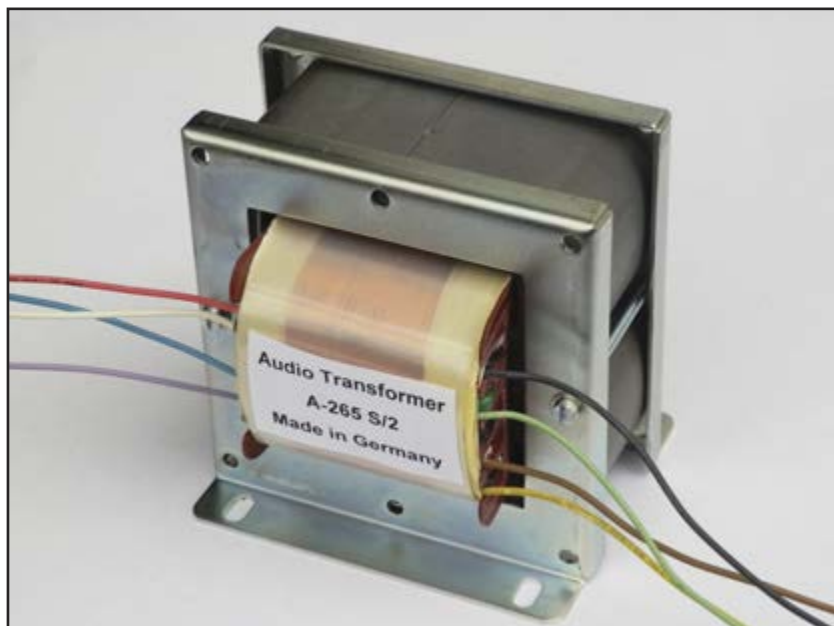


Photo 2: For this amplifier, I used a German-made A-265 S/2 audio transformer.

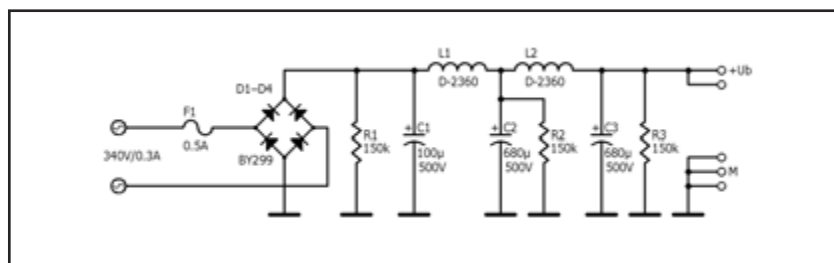


Figure 2: I chose a double-choke filter for the high-voltage power supply

ensures good filtering and the high capacitance provides abundant current with strong bass impulses. The filtering factor is so high that the remaining hum voltage is far below the power supply's physical noise voltage. In this way, the combination of DC heating, the special low-dispersion mains transformer, and good mechanical assembly results in a low interference voltage at the amplifier output.

DC Heating Circuit

Figure 3 shows the DC heating circuit. I used a proven voltage stabilizer, the 723. Even if the effort seems to be slightly greater, it is a low-drop regulator with a low voltage loss. This saves a lot of power loss and the unnecessary warm up of the amplifier.

The series regulator transistor T1 is a field-effect transistor (FET) that can be powered up so only a low voltage drop is required for a good output voltage setting. Since FETs require more than 3 to 4 V at the gate to become sufficiently conductive, IC1 gets a higher operating voltage via D1 and C6. The voltage drop makes the metal film resistors R4 and R5 limit the output current. To do so, IC1 features the pins 2 and 3 that query the voltage drop at the resistors.

To prevent high losses, the internal current limiting transistor IC1 receives a bias voltage via R8 and R7, which is stabilized by D2 and filtered by means of C3. R17 ensures that the current limitation is not prematurely triggered if IC1 heats up due to

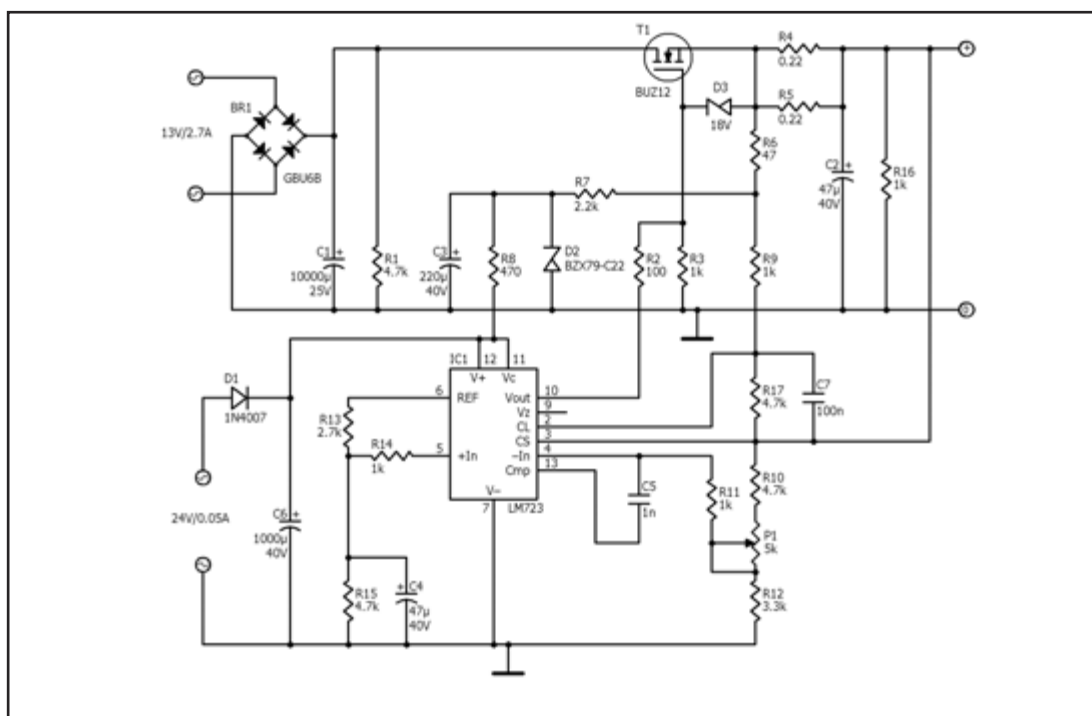


Figure 3: I used a proven voltage stabilizer, the 723, for the DC heating circuit.

an increase in temperature. The internal voltage reference of IC1 is divided via R13 and R15 and filtered by means of C4. This reduces the noise of the voltage controller. The current limitation, together with C4 and its wiring, acts as a soft start function for the tube filaments. To make the filament ramp up more slowly, the capacitance of C4 can be increased. However, you should not overdo here because during the ramp up a lot of power dissipation occurs at T1, which can destroy the transistor if it overheats.

During commissioning, first set the output voltage to the nominal value without the tubes. As soon as the filaments are applied as load, set the correct value directly at the power tube sockets. This also compensates for the voltage drop of the supply lines. It is important that the negative pole of the DC heating is connected to the amplifier ground on the high-voltage circuit board. Otherwise, the DC heating and its wiring act as an antenna and the hum strongly increases. The rectifiers GI1 and T1 need to be cooled, and T1 should be mounted so that it is electrically insulated.

The PCL 86 (= 14 GW 8) requires a heating voltage of 13.5 V. The filaments connected in series to the two KT 88s require 12.6 V. It is a good compromise if the heating voltage is set to approximately 13 V. As a rule, the heating voltage of the tubes may have a tolerance of $\pm 7\%$. If the temperature of the less stressed driver tube is slightly lower, while the more stressed power tubes are slightly overheated. This is an advantage during operation because the cathodes of the power tubes emit current more easily at a higher temperature than they do at lower temperature.

Crackle Suppression Circuit

Figure 4 shows a simple power-off crackle suppression circuit. With single-ended Type-A power amplifiers, the capacitors discharge asymmetrically, which may cause a high voltage transient at the speaker output when switching off the amplifier. This may damage sensitive horn speakers (e.g., if the voice coil carrier hits the magnetic plate or if the bead or the membrane's leads break heavily moves in the other direction).

The rectifier is directly connected to the heating coil with 13 V. The resistor R1 is installed since the voltage for the 12 V relay would become too high. Behind the rectifier, an electrolytic capacitor with low capacitance is installed, which flattens the ripple of the rectified voltage. Shortly after it turns on, the relay trips, enabling the speaker output. When switching off the amplifier, the relay immediately drops out and the speaker output is short-circuited.

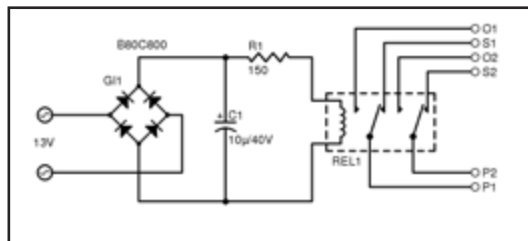


Figure 4: With single-ended Type-A power amplifiers, the capacitor discharges asymmetrically, so I used a simple power-off crackle suppression circuit.

Amplifier Parts List (Figure 1-EE-282 B)

Capacitors

C1		1 µF/63 V
C2		10–47 pF ceramic (see text)
C3		0.47 µF/630 V, MKP
C4,5		22 µF/500 V
C6,7		0.47 µF/630 V, MKP 10
C8		22 µF/500 V
C9,10		220 µF/63 V

Miscellaneous

1		socket, noval, PC-mountig, ceramic
1		socket, octal, PC-mounting, ceramic
1		input transformer, E-1220 (optional)
1		output transformer, A-265 S (Ra = 1.2 kΩ, 4, 6, 8 Ω)

Resistors

R1		47 kΩ
R2		33 kΩ
R3		2.7 kΩ
R4		1 kΩ
R5		100 kΩ
R6		680 kΩ
R7,8		470 Ω
R9		2.7 kΩ
R10		68 kΩ/2 W
R11		470 kΩ/2 W
R12,13		18 kΩ/4 W
R14		2.2 kΩ/4 W
R15		1 kΩ
R16,17		150 kΩ
R18,19		1 kΩ
R20,21		10 Ω/2 W
R22,23		100 Ω/2 W
R24,25		22 kΩ
R26		1 kΩ
P1,2		25 kΩ trimmpot

Transistors

D1		1 N 4007
Rö1		PCL 86 (14 GW 8)
Rö2,3		KT 88 or 6550 A

About the Author

Gerhard Haas has written books as well as articles for *Elektor Germany* since 1995. The *Roehren Sonderheft* magazine was his creation, along with some partners. The first issue came out in 2005. Now, there are 10 issues containing a collection of interesting articles. Gerhard has also developed many electronic circuits and equipment. In the last two decades, he primarily worked on tube amplifiers and the circuitry around them, including the audio transformers, filter chokes, power transformers. He sold his company, Experience Electronics, several years ago to focus on designing tube amps and audio transformers, write articles for *Sonderheft* and finish his book *High-End Mit Roehren*. The first copy of his book was printed in 1995, and he is now working on an updated followup.

Parts List for High Voltage Supply (Figure 2-EE-290 A)

Capacitors

C1,2	100 µF/500 V
C3	680 µF/500 V

Power transformer

NTR-13 F	(340 V/0.3 A, 50 V/0.05 A, 24 V/0.05 A, 13 V/2.7 A)
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Resistors

R1,2,3	150 kΩ/2 W
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Transistors

D1,2,3,4	BY 299
Si 1	0.5 A slow blow
Dr1,2	D-2360 (2.3 H/0.3 A) filter choke

Parts List for DC Heating (Figure 3-EE-201 B)

Capacitors

C1	10,000 µF/25 V
C2	47 µF/40 V
C3	220 µF/40 V
C4	47 µF/40 V
C5	470 pF ceramic
C6	1,000 µF/40 V
C7	100 nF ceramic

Resistors

R1	4.7 kΩ
R2	100 Ω
R3	1 kΩ
R4,5	0.22 Ω metall band 5 W
R6	47 Ω
R7	2.2 kΩ
R8	470 Ω
R9	1 kΩ
R10	4.7 kΩ
R11	1 kΩ
R12	3.3 kΩ
R13	2.7 kΩ
R14	1 kΩ
R15	4.7 kΩ
R16	1 kΩ
P1	5 kΩ trimpot

Transformers

T1	BUZ 12 or IRF1010ZBF
D1	1 N 4007
D2	BZX79C22 (zener diode 22 V, 0.5 W, low temperature drift)
D3	ZY 18 (Zener diode 18 V/1.3 W)
IC1	723 DIL
GL1	GBU6B (bridge rectifier 70 V/6 A)

Parts List for the Crackle Suppressor (Figure 4 -EE-346)

R1	150 Ω
C1	10 µF/40 V
GL1	B 80 C 800
Rel1	12 V/40 mA, DPDT, gold plated contacts

Kits and special parts are available at www.experience.electronics.de

This protects the speaker against destruction. The transformer output is short circuited by default. This helps avoid additional contacts in the signal path with all the related problems. The transformer will not be damaged.

The use of the relay features two gold-plated switch contacts. As a result, this circuit can also be used in any kind of stereo power amplifier. In monoblocs, the relay contacts are connected in parallel. However, be sure the supply lines at the relay contacts feature a cross-section of at least 0.5 mm², otherwise the efficiency is limited because of the too-high resistance wires.

Output Transformer

Finally, I will discuss the output transformer. It is manufactured using a high-quality and generously dimensioned cut-tape wound core of the size SM 102 B (= double C-core). When using it as a mains transformer, this core can handle 300 VA of power! This type of core allows an exact air gap, which is necessary because of the DC preload. C-cores are, in principle, manufactured using good thin sheet alloys. Therefore, they feature a higher magnetic resilience and lower losses.

I chose a large-dimensioned core so 20 Hz can be transmitted at full power without distorting the signal. It would have been possible to achieve the same result with a smaller core, but this would have required an increased number of turns and a thinner copper wire. This change would have also significantly increased the losses in the transformer, which result in performance losses and an impaired bass response. With this single-ended Type-A power amplifier, you have the cream of the crop in hi-fi amplifier construction with tube technology.

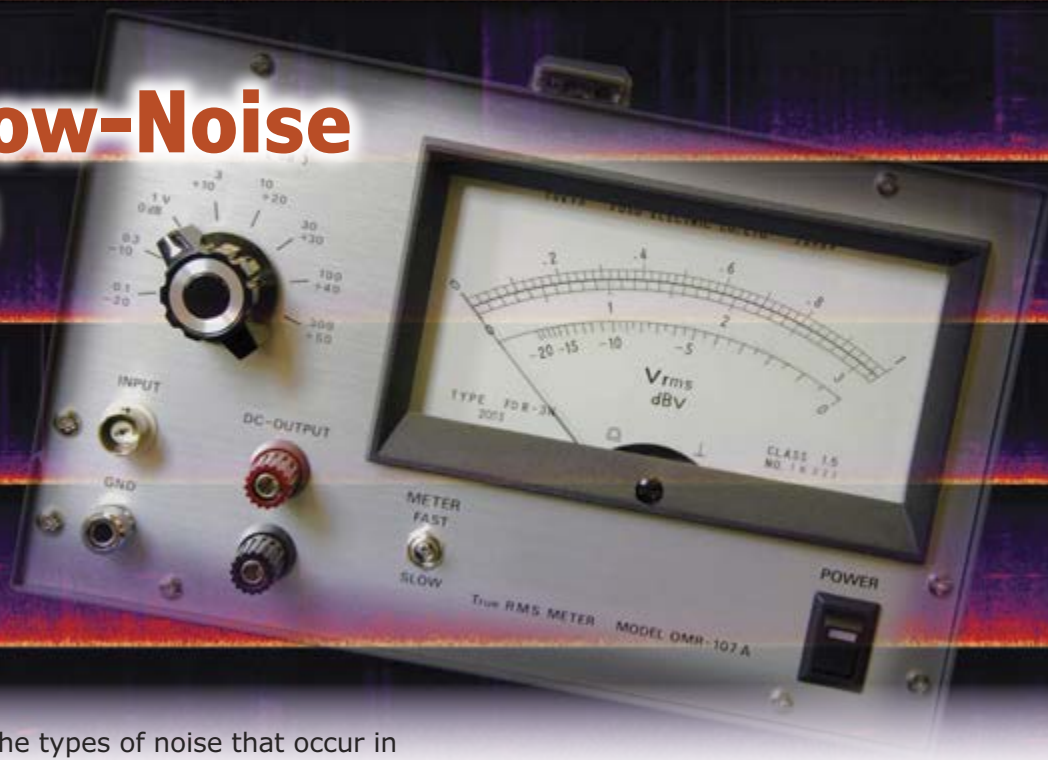
Before publishing this circuit design, I had various music enthusiasts perform several tests on a series of specimens. The results were always the same: every music lover was totally enthusiastic about the sound. As a side note, it is also important to use matching high-quality loudspeakers. Horn speakers and speakers with high efficiency of more than 90 dB/1 W/1 m are preferable. 

Author's Note: The PCB layouts are available in the Supplementary Material. Kits and special parts are available at www.experience-electronics.de.

Project Files

To download additional material and files, visit <http://audioXpress.com/page/audioXpress-Supplementary-Material.html>.

Creating Low-Noise Amplifiers



In a prior article, we discussed the types of noise that occur in hollow-state circuits. External noise enters the circuit from an external source, and internal noise is produced within the electrical system and electronic components themselves. In this article, we focus on ways to eliminate the noise.

By
Richard Honeycutt
(United States)

As we try to design for minimum internal noise, we can review the four categories of internal noise.

- For a temperature of 25°C, Johnson noise voltage is given by:

$$v_n = \sqrt{4kTR\Delta f}$$

where k = Boltzmann's constant: $1.38 \cdot 10^{-23}$ J/K,
 T = temperature in Kelvins,
 R = resistance in ohms, and
 Δf = bandwidth over which the measurement is made, in hertz.

- In general, shot noise current can be calculated from:

$$i_s = \sqrt{2qI_{DC}\Delta f}$$

where q is the charge in coulombs of a single electron, and I_{DC} is the direct current (DC) in the component. Shot noise current produced in a vacuum diode is increased or decreased by a factor F , which is less than 1 for an unsaturated diode and greater than 1 for a saturated diode.

- Johnson noise and shot noise produced by triodes and pentodes can be calculated together and presented as an equivalent noise resistance R_{eq} , which equals $2.5/g_m$ for a triode, or:

$$R_{eq} = \left(\frac{I_p}{I_p + I_{screen}} \right) \left(\frac{2.5}{g_m} + \frac{20I_{screen}}{g_m^2} \right)$$

for a pentode. Then the equivalent input noise from the tube and its grid resistor is:

$$v_{ns} = \sqrt{4kT\Delta f(R_g + R_{eq})}$$

- Many devices, especially carbon-composition resistors, generate $1/f$ noise, with a level inversely proportional to frequency. Its level is given by:

$$i_f = \sqrt{\frac{K_f I_{CD}}{K_2 f^\alpha}}$$

where the constants K_f , K_2 , and α depend upon the component.

Notice that Johnson noise depends on resistance, temperature, and bandwidth. Shot noise depends on direct current and bandwidth, and $1/f$ noise depends

on direct current and inversely upon frequency. Vacuum-tube-generated noise depends inversely upon transconductance, and in a pentode, upon screen-grid current. These facts give us some handles by which we can optimize a design for lowest noise. Unfortunately, the component-dependent factors controlling $1/f$ noise are seldom available to us, so although we can take steps to reduce this form of noise, we cannot predict its amplitude in advance.

Noise Levels in a Triode Amplifier

Figure 1 illustrates the places in a triode amplifier where we can calculate the noise level.

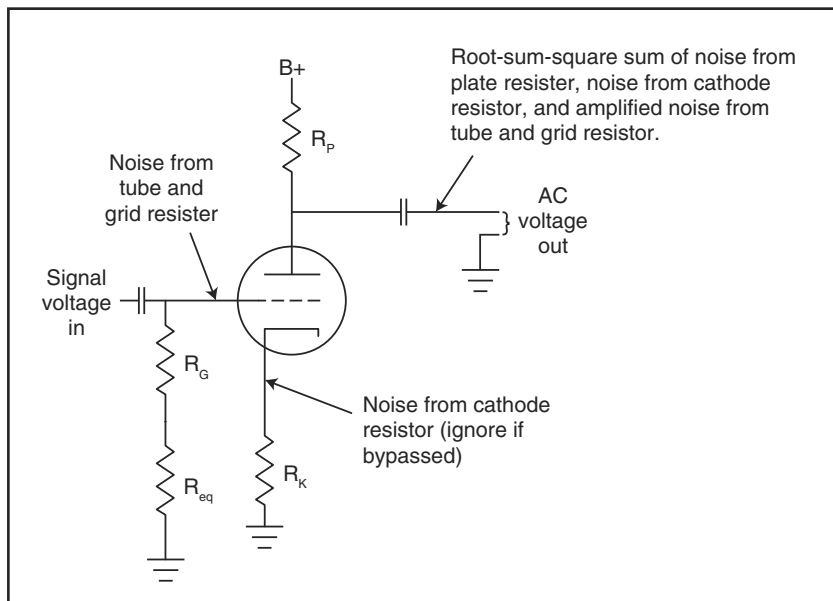


Figure 1: The internal noise in a vacuum-tube stage can be examined at several places.

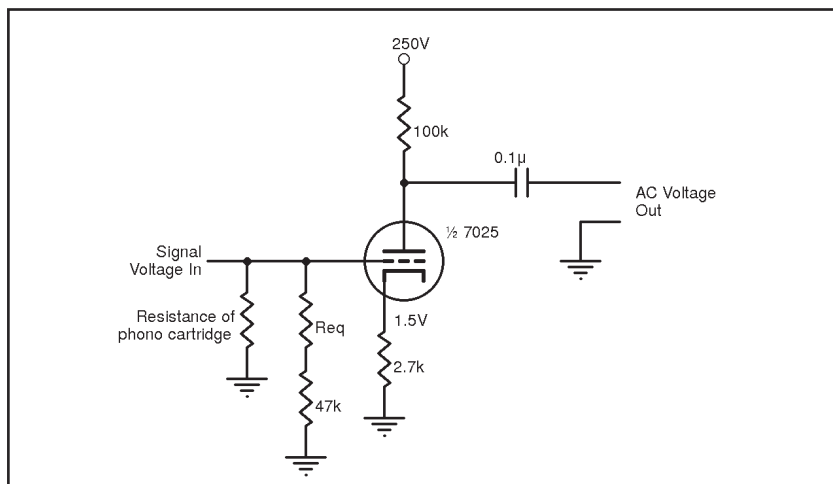


Figure 2: This is the first stage of a phono preamp.

Figure 2 shows the first stage of an RCA-designed phono preamp. This particular type of circuit was chosen as an illustration because it is commonly of interest to hollow-state audio aficionados. However, it has a peculiarity that is not common to all types of preamplifiers. The source resistance (the resistance of phono cartridge) is typically about $10\ \Omega$, although the impedance is higher due to the inductance of the windings. Therefore, when we calculate the equivalent input noise voltage, we will get a very small value.

Equivalent Input Noise Voltage

We calculate equivalent input noise voltage in seven steps:

1. To find the triode's g_m , we'll need the plate current and the voltage. We can find I_p by dividing the $2.7\text{-k}\Omega$ cathode resistor into the 1.5-V cathode voltage, which gives us a plate current of about 0.556 mA . The plate voltage equals the $B+$ voltage minus the $(0.556\text{ mA} \times 100\text{ k}\Omega = 55.6\text{ V})$ drop across the plate resistor: V_p is thus about 195 V .

2. From the General Electric (GE) datasheet at <http://frank.pocnet.net/sheets/135/7/7025.pdf>, we find a graph showing the g_m of a 7025 triode to be about $1020\ \mu\text{S}$. By the way, the same datasheet shows r_p to be $90\text{ k}\Omega$. We'll need to know that in step 5.

3. The equivalent noise resistance of the triode is given by:

$$R_{eq} = \frac{2.5}{g_m} = \frac{2.5}{1020 \times 10^{-6}} = 2450\ \Omega$$

4. The noise contributed by the tube and grid resistor shunted by the resistance of the phono cartridge is calculated from:

$$\begin{aligned} v_{ns} &= \sqrt{4kT\Delta f(R_G + R_{eq})} \\ &= \sqrt{4 \times 1.38 \times 10^{-23} \times 298 \times 20000 \times (10 + 2450)} \\ &\approx 0.9\ \mu\text{V}_{\text{RMS}} \end{aligned}$$

We chose 20 kHz as the bandwidth of the amplifier.

5. This equivalent input noise voltage is amplified by the triode stage's gain:

$$A_v = \frac{\mu r_L}{(r_p + r_L)} = \frac{100 \cdot 87000}{90000 + 87000} = 49.15$$

We assumed that the next stage has an input resistance of $680\text{ k}\Omega$, making the AC load resistance $r_L = 87\text{ k}\Omega$, a fairly normal value for such a circuit. The output noise voltage becomes:

$$0.9\ \mu\text{VRMS} \times 49.15 = 44\ \mu\text{VRMS}$$

6. The noise generated in the cathode resistor can be ignored since the resistor is capacitor-bypassed.

7. The noise generated in the plate resistor is:

$$v_{ns} = \sqrt{4kT\Delta f R_p} = \sqrt{4 \cdot 1.38 \times 10^{-23} \cdot 298 \cdot 20000 \cdot 100000} \cong 5.7 \mu V_{RMS}$$

In spite of the unusually low equivalent input noise due to the very small source resistance, the noise at the stage's output is predominately due to the amplified equivalent input noise. The contribution from the plate resistor is insignificant, and even if the cathode resistor were not bypassed, its noise contribution would be even less, since it is only 1/37th the resistance of the plate resistor. It is true in almost all cases that a good approximation of the total noise in the output of a vacuum tube stage can be obtained by simply multiplying the equivalent input noise by the stage gain, and ignoring the contribution due to Johnson noise in the other resistors.

We did not look at the shot noise that we might have if we used carbon-composition resistors, although this contribution would also be small because of the tiny source resistance. Shot noise from carbon-comp resistors could be important in a preamp whose input was not shunted to ground by a very low source resistance; for instance, if the source were some sort of piezoelectric device, which would have a near-infinite source resistance so that RG is not shunted by another resistance.

Preamp's Second Stage

Now how about the second stage of the preamp? Let's assume that its circuit is similar to that of the first stage, except for ($R_G + R_{eq}$) being shunted by the first stage's output resistance of 87 k Ω (the 90-k Ω r_p in parallel with the 100-k Ω plate resistor). The value of 87 k Ω in parallel with, say, a 680-k Ω grid resistor is about 77 k Ω . The equivalent input noise of the second stage is then:

$$v_{ns} = \sqrt{4kT\Delta f (R_G + R_{eq})} = \sqrt{4 \cdot 1.38 \times 10^{-23} \cdot 298 \cdot 20000 \cdot (77000 + 2450)} \cong 5.2 \mu V_{RMS}$$

Adding this voltage in a root-sum-square fashion to the output noise voltage from stage 1 gives an equivalent input noise voltage of 44.3 μV_{RMS} .

Notice that the noise performance of the second stage is completely unimportant. This leads us to a general rule: in a multistage amplifier, the

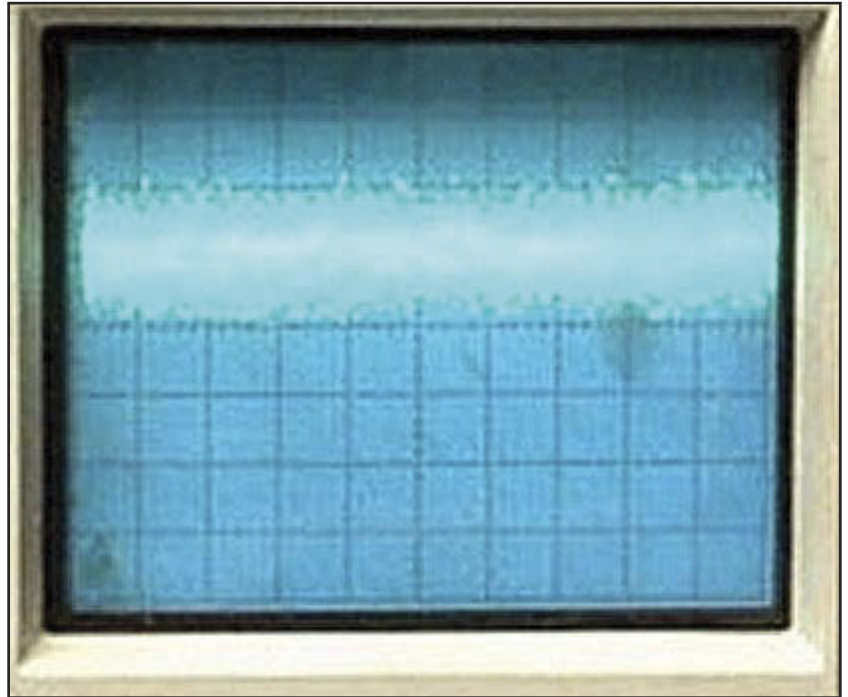


Photo 1: This is the noise signal as viewed on channel 1.

noise performance of the first stage pretty much determines the noise performance of the whole amplifier. Also notice that only with sources having very low resistance does the noise performance of the tube matter much.

In the second stage we've just examined, the resistance of the grid resistor in parallel with the output resistance of the first stage was 77 k Ω , 31 times the value of R_{eq} for the triode. Since the noise voltage is proportional to the square root of the resistance, this means that the triode's noise contribution is about 15 dB below the total noise.

Searching for Noise Sources

Now let's neaten up a couple of details. We have ignored 1/f noise, because it is difficult to predict in a hollow-state circuit, since it is mainly generated by carbon-composition resistors whose noise performance is usually unknown. In last month's article, "Some Background Information on Noise in Amplifiers," we showed that below a certain corner frequency, 1/f noise is dominant, and its amplitude can become very large at very low frequencies.

However, in an audio circuit, performance is ultimately judged by human ears, whose sensitivity is about 35 dB lower at 20 Hz than at 1 kHz, for sound played at the level of normal conversation. (At higher listening levels, the sensitivity discrepancy is

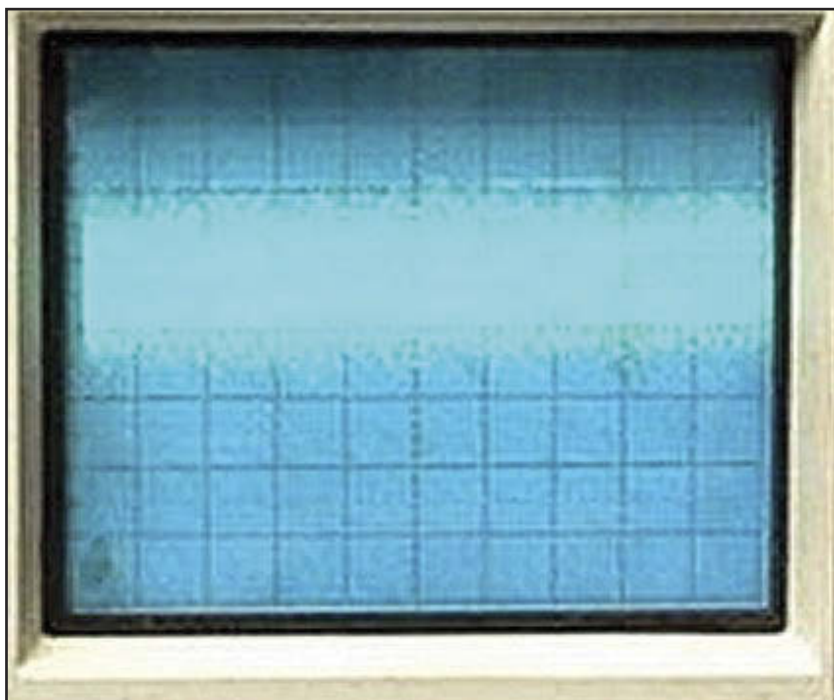


Photo 2: With the noise signal fed to both channel 1 and channel 2, and vertical positions set so the dark band disappears from between them, the display looks like this.

less.) So the increased low-frequency noise is not as noticeable as it might at first seem.

Aside from carbon-composition grid resistors in the first stage of preamps being driven by medium-to-high-resistance sources, three other components can cause serious $1/f$ noise: potentiometers (especially carbon-composition ones), switch and relay contacts having some corrosion, and input connectors that are somewhat corroded.

These noise sources are particularly egregious if there is DC in those components or contacts. Thus, it is important to avoid placing such components in the input path of a high-gain preamp, if at all possible.

A noise metric that is more informative than total noise voltage is signal-to-noise ratio (SNR): $SNR = 20 \cdot \log_{10}(V_{\text{signal, RMS}}/V_{\text{noiser, RMS}})$. The actual SNR specification uses the values of signal voltage and noise voltage at the device's output. However, knowing that the equivalent input noise voltage at the input of the first stage pretty much determines the final noise performance, we can approximate the SNR of the preamp we've been analyzing by calculating the SNR at the (equivalent) input.

Let's assume that we're using a moving-magnet cartridge with a fairly typical output of 5 mV for the standard groove velocity. Then the SNR at the input is: $SNR_{\text{input}} = 20 \cdot \log_{10}(5 \text{ mV}/0.9 \text{ } \mu\text{V}) = 75 \text{ dB}$. Actually, if we had not approximated at a few places,

our calculated SNR at the output would have been a little lower.

Hum and Buzz

The other fly in our SNR ointment is noise/hum/buzz from the power supply. If we use large enough filter capacitors, paralleled with 0.1- μF film capacitors for better rejection of high hum/buzz harmonics, we can control hum quite well. It is also important to twist filament leads tightly together and route them close to the metal chassis and as far from low-level signal leads as possible. Each ground should be run separately to a single point on the chassis near the lowest-level input. Power transformers should be encased in steel. Tubes used in the first stage should be shielded.

Noise Measurement

Accurately measuring electrical noise requires a bit of know-how. Noise measurements should always be made in RMS terms. Most AC voltmeters, including the digital ones, use detectors (the part that changes the AC into the DC to be measured) that sense the average value of the voltage wave. For a sine wave, the average value is 0.636 times the peak value; the RMS value is 0.707 times the peak value. For other waveforms, these conversion factors do not apply. But even for a sine wave, a meter with an average-responding detector would read about 1 dB low, except for a correction factor that is used in calibrating voltmeters. However, this correction factor only applies to sine waves, and random noise does not produce a sine wave.

The correct way to measure noise is to use a true-RMS meter. Unfortunately not everyone has one. In years past, true-RMS meters were expensive to produce, and so Western Electric invented a quasi-RMS detector for telephone-company use in measuring noise. The Western Electric model 3A NMS (noise measuring set) is such a device.

With modern microprocessor-based voltmeters, true RMS capability is more affordable than it used to be. However, there is another technique using a dual-trace oscilloscope that produces pretty accurate measurements. First, feed the channel 1 probe (set on $\times 1$, not $\times 10$) with the noisy signal, and adjust the channel 1 vertical volts/division switch so that the noise takes up about a quarter of the screen height. Leave the continuous sensitivity adjustment locked in the calibrated position. Use the vertical position control to move

the trace into the upper half of the screen. The sweep speed (time/division) should be adjusted so that the noise appears as sort of a ribbon across the screen (see **Photo 1**).

Set the channel 2 controls to match the channel 1 controls and connect the channel 2 probe to the point where the channel 1 probe is connected. Use the channel 2 vertical position control to adjust the position of the lower “noise ribbon” so that the dark band between the two ribbons just disappears (see **Photo 2**).

Now switch the probes’ switches to “ground.” If they don’t have switches, disconnect the probes from the circuit under test and clip the two probes together to the oscilloscope ground. You will see two horizontal lines on the screen. Using the volts/division setting, calculate how many volts are represented by the separation between these lines. In **Photo 3**, the lines are one vertical division apart. If the volts/division switches are set on 10 mV, this separation represents 10 mV, which is the RMS voltage of the noise signal. 

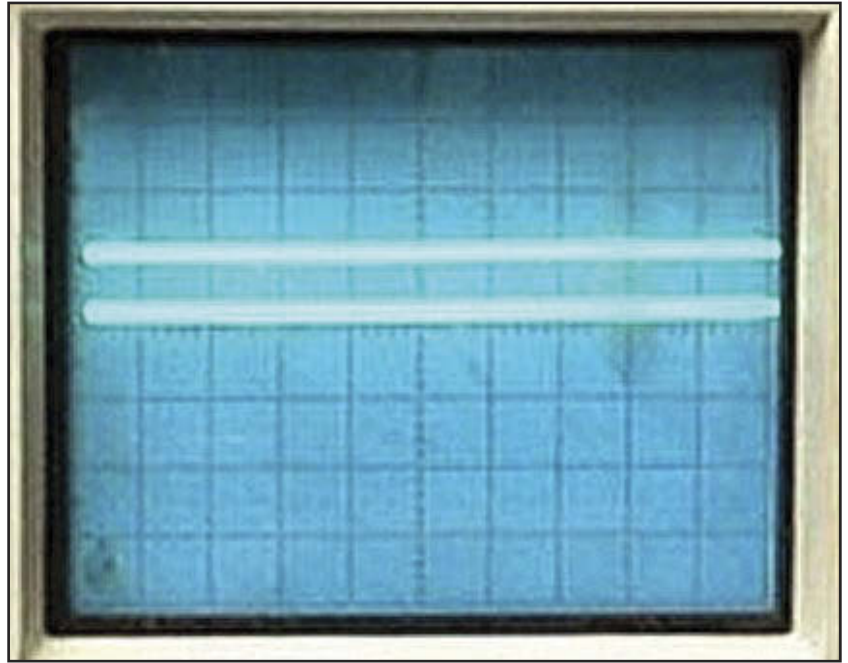


Photo 3: After the channel inputs are grounded, these two lines remain.



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Industry Calendar

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January 3–4	ALMA International Symposium & Expo 2016	www.almainternational.org
January 6–9	Consumer Electronics Show (CES) 2016	www.cesweb.org
January 21–24	The National Association of Music Merchants (NAMM) Show	www.namm.org
February 2016		
February 3–5	AES 60 th Conference on Dereverberation and Reverberation of Audio, Music, and Speech	www.aes.org/conferences/60
February 9–12	Integrated Systems Europe (ISE)	www.iseurope.org
February 10–12	AES 61 st Conference on Audio for Games	www.aes.org/conferences/61
February 13	World Radio Day	www.timeanddate.com/holidays
February 22–25	2016 GSMA Mobile World Congress	www.mobileworldcongress.com
February 29–March 3	Prolight + Sound Guangzhou 2016	www.prolight-sound.com
March 2016		
March 15–17	electronica China	http://electronica-china.com
March 18–20	Salon Son & Image	www.chestergroup.org/salonimage/2016
March 19–17	CanJam SoCal 2016	www.head-fi.org
April 2016		
April 5–8	Prolight+Sound Frankfurt	www.prolight-sound.com
April 7–10	Musikmesse	www.messefrankfurt.com
April 15–17	AXPONA (Audio Expo North America)	www.axpona.com
April 18–21	National Association of Broadcasters (NAB) Show 2016	www.nabshow.com
April 29–May 1	Lone Star Audio Fest	www.lonestaraudiofest.com
May 2016		
May 5–8	HIGH END 2016	www.highendsociety.com
May 9–11	PALME Middle East	www.palme-middleeast.com
May 16–19	Audio Engineering Society (AES) Brasil Expo 2016 & 14 th Brasil Conference	www.aes.org
May 26–28	PALM Expo—India	www.palmexpo.in
June 2016		
June 3–5	The Home Entertainment (T.H.E.) Show	www.theshownewport.com
June 4–10	InfoComm 2016	www.infocommshow.org
June 4–7	Audio Engineering Society (AES) 140 th International Convention	www.aes.org/events/140
June 13–15	European Acoustics Association (EAA) EUROREGIO 2016	https://www.euracoustics.org
June 23–25	Summer National Association of Music Merchants (NAMM) Show	www.namm.org/summer/2016
July 2016		
July 18–20	2016 Audio Engineering Society (AES) International Conference on Sound Field Control	www.aes.org/conferences/2016/sfc
August 2016		
August 19–22	Palm Expo 2016—China	www.palmexpo.net
August 24–26	2016 AES International Conference on Headphone Technology	www.aes.org/conferences/2016/headphones
September 2016		
September 2–7	IFA Berlin	www.ifa-berlin.com
September 8–13	IBC 2016	www.ibc.org
September 14–17	CEDIA Expo 2016	http://expo.cedia.net
September 15–17	Prolight+Sound NAMM Russia 2016	www.prolight-sound.com
September 18–20	PLASA Show London	www.plasashow.com
September 29–October 1	141 st Audio Engineering Society (AES) Convention	www.aes.org/events/141
October 2016		
October 7–9	Rocky Mountain Audio Fest (RMAF)	www.audiofest.net
October 13–16	HKTDC Hong Kong Electronics Fair (Autumn Edition)/Global Sources Electronics Show	www.hktdc.com/hkelectronicfairae
October 13–16	electronicAsia	www.hktdc.com/hkelectronicfairae
October 26–29	Prolight+Sound Shanghai 2016	www.prolightsound-shanghai.com
November 2016		
November 8–11	electronica 2016/ 27 th International Trade Fair for Electronic Components, Systems, and Applications	http://electronica.de

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