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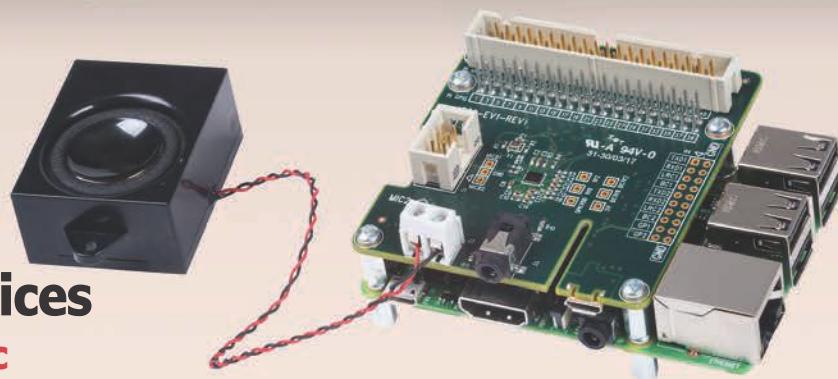
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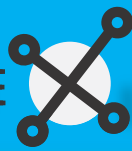
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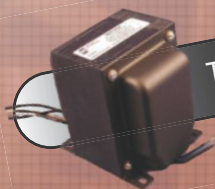


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The Year of Disruption

I recently supported a new Kickstarter project from an European company, EOZ Audio (www.eozaudio.com), which promises to deliver new Bluetooth 5.0 true wireless earbuds with high-resolution audio, 52 hours playtime (four on the buds plus 48 hours on the charging case), two 8 mm large custom-made dynamic drivers, a beautiful design with secure fit, and instant pairing. Plus, it already uses a USB Type-C cable for charging the case—truly cutting edge!

The EOZ Air, dubbed The World's Most Advanced True Wireless Earphones, looked like an ideal and irresistible proposition for just \$99, even if knowing that previous campaigns on crowdfunding platforms didn't exactly deliver on claimed promises or made me wait for more than a year until I received the promised product.

Still, it has been an entertaining and educational experience to follow the reports and updates from entrepreneurs like these, dreaming to build their ideal cutting-edge devices a little too early, and in most cases, without a single established manufacturing partnership in place.

To be fair, in the case of EOZ Audio, this is not its first crowdfunded product. In fact, the EOZ Air is simply an evolution of the company's EOZ One Bluetooth earbuds it is currently selling, following a successful 2016 Kickstarter campaign (shipping less than five months after being funded.) That made me curious. Would a Kickstarter campaign actually deliver in 2018 what all the established brands are not even daring to announce yet? The EOZ Air truly wireless earbuds are promised to be shipping February 2018. The campaign was supported by 9,099 backers that pledged a whopping \$1,050,200!

This is the industry we are in. This is called "disruption."

I'm writing this with still two months to go before CES opens its doors, and to be honest I'm not expecting a lot of Bluetooth 5 products at the show, even if this should be the year for companies to confirm their transition to the new standard, as well as USB 3.2 and USB Type-C. I expect CES 2018 to be a lot about wireless charging and voice. Hearing aids and personalized true wireless earbuds will converge into hearing enhancers. Real-time acoustic room correction promises to create perfect sound reproduction for new entertainment experiences, while we dream of virtual acoustics. And I wish we could quickly start to see high-resolution audio streaming available on major streaming platforms, MQA or otherwise.

Still, I believe the larger promise of improved wireless audio and personal networks will take until 2019 to be fulfilled, judging from the way the leading brands play those technology cards.

That's why the role of companies exhibiting at the Eureka Park (Sands Expo, Level 1) is so important. Among those companies, there will be future Apples and Teslas, the disrupters who anticipate consumers' needs and get to market.

Particularly exciting for 2018 will be the role of connected devices, and the smart home. Traditional audio companies need to take notice. The future of high-end, home theater, and luxury audio will be connected and integrated within the smart home. Not because consumers will be requiring your high-end DAC or amplifier to be Alexa-connected, but because its use will only make sense integrated within the new ecosystem of streamed content and wireless standards, offering seamless control options. Speaker companies in particular should take notice of what's happening with "smart speakers" and home wireless speaker designs. Those integrated active systems and highly optimized DSP-capable designs are sounding surprisingly good. Apple, Sonos, Bang & Olufsen, RIVA, Devialet, and others are showing the way, but that's just the start.

As it pertains to audio technology, 2018 will be an exciting year.

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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.

Mike Klasco is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is the organizer of the Loudspeaker University seminars for speaker engineers. Mike specializes in materials and fabrication techniques to enhance speaker performance.

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.

Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at White Sands Missile Range. In 1957, he started Testronic Development Laboratory, which became TDL Technology, to develop audio electronics. All product sales and services were terminated on December 31, 2015, but the TDL website is still online with a variety of audio information and downloads.

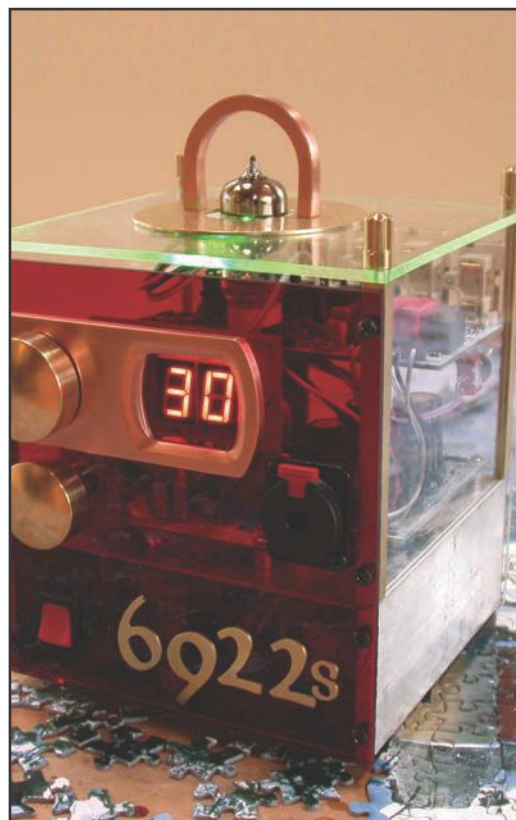
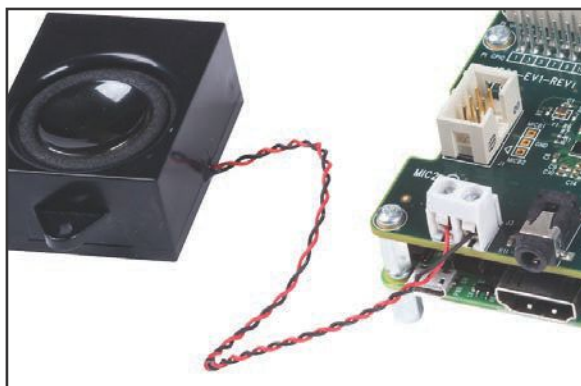
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Music Streaming Services with Volume Normalization

It seems that streaming has become the most popular way to listen to music but the services have been slow to adopt volume (loudness) normalization. Online polls of streaming service users indicate that “having to constantly adjust the playback volume control” is the most frequent complaint. Among the music services with reasonably good fidelity that use normalization, I have once again examined Spotify Premium and TIDAL HiFi.

By
Ron Tipton
(United States)

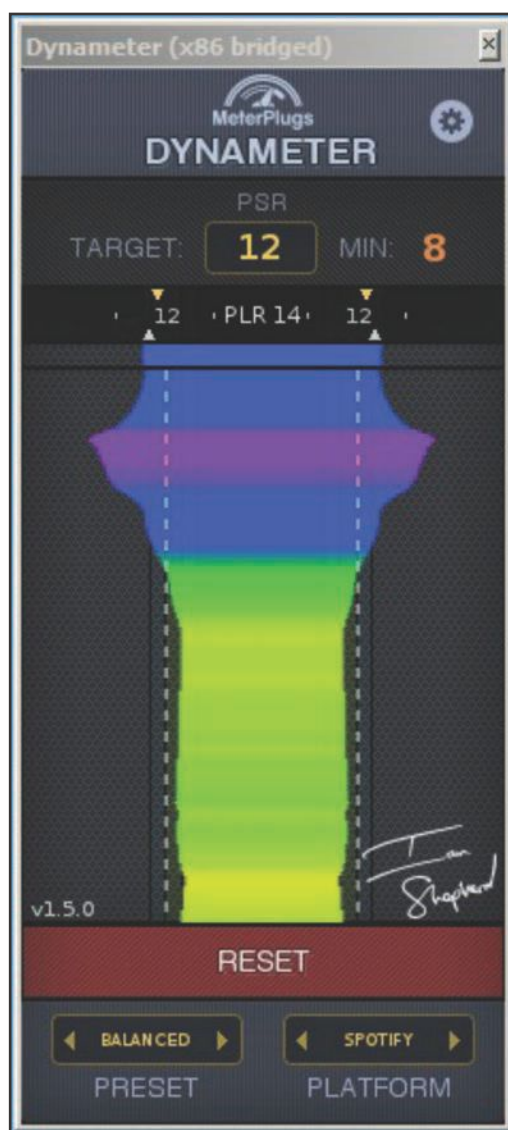


Photo 1: The Meterplugs Dynameter measures the peak to long-term (integrated) loudness ratio (PLR) and the peak to short-term loudness ratio (PSR). The multi-colored display scrolls down the screen as the file is analyzed. There are several Presets and the Platform choices include Spotify, Tidal, AES, and Broadcast. A copy of the User Manual is included in the Supplementary Material.

Spotify Premium streams Ogg Vorbis files at up to 320 kb/S. This is a lossy compression format and the rate depends on the music content. If you are not familiar with Ogg Vorbis, we published information about it in “Streaming Music Experiences (Part 4),” *audioXpress*, April 2017. TIDAL HiFi streams 16 bit, 44.1 kHz FLAC files at 1,411 kb/S, which is CD quality. But before I jump into the specifics of these two music services, I need to present some background.

Album streaming, in the sense of a CD or vinyl original, is usually not a problem. The mastering is done to set the volume at pretty much the same level for each track. Playlists, with tracks taken from several (or many) singles or albums, cause the problem. And it can get worse if the playlist includes classical tracks because of their usually higher dynamic range. In a quiet listening room, all may be well, but with a mobile device or in a moving vehicle the ambient noise level may swamp the music. This poses a problem for the streaming services, which try to live up to “listen everywhere.”

Dynamic compression should not be part of the answer. A way must be found to apply the normalization so the perceived volume stays nearly constant. I am emphasizing “perceived” because this may not agree with our usual notions of calculating and measuring the normalization factors. In other words, a bit of “magic” is needed.

In cooperation with TIDAL, the University of the Arts Utrecht in The Netherlands (HKU) prepared a document: “Recommendation for loudness normalization by Music Streaming Services” (an Adobe pdf copy is included in the Supplementary Material available on audioxpress.com).

Quoting from page 2 of the document: “A proposal was developed to use album normalization where the

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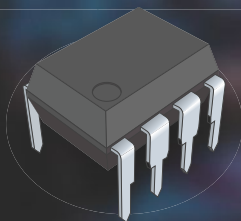
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loudest track of each album is normalized to -14 LUFS and the other tracks are aligned to the relative level they have in the album. These levels will also be used when tracks are played in a randomly shuffled playlist with other albums' tracks. This proposal was tested against track normalization in a shuffled playlist of 24 songs with 38 subjects. It turned out that 80% of the subjects

preferred album normalization, even though the tracks were selected for an extreme difference in loudness, of up to 10 LU." (1 LU = 1 dB) Based on this test, the authors of the article wrote a list of recommendations.

As you will see in the following Spotify and TIDAL sections, Spotify is not doing it this way but perhaps TIDAL is. My example TIDAL playlist has a track with an integrated loudness 6.6 dB lower than the nominal normalization level. It is acceptable in this case because of the nature of the music, but I think a 10 dB difference would require a volume control adjustment. I understand you can't please all the people all the time but volume normalization is far from settled—there are eight recommendations in the HKU document.

Spotify

In trying to discover how Spotify performs normalization, I found a very informative article by Mikko Lohenoja (an Adobe pdf copy is included in the Supplementary Material). Lohenoja writes that Spotify uses the ReplayGain parameter in the file's metadata to set the playback level. The big question is how this value is calculated. After making lots of measurements and trying to correlate one thing with another, he contacted Ian Shepherd, the creator of the Dynameter Loudness Meter (see **Photo 1**), a rather different kind of loudness meter. It measures the peak to short-term loudness ratio (PSR) and the peak to long-term (integrated) loudness ratio (PLR).

Lohenoja continued to experiment. Music that sounded fine in his studio sounded, in his words "wimpy" or "rubbish quiet," when uploaded and played back through Spotify. He continued by writing: "I heard stuff coming out of Spotify that defied all known laws of physics." He continued to experiment and finally came up with his empirical "Magic Formula:"

$$\text{Playback LUFS} = \text{Minimum PSR} - \text{PLR} - 8$$

In the majority of cases, this equation works in Spotify, although there is no theory that explains why. To test it, I found a pleasant Spotify playlist named "Peaceful Guitar" and captured the first 14 tracks in wav and flac formats, using Streaming Audio Recorder from www.apowersoft.com. **Table 1** shows my analysis using the Toneboosters Loudness Meter (see **Photo 2**) and the Meterplugs Dynameter. Neither the Integrated nor Short-Term numbers show much correlation so I used the Magic Formula on the Dynameter PLR and PSR columns and found good correlation at nominally -14 LUFS. In addition, the music sounded normalized—I had no reason to "fiddle" with the volume control.

I then tried the Magic Formula on my playlist from my "Loudness Revisited (Part 3)" article (*audioXpress*, August 2017). This analysis is summarized in **Table 2**.

Toneboosters—LUFS EBU R128 (2014)				Dynameter—Spotify		
	Integrated	Short Term	PLR	PLR	PSR	Playback
1	-15.7	-13	14	14	8	-14 LUFS
2	-17.2	-13.5	15	15	9	-14
3	17.7	-13.1	17.5	17	10	-15
4	-15.9	-12.5	15.9	16	10	-14
5	-17.3	-11.4	15.6	15	8	-15
6	-17.1	-13.2	16.7	16	10	-14
7	-16.7	-14	16.3	16	9	-15
8	-15	-13.3	14.2	14	8	-14
9	-16.4	-13.1	16.2	16	9	-15
10	-16.2	-13.8	14.7	14	10	-12
11	-14.4	-10	13.1	13	8	-13
12	-16.3	-14.1	15	15	10	-13
13	-16.7	-12.9	15.4	15	10	-13
14	-16.5	-10.5	16.4	16	9	-15

Table 1: This is my analysis of the first 14 tracks of the Spotify "Peaceful Guitar" playlist. There is little correlation between the Integrated values or between the Short-term values. But applying the "Magic Formula" to PLR and PSR shows good agreement at a nominal playback of -14 LUFS.



Photo 2: The Toneboosters Loudness Meter continues to display all the final values when the file analysis ends making it easy to write the numbers down. The Meter modes include LKFS ITUR BS.1770/3, but it has not yet been updated to include BS.1770/4. In all my measurements, Meter modes LUFS EBU R128 (2014) and LKFS ITUR BS.1770/3 gave the same results.

Normalization to -16 LUFS would be possible if the True Peaks for the three -17 LUFS values were less than 1 dB because a gain of 1 dB would be needed to bring them up to -16 LUFS. I checked with the Toneboosters Meter and found the three True Peaks to all be equal to or less than -1.5 dB.

The correction values to normalize to -16 LUFS in Table 2 are certainly different from the correction numbers in my “Loudness Revisited (Part 3)” article (see **Table 3**). But I made the corrections using the AVS Audio Editor and set up a foobar2000 playlist. I was somewhat surprised to hear properly normalized music, Spotify is apparently using track normalization and I think it sounds fine.

TIDAL

According to an article, “TIDAL implements loudness normalization – but there’s a catch,” by Ian Shepherd, the catch is quiet tracks will not be amplified up to the normalization level. Also, according to this article, “...on mobile devices and browsers, all music will be played back at an integrated loudness of -14 LUFS.” This appears to conform to the HKU recommendations.

I have a HiFi account and TIDAL recommends on its Streaming page (see **Photo 3**), “The best HiFi/Master audio quality is only available on the TIDAL desktop application—Download here.” I’m using that program for my listening and apparently normalization is always enabled in HiFi accounts.

I began by capturing the first 14 tracks of the TIDAL playlist “Classical Relaxation.” This is the playlist I used in my article “Streaming Music Experiences (Part 3)” (*audioXpress*, March 2017). This music was not normalized when I visited it for that article—had it now been normalized? **Photo 4**, a screenshot of the first four tracks, says it all—no normalization. The other 10 tracks are just as varied or worse.

I thought, maybe TIDAL doesn’t normalize old playlists so I went to the New Playlist section and captured the first 14 tracks of “Legends: Beach Boys.” These tracks are mostly from different albums and singles and include some mono and re-mastered. Although some tracks were a bit quiet, I had no reason to adjust the volume—a very enjoyable collection.

I analyzed the tracks with the Toneboosters Loudness Meter, with my results shown in Table 3. The normalized level from the Integrated column is nominally -17 LUFS even though Track 12 is slightly louder at -16.4 LUFS. Track 4 sounded fine at -23.6 LUFS because its “mood” fits that playback level. I tried analyzing with the Dynameter plug-in but there was no correlation between the Magic Formula values and the Integrated values in Table 3.

I suppose TIDAL is also using ReplayGain simply because it’s more practical than having to normalize

each track in its gigantic library. If, for some reason, the normalization value needed to be changed, it would be easy to change ReplayGain because it could be mostly automated through batch processing.

Both Spotify Premium and TIDAL HiFi are available on Sonos. This could be a good option if you have

Source	Normalizaton to -16 LUFS using Max “S”	Dynameter Magic Formula Min PSR - PLR - 8	Normalization to -16 LUFS
78 RPM record	-4.1 dB	6 - 14 - 8 = -16 LUFS	0 dB
Reel-to-reel tape	-5.6 dB	6 - 12 - 8 = -14	-2 dB
Audio cassette	-5 dB	8 - 13 - 8 = -13	-3 dB
33-1/3 RPM record	-5.8 dB	7 - 13 - 8 = -14	-2 dB
Purchased CD	-5.4 dB	9 - 14 - 8 = -13	-3 dB
VHS movie sound track	-3.6 dB	5 - 10 - 8 = -13	-3 dB
DVD sound track	-4.1 dB	8 - 17 - 8 = -17	+1 dB
iTunes movie sound track	-5.3 dB	6 - 13 - 8 = -15	-1 dB
Deezer capture	-6.2 dB	5 - 14 - 8 = -17	+1 dB
TIDAL capture	-1.6 dB	5 - 14 - 8 = -17	+1 dB

Table 2: The Source column lists the 10 diverse tracks I used in Loudness Revisited (Part 3), *audioXpress*, August 2017. The second column lists the values used to normalize each Short-Term “S” number to -16 LUFS. In Column 3, I applied the Magic Formula to the PLR and PSR numbers as measured by the Dynameter. The fourth column lists the correction to the third column numbers to normalize to -16 LUFS. This method also produced a normalized playlist as verified with foobar2000.

Toneboosters Loudness Meter —LKFS ITUR BS.1770/3				
Wave File	Integrated	Short-Term	True Peak	PLR
BB-01	-19	-16.3	-6.8	12.2
BB-02	-17.4	-15.1	-4.6	12.8
BB-03	-17.8	-14	-3.2	14.6
BB-04	-23.6	-19.6	-7.3	16.3
BB-05	-17.1	-14.6	-4	13.1
BB-06	-17	-15.1	-5.2	11.8
BB-07	-19.8	-17.6	-6.7	13.1
BB-08	-18.6	-16.3	-5.3	13.3
BB-09	-18.3	-14.6	-5.5	12.9
BB-10	-21.2	-18.7	-8	13.1
BB-11	-21	-16.1	-6.1	14.9
BB-12	-16.4	-14.3	-3.4	13
BB-13	-17.2	-15.1	-4.6	12.6
BB-14	-17.3	-14.1	-2.9	14.5

Table 3: The first 14 tracks of the TIDAL “Legends: Beach Boys” playlist shows several tracks with an Integrated loudness near -17 LUFS, which was probably the normalizing target. Track 4, at -23.6 LUFS, was still pleasantly loud but this may have been due to the music’s mood. The PLR values showed no correlation with the Integrated numbers.

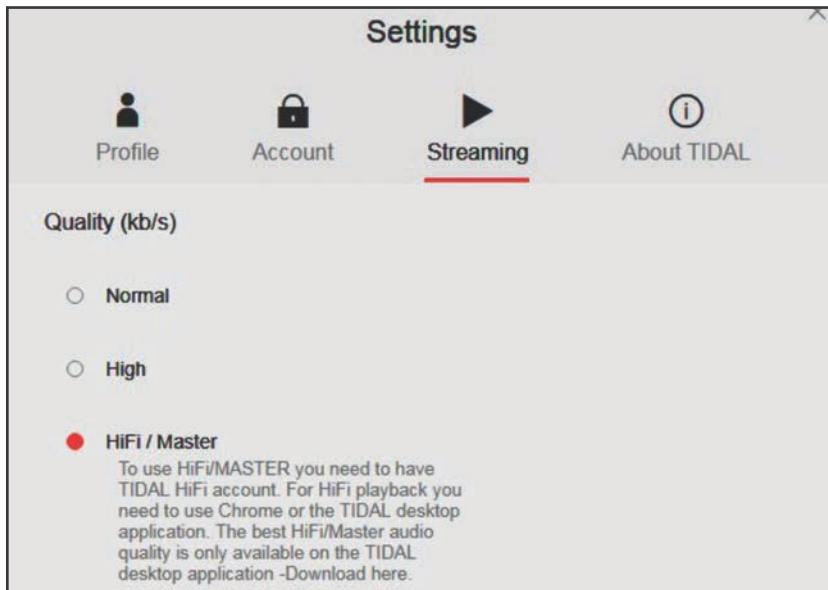


Photo 3: To the best of my knowledge, this TIDAL Settings screen offers the only opportunity to download the HiFi/Master desktop application, which is needed for the best quality audio.

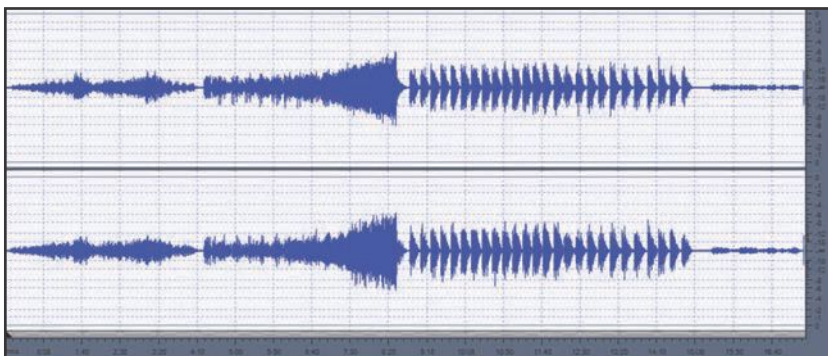


Photo 4: This screenshot of the first four “Classical Relaxation” TIDAL tracks illustrates the large loudness variation in this un-normalized playlist. It may be that older playlists are not normalized.

a Sonos-Connect or Sonos-Amp box because the DAC quality might be better than your computer’s sound card. An Ethernet connection is required to your computer but not necessarily to the Sonos box. Installation and setup is easy, just follow the on-screen instructions. Sonos has free control software for iOS and Android mobile devices, and Mac and Windows (version 7 and higher) computers.

Other Streaming Services

Table 4, an excerpt from “Music Streaming Comparison,” by Michael Potuck, was current as of March, 2017. I did omit Apple Music from my comparison for two reasons: the 256 kb/s bit rate and because its Sound Check volume normalization does not work very well according to an article by Alvin Alexander (an Adobe pdf copy is included in the Supplementary Material.)

It seems odd that he did not include Deezer. I think its screens are easier to navigate than either Spotify or TIDAL, but there is no mention of their using volume normalization. I expect this is just a matter of time. Qobuz was not included because it is not available in the United States.

Final Thoughts

I prefer Spotify’s Premium volume normalization over TIDAL’s HiFi. This is rather a shame because TIDAL’s HiFi is much “higher” with its FLAC file streaming. But the music streaming industry is still growing and changing rapidly. I expect this article will need serious updating in six to eight months.

Editor’s Note: Measurements for this article were performed in June and July 2017. However, TIDAL recently confirmed it has officially adopted the -14 LUFS HKU recommendation for mobile players (see more at

Project Files

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

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foobar2000

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Toneboosters

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Streaming Music Services — Compared					
	Apple Music	Spotify Premium	TIDAL HiFi	Pandora Premium	Google Play Music
Single user price per month	\$9.99	\$9.99	\$19.99	\$10	\$10
Family plan per month	\$14.99—6 users	\$30—5 users	\$30—5 users	N/A	\$14.99—6 users
Song catalog	40 million +	30 million +	25 million +	40 Million +	35 million +
Streaming quality	256 kb/S AAC	Max 320 kb/S Ogg	1,411 kb/S FLAC	N/A	320 kb/S
Offline?	Yes	Yes	Yes	Yes	Yes
Free Trial?	3 months	30 days	30 days	Varies	30 days

Table 4: This table, excerpted from “Music Streaming Comparison,” was current as of March 2017. It seems as of July 2017 that Pandora Premium is only available on iOS and Android phones that meet Pandora’s minimum guidelines.

www.audioexpress.com/news/ tidal-implements-album-loudness-normalization-and-activates-it-by-default-for-mobile-players.) This means that the loudest tracks of the albums will be aligned to -14 LUFS, the other tracks will keep their relative level. The TIDAL desktop app does not yet have loudness normalization at the time of writing. We learned that TIDAL is working on it.

The loudness metadata is, however, already streamed and third-party media player systems, such as Roon Labs, can read that data and merge it with the user set volume control. TIDAL will aim for -18 LUFS on the desktop app. This means that the tables on the article are still valid because the procedure remains the same, the “answers” should just change by 2 dB.

Next Generation Headphone Testing

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Rocky Mountain Audio Fest (RMAF) 2017

America's Benchmark High-End Consumer Audio Show

Rocky Mountain Audio Fest (RMAF) took place once again in Denver, CO, from October 6–8, 2017. This was the perfect event to hear the trends and experience new products at the Denver Marriott Tech Center.

By
Oliver A. Masciarotte
(United States)

With a gun shop and a shooting range next door to my hotel and a Whole Foods just down the road, Colorado is a study in contrasts. With tony suburbs slammed up against the rugged Sawtooths, rawness and refinement exist hand in hand. Same goes for audio at this year's Rocky Mountain Audio Fest (RMAF).

The first person I bumped into was my friend Matt Reilly from Audio Plus Services. After a brief catchup, he said the word "premier" and dragged me to their suite to check out Focal's new \$10,000 Kanta (www.focal.com), a good looking and sounding ported three-way floorstander premiering at the show. Kanta's truly distinguishing characteristic is not the Focal signature smooth yet extended delivery, it's that the model's molded polymer front baffle is available in eight colors!



Focal's modern Kanta is sure to please everyone in the family.

Around the corner, I was hailed by Luke Manley of VTL (www.vtl.com). Luke was excited about a new, entry-level preamp on static display. His \$3,750 TP-2.5i phono stage revealed VTL's thoughtful and solid construction, I'm sure it'll sound good as well. His active demo, combining VTL's \$65,000 Siegfried Series II hollow-state power amps with \$62,000 Vandersteen Model Seven MKII, a \$18,900 Sub Nine for low-frequency duties, and a vinyl front end composed of Brinkman's \$32,000 Balance turntable with Lyra's \$12,000 Atlas cartridge, a \$12,500 VTL TP-6.5 phono stage, and a \$25,000 VTL TL-7.5 Series III preamp, demonstrated how incredibly well an all-analog, megabucks rig can sound.

Speaking of analog, Cambridge Audio (www.cambridgeaudio.com) has been working on a new range of higher end electronics aimed at elevated performance. With a minimal signal path, intelligent biasing, and no caps to degrade the audio, its prototype component line is a departure from its normal budget-priced offerings. The stereo power amp is joined by a modern, all-inputs-welcome preamp that also builds some fresh thinking into the design and layout. Paired with a set of Bowers & Wilkins 805 D3 stand mounters, it was the most lifelike presentation I've yet heard from these guys, very promising.

The Adante, ELAC's answer to more value and fidelity for less, is now in full production and are shipping (www.elac.com). The three-way AF-61 "Floorstanding Tower," driven by Audio Alchemy electronics, sounded crazy good for the price. With a broad sweet spot, Adante produced exceptionally natural sounds for those on a moderate budget. As

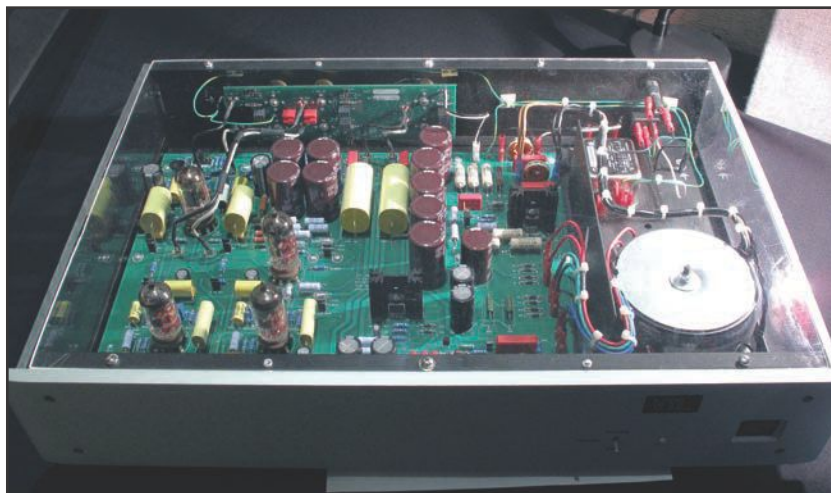
to sister company Audio Alchemy, inside the current casework was new circuitry. The DPA-2 amp now produces 650 W in mono mode by leveraging Hypex tech with special Peter Madnick sauce, and it does it nicely via a Class A input stage. The DDP-2, their next-generation Digital Decoding Preamp, now has more “DSP horsepower,” according to Madnick. In addition to Audio Alchemy’s well-known perceptual quality enhancement algorithms, the extra math power will support complete MQA unfolding. All that and more for about the same prices as the previous models. Also for the budget crowd, I got a demo of a complete Atmos rig assembled entirely from ELAC’s Unify line, and it too sounded great for, dare I say, a paltry \$5,000 outlay.

More Speakers

For more bucks with more performance, studio monitor stalwarts ATC (www.atcloudspeakers.co.uk) had the first US showing of its \$10,000 SCM19A active tower two-way speakers. Revealing yet not harsh, these little guys convey clean truthful sound. As with their other active examples, power is provided by in-house designed “...discrete MOSFET Class A/B modules, channeling 32 W to the high-frequency section and 150 W to the bass.”

Another speaker that delivered good sound was Paradigm’s \$10,000 Persona 3F, a ported four driver, three-way floorstander (www.paradigm.com). The Persona line atypically enlists Truextent beryllium in both tweeter and midrange drivers, (love those Fibonacci grills) plus dual voice coil double woofers. Paired with pre-production versions of Anthem’s STR preamp and 800 W into 2 Ω dual mono \$6,000 STR power amplifier, the slim tower from Paradigm provided exceptional soundstage depth and finely wrought transients...and they rocked Stevie Ray for my demo! The new two-channel STR pre comes with ARC room correction, DSD2 and DXD support via USB plus bass management for dual subs, balanced outs and MM/MC phono, all for \$4,000.

There’s something in France’s water that compels designers to build highly integrated, high-performance, monolithic gear. Micromega (www.micromega.com/en), the makers of some of my favorite budget wares, has moved upscale with its latest design. A far cry from its excellent, inexpensive MyDAC, the M-One is a slab of versatile integrated analog power, from the \$5,000 entry level M100, with 100 W, up to 150 W into 8 Ω for the \$8,500 M150. With a plethora of I/O choices, including Ethernet, Bluetooth, USB, AES/EBU, and HDMI, the higher powered model even includes Micromega’s own M.A.R.S. acoustic correction software. Paired with a set of \$14,000 Focal Sopra No2 in the demo



VTL's prototype TP-2.5i phono stage



Early prototypes of Cambridge Audio's as yet unnamed upmarket line augurs good sounds to come.



Audio Alchemy's reserved Peter Madnick and ELAC's irrepressible Andrew Jones hold court.

ATC's small yet exceptional SCM19A



room, M.A.R.S cleared up an upper bass chestiness and brought the soundstage into tight focus for a very engaging playback experience. Control is via an included remote or iOS and Android, and MQA support is in the works. Another plus: It can be ordered in seven standard finishes or more than 100 custom glossy colors.

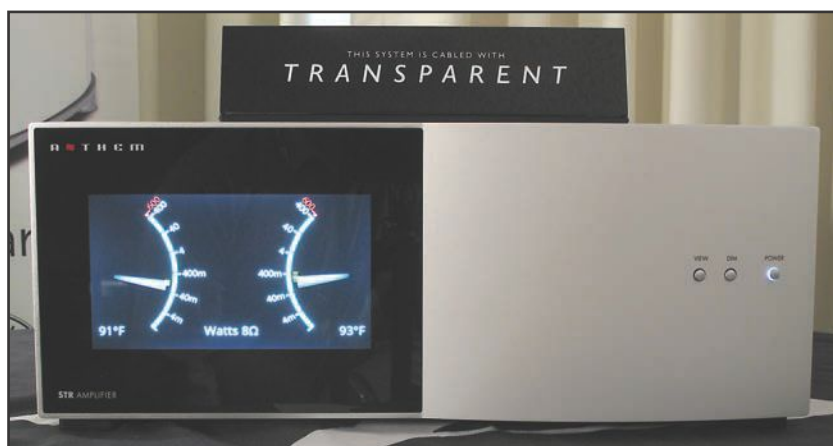
Usually sporting fat, natural aluminum facia, Jeff Rowland (www.jeffrowlandgroup.com) is defying its own convention with the \$4,000 HA 60. A Rowland-class headphone amp, the diminutive HA 60 also acts as a compact and less costly 60 W stereo power amplifier.

Unlike last year's RMAF, this show was short on crappy sounding rooms and long on excellence in audio. This year, I also acquired a new found appreciation for two brands which, in the past, were far from my favorites. The first was Dynaudio, a brand that I have not had much appreciation of, even its pro lines. But its new little Special Forty is something else again. Paired with an all-Ayre signal chain including a prototype of the new QX-8, they yielded a very pleasing yet exacting sound. As the name implies, the two-way stand-mounted Dynaudio is an anniversary special edition. At \$2,500 a pair, these are exceptional speakers but, with a single port on the back, you can't call them "bookshelf." By the way, the QX-8 integrated amp, which will run from about \$3,500 to \$4,500 depending on options includes, when fully loaded, USB, AES and Ethernet inputs, and can understand Roon, TIDAL, and Qobuz. Bonus birthday beauty from Dynaudio (www.dynaudio.com), and more yummy goodness to love from Ayre (www.ayre.com).

Another heretical opinion I hold is my displeasure, up 'til now, with what's come out from GoldenEar Technology (www.goldenear.com). I know, Sandy Gross is the quintessential audiophile speaker designer but I simply didn't "get it." I found his previous models to be timbrally weird, and their imaging vague. Well, I'm prepared to shut my mouth. His new \$8,500 Triton Reference, first shown at CES 2017, is not what I'd call "budget," but they exhibit a major leap in fidelity from past models while still being relatively affordable for such high performance. Even in the cramped quarters available, the 58" tall and 9.25" slim References, paired with all Hegel electronics, provided nicely controlled low-frequency response, with excellent imaging and even, credible voicing.

Personal Audio

Time to turn my gaze to personal audio, starting with Klipsch (www.klipsch.com). I'm a total fan of its Xi series of in-ears, but was surprised by a breakout new product, the beautifully constructed



The yummy STR power amp...Is Jacques-Arsène d'Arsonval spinning in his grave?



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Small, black, and unassuming, the HA 60 bucks the trend.

\$1,200 Heritage HP-3 over-the-ear headphone. Accompanied by a bundled universal headphone stand, 0.25" adapter and detachable cable, the semi-open cups sport sheepskin pads and are available in your choice of three solid wood species. The sound was signature Klipsch and in keeping with the price—accurate, extended, and smooth. Klipsch also showed a new DAC/HPA, the \$500 Heritage Headphone Amplifier. With an ESS Sabre32 ES90128K2M at its digital heart, and dual



The redoubtable Sandy Gross with his newest tower, the shiny Triton Reference

TI TPA6120A2 monolithic amplifiers driving its balanced and unbalanced analog outs, it sounded great and is very affordably priced. I particularly liked the very wide unity gain bandwidth, DSD2 support, walnut top and bottom, analog gain and real physical controls; long bat toggle switches; aluminum volume control and rotary selector; all clad in copper. These are not features one would expect at that price point.

Walking the aisles of CanJam, the Sennheiser (www.en-us.sennheiser.com) table stopped me in my tracks. Was that a set of headphones with, gasp, microphones on the exterior?! Yes indeed, the \$250 Ambeo Smart Headset serves as both single driver in-ear headphones and binaural recording rig. Currently for iOS only, the Ambeo Smart's cable terminates in a Lightning connector. The mic preamp has variable sensitivity and a soft knee limiter, and can be controlled manually or through its own app. As you'd expect with strategically placed mics riding on each ear, Ambeo Smart can also provide active noise canceling or "situational awareness," with external sounds mixing into the feed. Listening to a video capture of an amateur soccer contest with the product, I was pleasantly surprised by the naturalness of the audio, with a very believable match between visual and aural presentations.

Sennheiser also demo'd the new HDV 820 DAC/HPA in the US, first shown at Munich High End 2017. The HDV 820 is also based on the ESS Sabre, and is both DXD and DSD4-capable, and has crazy wide bandwidth. It has balanced outs and provides several choices for connector type, including XLR-3 XLR-4, 6.3 mm as well as the new 4.4 mm Pentaconn. Several manufacturers at the show supported



Klipsch's lovely sounding Heritage dynamic duo



Sennheiser's new Ambeo Smart Headset

Sony's TRRRS Pentaconn design. This emerging JEITA spec may become a new industry standard for fully balanced cans.

ZMF (www.zmfheadphones.com), known for fine quality closed back headphones, introduced Auteur at the show—its first open back design. Featuring the solid wood cups for which ZMF is known, the model will be available in several species options and a choice of perforated or non-perforated pads to dial in the perfect sonic signature. Pricing starts at \$1,400, and an oxygen-free copper cable is available for an additional \$60 when ordered with the cans.

Abyss (www.abyss-headphones.com), occupying the upper echelon of planar magnetic merchandise, has moved downmarket slightly with its new Diana, a \$3,000 lifestyle alternative to its iron maiden AB-1266 Phi. Machined from aluminum, with a matte ceramic coating and spring steel, leather-covered headband, these 40 Ω cans are now shipping in three finishes that will handily blend in on the street.

My current fave for wide band in-ears now ships in a wireless version. beyerdynamic's \$1,300 Xelento Wireless mates its high flux Tesla magnetics and super comfy, not quite in-ear fit with Bluetooth and true 48 kHz/24-bit aptX HD. Their never-tangle, silver plated cable terminates in a AAA battery-sized transceiver with a clip for tethering to apparel and over five hours of run time. An inline remote for taking calls completes the headset picture, and an included additional cable converts the Xelento Wireless into a classic wired in-ear (www.north-america.beyerdynamic.com).

Also new from beyerdynamic is the \$550 Aventho Wireless, one of the first of a new class of headphone that offers up custom DSP voicing. These wire-free versions of the on-ear T51i work in conjunction with beyerdynamic's MIY app, which generates a custom profile and stores it in Aventho's transceiver. MIY has a "Sound Watching" function, tracking exposure to stave off permanent hearing damage. Utilizing aptX HD, run time is spec'd at 20 hours before



Abyss sheds its geeky demeanor with Diana



beyerdynamic's Xelento Wireless—lightweight, comfortable mobile fidelity



Effect Audio's wide range of accessory cables

The new AR-M200 DAP



RHA's entry level MA390 at left, with its fancier MA650 wireless stablemate at right

recharge. One more item from the crew in Heilbronn is Impacto Essential, a tiny, DSD2-capable inline DAC/HPA with choice of USB-A, USB-C or micro-USB termination. A Lightning version is in the works.

With a broad range of aftermarket headphone cables, Effect Audio's product ranges start at \$149 and head north of a breathtaking \$1,500 for its gold-plated ultra pure silver, multi-gauge, multi-strand Horus (www.effectaudio.com).

Also at the show with pre-production prototypes, CTM (www.cleartunemonitors.com) let me listen to its high-end Da Vinci IX and X in-ears. As the name implies, the \$2,000 IX has nine balanced armature drivers and the X has 10! The case is CNC-formed aluminum, and the cables are oxygen-free pure copper. The Da Vincis will ship in the first quarter of 2018.

Acoustic Research (www.acoustic-research.com) has been busy since the last RMAF, with a new IEM, a new HRA DAP, and new planar magnetics. The \$200 AR-E10 in-ear phones combine an 8 mm beryllium dynamic driver with a single balanced armature. Frequency response for the 40 Ω unit is roughly spec'd at 20 Hz to 40 kHz, and a Pentaconn-equipped balanced cable is an option. Its \$400 AR-M200 DAP boasts Bluetooth aptX HD send *and* receive, a Class-A amp, individual grounding for left and right channels, and has both standard 3.5 mm and balanced Pentaconn outs. In Receive mode, a paired phone can stream data to the AR-M2000. Can't forget headphones! The AR-H1 is a new, \$600 circumaural planar magnetic headphone. With an impedance of 33 Ω , and vaguely spec'd 10 Hz to 70 kHz frequency response, this open back design was shown earlier at Munich High End 2017. A last Acoustic Research note: Its top of the line AR-M2 DAP has new firmware, and a lower \$1,000 price.

Those unconventional Scots at RHA (www.rha-audio.com) saw an opportunity in entry-level headphones, and answered the call with the new MA390 universal in-ear headset. With a brushed aluminum shell, universal remote, and understated good looks, these \$30 cans should have been in every Christmas stocking. Okay, some folks dislike encumbrances, and the MA650 fits that bill. The NFC/Bluetooth wireless MA650 headset is water resistant, has a 12-hour battery life, and includes a full Android remote. In case you're wondering, list price is \$60!

Compact Components

Let's turn to a different form of personal, the small form factor version. Those hollow-state folks from Texas, Raven Audio, had early versions of their new compact components on static display (www.ravenaudio.com).

About the Author

Oliver A. Masciarotte has spent more 30 years immersed in the tech space, working on facilitation, optimization, marketing, and product development for clients worldwide. As an author and speaker, he enjoys informing folks about technological best practices. More information is available at seneschal.net and othermunday.com.



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Two members of Raven Audio's Goldfinch Tabletop collection



The Goldfinch Tabletop Audio System is composed of four pieces, each in identical casework. The all-tube preamp, power amp and DAC with point-to-point internals are \$2,295 each, and the power amp develops 7.5 W into 8 Ω . A separate power supply for the three boxes is \$1,995 and, all together, the bundle runs \$6,995.

In Gamut's room, there was only a minor new item, a CCIR EQ option for its D3i preamp (www.gamutaudio.com). What interested me was the open reel deck they were using, a playback-only Lyrec FRED, originally designed as an editorial



Benno Baun Meldgaard and his monster Zodiac... (note the FRED reel-to-reel to his left)

workhorse. In Gamut's example, the internal electronics were bypassed, with the repro head directly feeding the new board. With their giant Zodiac flagships pumping out the tunes, the sound was, according to my notes, "...in your face, in a good way!"

Another brand new to me at this show was lumen white research (www.lumenwhite.com). Even though their White Light Anniversary model has been shown at previous shows, I was so taken by the model demo'd—the Kyara Lumenwhite, the White Light's baby brother—that I had to mention it here. Driven by an entire raft of new Ayon Audio hollow-state separates and connected via Synergistic Research cabling, these elegant towers yielded a very dynamic and resolving sound, one of the standouts at the show. At \$49,000, I can assure you that there were more expensive speakers at RMAF that I've not written about because of, in my own opinion, their laughably atrocious sound. This model is a tall slim, three-way design, with custom Accuton ceramic mid and low-frequency drivers. An interesting feature is variable acoustic termination of the slotted rear port for optimal in-room response.

Another unknown brand that rocked my world: Vehement (www.vehementaudio.com). As with the Focal Kanta, the \$7,000 Brezza Savant loudspeaker employs a solid polymer baffle, isteppped for time alignment. With dual custom 6" SEAS paper low-frequency drivers in addition to a RAAL ribbon tweet, the Brezza Savant manifest a wide, nicely delineated soundstage and extended but well-behaved high-



Lumenwhite gave me reasons to soldier on...

frequency regime. The bottom was a bit monotonous but it could have been the room. Driven by Exogal components, this was one spiffy speaker.

I was recently made aware of JERN (www.audioform.dk), a company know for its curvy cast iron enclosures. Microfactoid: Jern is Danish for iron. I had not yet heard any of its offspring, so RMAF gave me a chance to listen. The room setup included a single REL Acoustics T5i subwoofer, but I asked that it be disabled. The little, premium JERN14 ES was on active demo and they had a very pleasant hi-fi sound, in part due to Mundorf crossover components and other upgraded innards. Sitting on a toroidal rubber base for easy orientation and aiming, these \$1,750 acoustic suspension speakers are true bookshelf models, with no vents to mess up the low end. For tabletop use, a solid wood tripod is available, making for a stylish audio accent in any modern home. At the show, JERN's Steve French showed me a prototype on-wall, the model 8000. Imagine a giant tapered hockey puck, a very substantial two-way puck in particular. Pricing will run from \$500 to \$800, depending on component complement. A 14 plus 8000 pairing might be an interesting choice for an Atmos install.

Overall Impressions

There were many more interesting products at the show but, due to time and space limitations, some are on the cutting room floor. My apologies



The JERN 8000 on-wall is a 20 lb truncated cone of ferrous fun.

to those I left out. All in all, this year's RMAF was a refreshing change from last year's chaos, generally bad sounds, and inconvenient layout.

I was pleased to learn of several new companies, while my old favorites have only extended and improved their lines. Look for future reviews of some of the gear I've mentioned, and Stuart Yaniger's review of Vanatoo's Transparent Zero elsewhere in this issue. Until next time, keep listening! 🎧



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Audio Editing Software Roundup (Part 2)

Affordable Tools and Free Software

By
Fernando Rodrigues
(Portugal)

Iced Audio AudioFinder

Iced Audio AudioFinder is another commercial product that caters to its own niche market. Exclusive for Mac OS X (compatibility started in OS X 10.4, but the current version demands OS X 10.8 or higher), the application is intended for the music collectors and sample/sampler enthusiasts, (which reminds us of another one that exists exclusively for Windows, but was left out of this roundup—Awave Studio, from FMJ-Software).

AudioFinder works exclusively with stereo files. Resolution goes up to 32-bit float and sample rates supported go up to 192 kHz. As for audio file formats, AudioFinder doesn't go very far. Still, we can say that it supports the ones that matter for its target audience (e.g., Apple Loops, CAF, AIFF, AIFC, SDII, BWF, WAV, REX, RX2, MP3, ACC, M4A, MID, SYX, FLAC, and Ogg Vorbis). The fact that it supports both REX and Apple Loops, as well as MID and SYX files means that AudioFinder is a kind of hybrid application—half sampler and half audio editor.

As for plug-ins, AudioFinder supports basically everything: AU, VST, VST3, MAS, and AAX (with a very good Plug-In Manager). The support of the MAS format is a curiosity, since that format is exclusively used by MOTU (Digital Performer), and can be

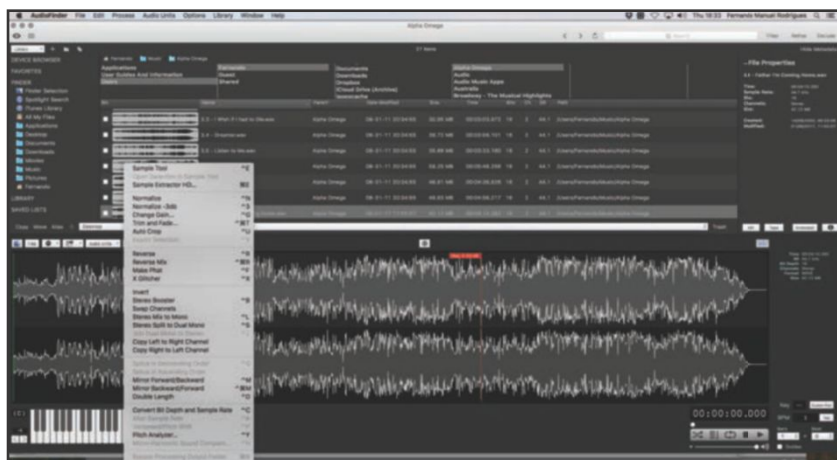
considered a kind of legacy. It also comes with several processors, basically related to sampling and looping.

It features some integrated DSP processing, also related to sampling and looping—its main application target. Regarding sampler support and looping support, it directly supports Pro Tools and Logic Pro and the Logic sampler EXS24 format.

AudioFinder also includes an integrated Sample Tools trim/loop/fade editor with beat slicer. Yet, it has no specialized loop tools, which we found surprising, considering its focus on samplers (Awave Studio, for example, has a dedicated loop editor window). But it has a good resampling algorithm.

As for special features in this program, we can mention a sophisticated Metadata Database with Database Notes for keeping track of things. It also has its own peak analysis and pitch analyzer tools, and Micro-Harmonic Sound Compare. Finally, it can find duplicate sounds by analyzing the sound content (not just by file name)—a useful feature.

For the sampling and sampler enthusiasts in the Mac world, this could be their tool of choice, as it combines some looping tools with good file management and metadata editing. In Windows, although not part of this roundup, the closer counterpart would be FMJ-Software Awave Studio.



Iced Audio AudioFinder:

www.icedaudio.com/site/#about
Mac OS X 10.6 and later: \$69.95

There are older versions available to ensure compatibility with older versions of OS X, from 10.4 up

FMJ-Software Awave Studio:
www.fmjsoft.com/awavestudio.html#main
Windows XP and later:

\$80/€ 70,00 (+ VAT for EU citizens)

Aurchitect Triumph

Formerly from Audiofile Engineering, and recently sold to Aurchitect, Triumph is a commercial audio editor that is very much focused on audio mastering. The current version only runs on Mac OS X 10.7 or later. Despite being a program dedicated to mastering, it supports up to 48 tracks. Audio resolution goes up to 32-bit floating and sample rates supported go up to 192 kHz.

In terms of audio file formats, Triumph doesn't go far, but we can say that it supports the formats that matter to its focus: AIFF, WAV, WAV64, Sound Designer, CAF, µlaw, MP3, M4A, FLAC, Ogg Vorbis, and DDP. With regard to lossless compressed formats, only APE is missing.

The only plug-in format supported is AU. It comes with some audio processors—essentially utilities—analysis and monitoring plug-ins (the analysis plug-ins are very good, basically they're the tools that are present in the Specter Realtime Audio Analysis Suite, also from Aurchitect).

Being an audio editor focused on mastering, it doesn't include any support for samplers, yet Triumph still has some loop tools and a loop window—quite good. Anyway, Aurchitect has another application especially tailored for loops, called Loop Editor. We sincerely hope they keep developing this one to reach the level of something like the much-missed Infinity, from Antares.

Mixing and automation are dealt via a feature called "shapes," which are a set of "curves" to craft fades and automations. There is another feature called "SmartEdits" that allows for more detailed editing with fully adjustable properties (e.g., fade-in and fade-out, fade length, etc.). Besides these, Triumph also includes some special "Actions" (kind of "macro editings" based in AppleScripts). Actually, Triumph has very good support for Apple Scripts, which enables users with enough skills to create customized macros.

Being an audio editor oriented to mastering, it's no surprise that Triumph includes a specialized DDP Player, as well as integrated CD recording and creation of DDP files. However, it doesn't support Batch Editing—again, Aurchitect offers another specialized application for this called Myriad (a powerful Batch Editor that perhaps any professional should buy).

Time Compression/Expansion and Pitch Shifting are done through

Varispeed Playback (basically, like a sampler). Transposition (Pitch Shifting) can be controlled through MIDI. Despite being a really powerful audio editor, there isn't any kind of spectral editing available.

Something worth mentioning about Triumph is the "Layer" concept. This was already present in the previous Wave Editor from Audiofile Engineering. Triumph's layers are like image editing in technique but specifically tailored for audio. The Layers method provides a new way to create combinations of sounds and saves time by keeping everything live and editable until the final product—if there's some processing that we feel doesn't fit, it's as simple as turning off that layer, and all the associated processes are suspended. This is perhaps the most important and original feature of the program. However, sometimes it can be somewhat convoluted to use, especially before you become familiar with it.

Triumph also includes some special tools such as First Aid (to try/help recover damaged audio files), a Tuner, and the ability to directly import/export to/from Final Cut Pro.

It also includes a technology called FHx, which Aurchitect says "creates a more spacious, natural soundstage over headphones." The new Render & Ship feature also enables users to render multiple formats simultaneously, which may be a time-saver for professionals.

Aurchitect Audio Software Triumph:

<http://triumph.aurchitect.com>

Mac OS X only – 10.10 and later: \$79



About the Author

Fernando Rodrigues began studying music and technology in the 1970s. His goal was to marry his two passions: music and computers. As a student, he helped assemble the electronics music studio at the music college in Porto. Later, he directed the technology department at one of Portugal's major distributors while pursuing a career teaching musical analysis and composition techniques. He now concentrates on research and writing about music and technology, sharing his own perspectives about music and sound.

TwistedWave

This is another commercial audio editor that is Mac only (with the curiosity of also being available for iOS and online). Currently, it is compatible with Mac OS X 10.6 or higher. Its main characteristic is the fact of being straightforward, and with a very clean (almost simplistic) interface.

However, it has some hidden power. Apparently, there is no limit for the number of tracks (we kept adding until we reached 100 and stopped there—not that we find this feature useful in an audio editor, since as far as it can accommodate the surround formats, in our opinion it is enough). Resolution goes up to 32-bit floating, and sample rates go up to 192 kHz (so far, up to the standards).

It imports and exports a wide range of formats: WAV, AIFF, AU, SND, SD2, MPG, MP2, MP3, MP4, M4R (iPhone ringtones), M4A (iTunes), M4B (audiobooks), AAC, CAF, FLAC, Ogg/Vorbis, WMA, WavPack, and Wave64. It can also import sound from video in formats (e.g., MOV, AVI, FLV—Flash Video—WMV or MPEG).

Plug-in formats supported are AU and VST2, and the program has Effect Stacks (chains of effects). However, it doesn't include any plug-ins of its own—clearly, the authors expect users to have their own stock of plug-ins for this task.

There isn't any support for loop editing whatsoever, nor for samplers. This is completely audio oriented, and the sample/sampler part of audio editing is out of reach for it.

On the other hand, it is clearly oriented for the creation of podcasts. In their own words: "Radio package editing becomes very easy with the clip list. In just a few steps, you can quickly zip through a long recording, select parts of the wave you are interested in, and copy them to the clip list. You can then create a new document by pasting your clips from the list. The clip list can also act as a playlist."

It includes a good time compression and pitch shifting algorithm (ZTX) licensed from Zynaptiq, which are among the best DSP programmers in the field. It also includes algorithms for noise reduction, de-esser, and vocal removal.

Although we can zoom in up to sample level, there is no special tool (e.g., the pencil) to perform detailed editing at this level.

As special features of this program, we should mention the automatic silence detection, split by markers, detecting transients, and special pasting (with three options: Insert, Mix or Replace). It can also automatically fade in and out when copy/pasting using insert, to make the transitions smoother. The first two can be useful when processing LP recordings, allowing for some automated separation of a single recording in separate tracks.

It can also import audio from video, but only the regular OS X formats. Finally, it has a

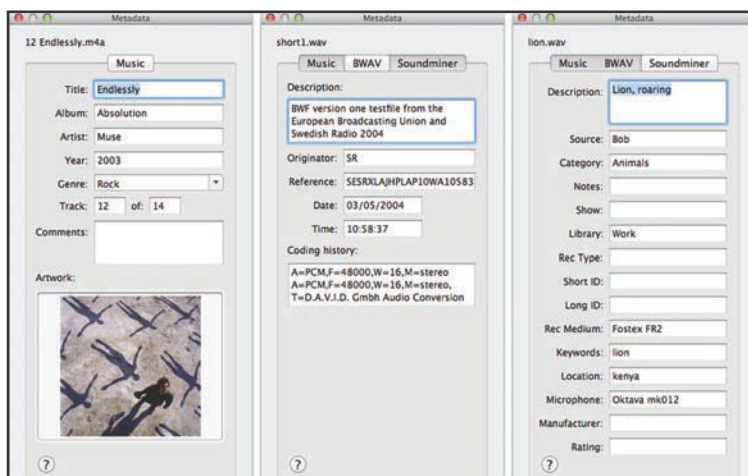
very good batch processor, which enables us to perform all the usual aspects for what a batch could be used (e.g., saving a final master in several optional formats). This can even be used to create waveform images of the sound files (now, this is original).

TwistedWave is a favorite by many audio engineers because of its streamlined and clean interface, allied to the good batch routines. Sure, it lacks in some fields, but for someone with a good collection of plug-ins (which basically includes all the audio professionals that deal with audio technology), it offers a solid host that can accommodate almost any audio file and deliver a good final result. For those looking for something more specialized, though, it has little to offer.

TwistedWave: <https://twistedwave.com/mac.html>

Mac OS 10.6 and later: \$79.90

There are older versions available to ensure compatibility with older versions of OS X, from 10.4 onward



TwistedWave can read and save music metadata from files in the AIFF, WAV, MP3, MP4, FLAC, and Ogg/Vorbis formats. Additionally, TwistedWave supports BWAV metadata, as well as Soundminer metadata for WAV and AIFF files.



2nd Sense Audio ReSample

ReSample is a commercial audio editor, available cross-platform (Windows and Mac OS X) in both 32-bit and 64-bit versions. A very recent launching from 2nd Sense Audio (it's only at version 1.1.x), it is still a little undefined in terms of features and targeting, in our opinion. It seems like a "Work In Progress" and we will probably see more features added as time goes by. (We already saw some during our testing period.)

The program works with an audio resolution of up to 32-bit floating, and with sample rates up to 384 kHz (384 kHz and 32-bit floating are becoming more and more the standard among audio editors). In terms of formats supported, there aren't many (fewer than we can see in older applications). The program supports linear PCM (.wav and .aif), FLAC lossless compressed file format (.flac), Ogg Vorbis lossless compressed audio (.ogg), MP3 file format, obviously (.mp3), and Apple Lossless Audio Compressed format (.m4a).

In terms of plug-ins, it supports VST2 in Windows, and VST2/AU in Mac OS X. Plug-in support is transparent, through scanning, as it happens in more modern applications. The program scans the system defined plug-in folder and lists the validated plug-ins.

Unfortunately, the scanning process is prone to errors, especially when we also have virtual instruments in the same folder—the program starts over every time. As we already said, plug-in scanning is often the weakest point in audio editors. The solution was to create a special VstPlugIns folder with a cleaner set, which was then successfully scanned (a solution we had to follow several times, even with Sound Forge).

Besides third-party plug-ins, ReSample ships with a comprehensive collection of good quality audio processors (more than 21). Actually, 2nd Sense Audio also made these available to purchase separately as plug-ins, although ReSample owners will not have these available (apparently, the ones in ReSample are built-in, therefore, not available outside the application). Among the processors available, we have Parametric Equalizer, Vocal Removal, Time Stretch and Pitch Shift, Reverb, Noise Reduction, Engineering Filter, and more. The amount and quality of the audio DSP is one of the strengths of this program, but considering the quality we usually have at our disposal, and probably already own, especially when we are users of DAWs such as Cubase, Digital Performer, and SONAR, 2nd Sense Audio needs to offer something extra to make the program appealing.

There isn't any support for loops, loop information, or for samplers (being a new audio editor, we weren't expecting any special tools, but at least support for reading the loop chunk already recorded in the audio files header should be implemented). We may play a selection looped, but there isn't a way to save loop points into the audio file. With regard to automation, we only have customizable fade curves. There also aren't any features connected to CD recording or deliverables preparation (e.g., playlist and metadata editing), which is strange, since this program should have mastering engineers among its target users.

Time Compression/Expansion and Pitch Shifting algorithms are good, and we have live preview, but there are no special features (e.g., formant correction). We also have Noise Reduction among the processes available.

Another strength of the application is the speed and smoothness of screen redraw when browsing audio, as well as the ability to select audio through multitouch trackpad.

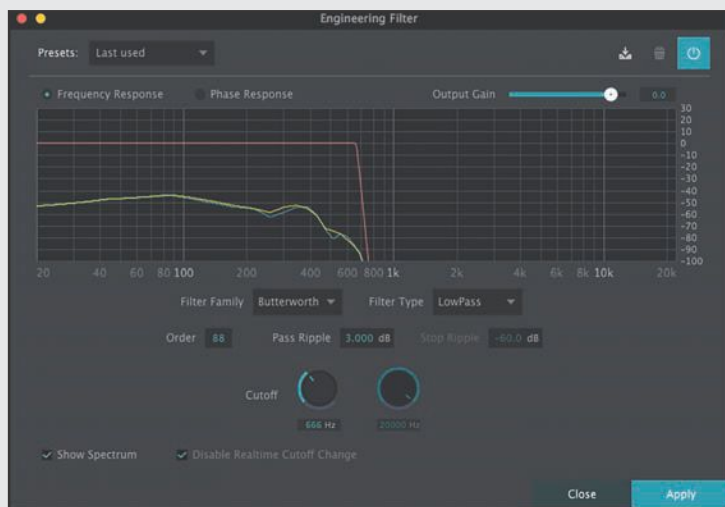
The analysis tools are very good and comprehensive. We have an oscilloscope, a spectrum, a phase scope, and a loudness meter. We also may have double screen with waveform and spectrum view available simultaneously. This is good for monitoring, but again, we miss some tools to take advantage of the spectrum display and to perform some special editing, as we have in other applications.

Special features available include Vocal Removal (it only works when the voice is on the center channel) and the ability to apply compressing/expanding effect to a specific frequency band.

2nd Sense Audio ReSample:

<https://2ndsenseaudio.com/resample>

Windows 7 and later / Mac OS X 10.7 and later: \$89.00



ReSample ships with more than 20 factory audio processors and effects, including this fast Engineering Filter, with up to 100th order Butterworth low-pass.



Wavosaur

Wavosaur is a free audio editor for Windows only. There are 32-bit and 64-bit versions. It works with Windows XP or later, including Windows 10. Supports mono, stereo or multichannel WAV files, with resolutions from 8-bit up to 64-bit floating, at any sample rate. Audio output goes up to 192 kHz at 24-bit.

The program reads and writes WAV, AIFF, MP3, IFF, AU, SND, VOX, VOC, S1000 samples, Ogg Vorbis, Wavpack, and raw files.

It supports VST plug-ins, with a simple but very direct way to deal with them. There are no lists, no automatic scanning, etc. We open the Plug-In Manager and point to the directory we want to be scanned. It then scans the folder (and optionally also sub-folders, and either creates a new plug-in list or adds the plug-ins to a pre-existing list).

It may be a little more time consuming, but we were able to isolate the problematic plug-ins and create a plug-in list without problems with all the plug-ins we wanted to use. Time consuming, yes, but proven safe and we just have to do that once. We also have the option to add just the plug-ins we want, if that better suits our workflow. Someone who knows his/her system, and has the plug-in folder well organized will easily find what he/she wants.

We apply VST effects using “racks” where we can have one or more plug-ins. Then, we simply must be sure that the “Processing” check-box is ticked. We like this method better than the convoluted ones we found in other packages (e.g., Audacity and Gold Wave). Simple but efficient, and it works.

As for internal DSP processing, we can find some handy functions, like auto detecting of regions (using silences after more than 1 second of audio), normalize, stereo-to-mono and mono-to-stereo, silence (inserts silence in a selected region), fade-in and fade-out (but with no controls), and some non-real-time analysis, including statistics, automation volume envelopes (graphic), and so forth. Not very comprehensive, but the basic tools are there. Anyway, now we mostly rely on plug-ins for this task so, most of the time, these tools are redundant.

There are some loop tools, but they are very simple and no loop editing window (yet, we have a crossfade loop tool). Although not the best in this chapter, this is more than we have in other audio editors. There isn't any sampler support either. This is another audio editor that seems more focused on audio conversions and restoration. The analysis tools are great, however.

Regarding automation, we have volume envelope automation, with multi-point envelope editing. Also, many Wavosaur commands can be triggered by an external MIDI controller: Play, Stop, Record, Rewind, Fast Forward, toggle windows, go to markers, go to start and control output volume. We also have a pencil (Draw Tool) to manually perform editing at sample level. Wavosaur has a spectral view window, but no editing support at spectrum level.

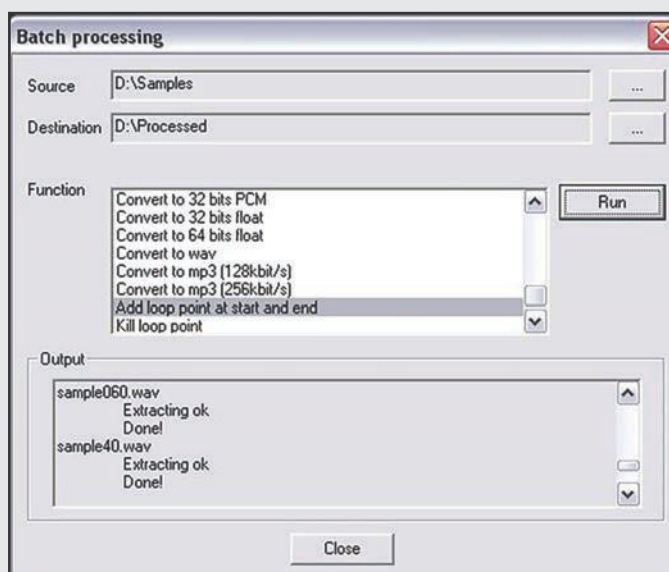
Regarding analysis tools, the program is well equipped. We have detailed statistics (RMS power, minimum and maximum value per channel) that we can export as a TXT document, frequency analysis with 2D spectrum and 3D spectrum, a sonogram, a real-time oscilloscope, a spectroscope, and a goniometer for monitoring audio input and output (not very good—slow video refreshing).

We also have some synthesis features, such as a waveform generator, a frequency impulse train (useful for calibration and measurement), and frequency sweep.

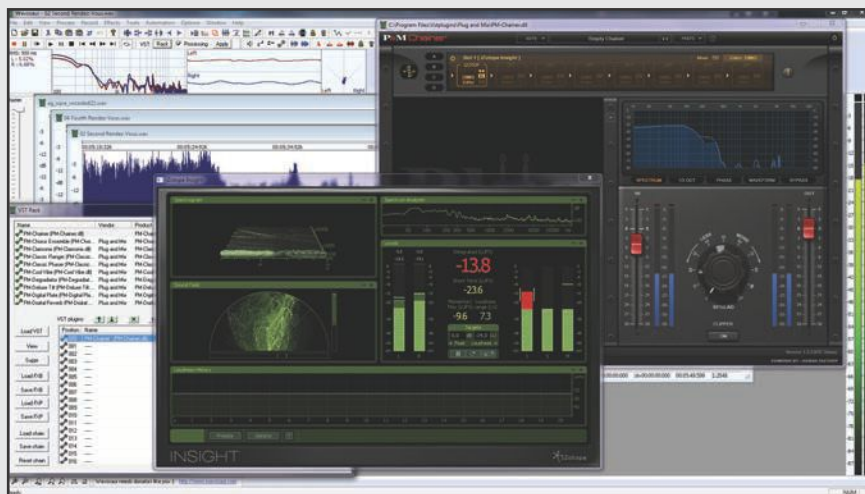
Considering this is a free audio editor, we found it to be surprisingly good and effective.

Wavosaur: www.wavosaur.com

Windows XP or later (32-bit and 64-bit versions): Free



Wavosaur includes a valuable batch processor allowing to process many files at one time, including auto trim and adding effects to an entire folder of files.



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Vanatoo Transparent Zero Powered Speakers

During the AXPONA 2017 show in Chicago, IL, Stuart Yaniger visited the Vanatoo room and was greatly impressed with the small and inexpensive proposition of the Transparent Zero powered speakers. As he noted in his report for our *Audio Voice* newsletter, they “seemed to punch far out of their weight class. I’d love to get a pair into my lab for extended listening and measurement.” That’s precisely what he did, and accounts for here.



Photo 1: The Transparent Zeros’ shape is a bit... unusual.

By
Stuart Yaniger
Photography by Cynthia Wenslow

Transparent Zero Wireless Speakers

Vanatoo
1600B SW Dash Point Rd.,
#51
Federal Way, WA 98023
855-771-1161
info@vanatoo.com
Price: \$359/pair

The Vanatoo Transparent Zero wireless speaker is an exemplar of the old saying, “Good engineering costs no more than poor engineering.”

Before diving into the review and measurements, I’ll wax philosophical a bit. The notion that in the audio reproduction chain the loudspeaker and the amplifier are a system, with the performance of each locked into the characteristics of the other, is a long-known truism. Ideally, the amp and the speaker would be sold as an integral unit (in an electrical sense), with the amplifier designed specifically for the requirements of the loudspeaker driver(s) with which it’s paired. That’s basic engineering logic, but the market hasn’t traditionally worked that way.

Through most of the past 50 years or so that I’ve been involved in audio, the audiophile consumer has demanded separates, with the

random mixing and matching of the speaker and the amp generally not optimized. To do it right, a consumer would need to know the optimal amplifier source impedance for the speakers, the speaker’s dynamic (large signal) impedance curves, the amplifier drive limits with load variation (real and imaginary parts), and the corresponding speaker drive limits (excursion and thermal). Although this is necessary to know to properly do the job, in reality this is something that can’t possibly be done correctly by a non-engineer who isn’t even armed with the needed data. The best one could reasonably do is to buy components that were at least engineered under the same aegis (and thus, the popularity of all-Quad or all-McIntosh systems with the carriage trade back in the day).

Nearly all attempts at integration by audio manufacturers resulted in market failures—

audiophiles wanted the sense of participation resulting from swapping amps, preamps, cables, and speakers to get the “sound” (or the illusion of the “sound”) that they sought. This was all part of the entertainment end of the audio industry, which forced speaker designers to engineer with a particular paradigm of what the power amps were likely to be. Likewise, power amp designers had to balance off cost and complexity vs. universality. For example, should the amp take into account the occasional badly engineered high-end speaker having impedance dips to 1 Ω ? Should the amp be designed to be unconditionally stable irrespective of the load reactance, and what’s the cost of doing so to the 99% of users for whom that’s not an issue?

In recent years, the traditional component audio market has devolved into a niche which, judging from audio show attendance, mostly caters to older audiophiles who are still set in their ways. But from a technical perspective, things are getting better—several major sociological changes have pushed the market toward the more rational integrated solutions. The advent of computer audio and smartphones, music streaming and downloads, the availability of efficient and inexpensive digital signal processing (DSP), and Class-D amplification have all contributed to a veritable “Great Leap Forward” in performance with a concomitant reduction in cost. This has been achieved by ignoring tradition and integrating amplification and custom DSP into complete speaker systems (in the truest sense of the word “system”). There are now dozens of fine examples on the market that have achieved technical and commercial success.

The Speakers

All of this came to mind for me during this year’s AXPONA show in Chicago. As I went from room to room, I heard a lot of expensive mediocrity, marinated in pretension and six figure price tags. Only a few things really stood out, but one of them was the sound I heard in the Vanatoo room, with two models of inexpensive active speakers that sounded far better than their price tags implied. After talking with Rick Kernan, the electrical engineer who is also one of Vanatoo’s two partners (the other being mechanical engineer Gary Geschellen), about the basics of the designs, I arranged for a pair of their least expensive (\$359 per pair) speakers, the Transparent Zeros, for review. A couple weeks later, a small box arrived with two speakers, a power brick suitable for either 120 or 220 VAC, a remote control, and an Ethernet cable to connect the right and left speakers together. There was also a laser-printed manual, which I actually took the trouble

to read—and it was a good thing I did! The Quick Start guide will get you through the basics, but some time spent with the available options will pay off in getting the sound from “well, it’s there” to “surprisingly good.” I should also mention that the package includes some thin foam rubber pads to isolate the speakers and prevent them from rattling against whatever hard surface they’re placed on (TV stand, desktop, or speaker stands).

The Transparent Zeros are a two-way system with a 4.5” bass-midrange cone driver, a similarly sized passive radiator, and a 1” soft dome tweeter, all housed in an enclosure smaller than a lunchbox. The enclosure is oddly shaped in a roughly trapezoidal manner (see **Photo 1**), and includes a plastic frame member that doubles as a prop for the speakers when they’re used on stands to allow the passive radiator to do its thing unimpeded. They are set up to be used on a desktop for computer sound, next to TV sets for home theater sound, and on stands as free-standing mini-monitors, depending on the orientation and the DSP settings. On a desktop (see **Photo 2**), the passive radiator points up, and the plastic prop can be used as a handle—or removed if the esthetics bother you. For mounting on stands, the passive radiator points down, with the prop giving it space to emit.

There’s a ton of features and flexibility packed in here, so I hope you’re sitting down and comfortable before reading through this.

The speakers have no passive crossovers, but are each bi-amped with D2Audio DAE3 Class-D digital input amps. These are described as “direct digital,” meaning that the amplifier runs in the digital domain right up to the output of the PWM drive. The speakers’ analog input is fed to an A-to-D converter, and the optical and Bluetooth are fed directly into the amps’ digital input. DSP is built into the D2Audio amp, with processing at 24 bit and 48 kHz. All signals coming



Photo 2: For desktop or TV stand use, the speakers are oriented with the passive radiator pointing up.

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.

into the amp get converted to and processed at this resolution.

In the speaker pair, one side (by default the Left speaker, though this can be changed by the user) contains all the electronics and sends analog signals to the Right speaker. A nice detail is that the Right speaker without the electronics has a solid plug in it so that the volumes (and hence bass alignments) of the two sides are equal. The back panel of the active side (see **Photo 3**) is a bit busy, but it has the amp/DSP inputs (analog and optical), a button to activate Bluetooth pairing, and a power input jack to go with the supplied power brick. The Ethernet connector is not used for any digital signals, but rather it is used to run a supplied RJ50 cable carrying the woofer and the tweeter analog signals from the electronics in the Left speaker to the drivers of the Right speaker. One can use a standard RJ45 cable if you need a longer run than the approximately 3 m provided, but the higher resistance could cause audible imbalance between the channels. Vanatoo offers a 7 m RJ50 cable as an option, and I'd recommend springing for it.

The V-T-B switch selects the function of the control knob: volume, treble, or bass. Generally, you'll set Volume to its maximum, then use either the source player or the supplied remote control to adjust volume. But here's where the complexity sets in: In order to take advantage of the DSP's

flexibility and the other speaker modes, you have to do a little dance with combinations of Volume knob, V-T-B switch power, and the Bluetooth pairing button. For example, to switch from Shelved response to Flat response, you unplug the power, set the Volume knob to the three dot position, set the V-T-B switch to V, hold down the pairing button, then plug the power back in and wait a few seconds before releasing the pairing button. Vanatoo provides a chart of all the button and knob combinations for parameter adjustment (I taped it to the side of the active speaker), but this is not a simple process, and cries out for a smartphone app with a control panel. Likewise, using the Remote to change any parameters requires a similar sort of button sequence and timing. I suppose the price of flexibility is complexity.

Inside the Speakers

Moving on to the internals, the crossover is done in the digital domain, and is an eighth-order Linkwitz-Riley (L-R) at 2,200 Hz, unusually low for such a small speaker. This is made possible by the certainty that you have with an active crossover, very steep slopes—which effectively keep low-frequency signals out of the tweeter—and the fine-grained control of the DSP. This low crossover point also enables the woofer to cross over below the frequency where it starts to become directional.

During the design phase, Vanatoo measured the average listening window of multiple pre-production woofers and tweeters to know what to expect in production. The average of these determined the low-pass and high-pass eighth-order responses. L-R crossovers are well behaved in the vertical off-axis, and the steep crossover makes for a narrow transition zone. With the low crossover point assuring good horizontal off-axis behavior, the combination would be expected to act pretty closely as a point source at any reasonable listening distance.

The woofer motor is underhung, with a 4 mm coil and an 8 mm gap. There's also an aluminum shorting ring to reduce distortion from eddy currents. The goal of these design elements is relatively low distortion for moderate cone displacements.

The DSP has limiters built in. The limiters are enabled by default, but can be turned off by the user. They only kick in at the last 3 dB of signal, so the overall sound has no limiting until the speaker starts to get near the system limits. When the limiters activate, it only dulls the peaks a little to keep things under control, so its effect is relatively subtle. No damage will occur to the system with the limiters off. They primarily protect the power supply from



Photo 3: The rear panel of the Transparent Zero contains all of the I/O connections and switches and knobs for programming.



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reaching its current limit, which can happen on occasion with the limiters off. This is a “soft failure,” as the power supply will cycle and the system will normally resume playing music in about 10 to 15 seconds. Presumably an informed user should realize that if the protection circuit cuts off the sound, they need to turn it down a little.

A subwoofer output is available and can be configured for an 80 Hz or 125 Hz crossover point. The output is low-pass filtered with a fourth-order L-R, and the appropriate high-pass filter for the Transparent Zeros is automatically activated when the subwoofer is connected.

The Setup

Used with our TV set, connected via the Analog input and set to Shelf mode, the Transparent Zeros nearly disappeared sonically, giving a good localization of images. What really struck us was how revealing they were of the different microphone signatures and choices made for signal processing by various TV shows, ranging from compressed and hot to remarkably realistic. The use of faders and pan pots during football games was particularly noticeable, which may or may not be a good thing! We never felt the need for a subwoofer, but admittedly, we don’t watch blockbuster movies, for which a subwoofer could be a real asset. I don’t think of TV as a true hi-fi source, so I adjusted my expectations accordingly—nonetheless, the Transparent Zeros received a high compliment when, after I took them downstairs to my basement lab for critical listening and measurement, my wife wanted to know when I was bringing them back up.

Next up, in the same mode, I hooked the Transparent Zeros to my laptop via the USB connection. This gave a most satisfactory sound when I played well-recorded music through them, and I ended up using them for mastering monitors for home recordings. I’ve seen some gamers complain that the Transparent Zeros lacked “excitement,” and I can absolutely see that—the basic sound was clean and uncolored, so for gaming use, an equalizer

or similar plug-in might come in handy if the Transparent Zeros’ tone controls aren’t adequate.

There’s no question that the Transparent Zeros are good enough to beg for use as “serious” hi-fi speakers, which means putting them on speaker stands, setting the Mode to Flat, and experimenting with room positioning. Being a relatively non-fancy person, I used some non-fancy “universal” stands I purchased from Amazon, which were pretty typical, comprising a steel base, an adjustable height pole, and a top plate with clamps to secure the speakers. Here’s where I ran into some issues, though solvable ones.

The Transparent Zeros’ odd cabinet shape and the plastic props means that conventional speaker stand clamps won’t work. Nor can you rest the speakers on the stands unless the top plate is exceptionally deep. At the suggestion of Vanatoo, I jury-rigged a solution: I took some 0.5” wood slats, cut them to the same width and depth as the Vanatoo rubber pads, and then clamped the slats to the speaker stands. The rubber pads and the speakers then rested on the slats. This worked from a sonics point of view, nothing rattled or moved around, but the lack of secure clamping means that in a household with pets or children, the speakers’ security is a precarious thing, and a crash to the floor is not an “if” but a “when and how bad.” After a close call with the swinging tail of our Great Pyrenees, I decided that these were not for living room use in our household, and moved them downstairs to the dog-free lab. Some redesign of the plastic props, or the inclusion of alternate props for stand mounting would be a blessing to those of us who want to use the Transparent Zeros as serious hi-fi speakers.

Sound

With the physical setup complete, I hooked these up to do some listening. Initially, I used an optical connection between an RME ADI-2 Pro DAC that I was finishing up reviewing, then once that was sent back, I connected to my laptop via Bluetooth and to my lab computer via USB. The lack of a coaxial connection

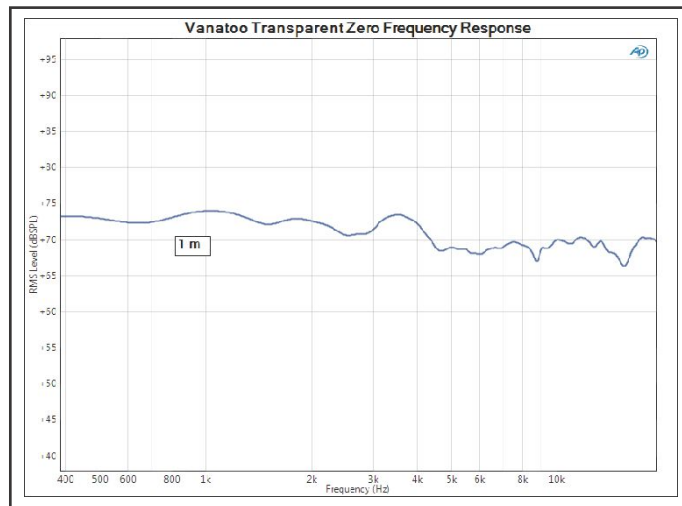


Figure 1: The on-axis frequency response of the Transparent Zero is reasonably flat, other than a 2 dB treble shelf and a notch in the tweeter response at about 8.5 kHz.

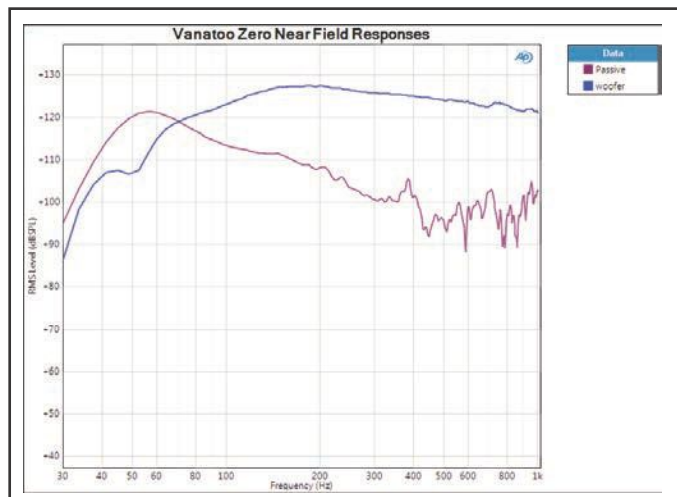


Figure 2: The near-field response of the Transparent Zero woofer and passive radiator indicate a smooth response and good damping.

remained a sore point for me, since neither computer had an optical connection, and the long USB cable that was needed was considerably more conspicuous and less flexible than the thin optical cable. Bluetooth is satisfactory, but comparatively still somewhat compromised because of the lossy compression.

My carping aside, the sound was what counted, and the Transparent Zeros delivered. I experimented with the setup, and ended up adjusting the stand height to 31" (about 80 cm), spacing them 7' (2.1 m) apart, and toeing them in so that I was on-axis at my listening position. Much like my first encounter with them, I was struck by the remarkable soundstage and imaging. Really solid and three dimensional, with the speakers almost completely disappearing. The tonal balance was slightly soft, which could be partially ameliorated with the treble control. But emphasize the word "slightly." It was quite subtle, and most noticeable when switching between the Transparent Zeros and my reference system. With the limiters set to Off, the dynamics were excellent, with the plucked string transients in my home recording of Lee Barber having a realistic snap, and the percussion and tenor sax on Clifford Jordan's "Live At Ethell's" (Mapleshade CD) having a delightful jump factor and transient edge. I had just done a live recording of renowned Chicago session artists LJ Slavin and Greg Hirte on vocals/harmonica and violin, respectively, and with their tonality fresh in my memory, I was impressed at how well it was reproduced by the Transparent Zeros.

As you might expect, deep bass wasn't there, but what bass was present seemed very clear and distortion-free, with an absence of lumpiness. The Transparent Zeros do not sound in any way lightweight, nor do they have the bass plumpness of mini monitors like the classic LS3/5A, but don't expect floor-shaking or pants-

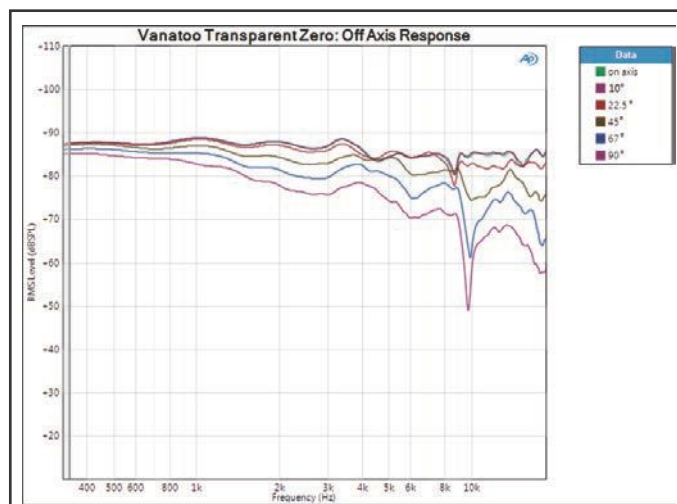


Figure 3: The Transparent Zero's horizontal off-axis response shows wide dispersion and a relatively smooth treble rolloff. Note that these curves are not normalized to the on-axis response and are unsmoothed.

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flapping unless you use a subwoofer. With some use of the bass control, you can warm things up, if that's your preference, but that comes with the inevitable trade-off of bass definition.

The subwoofer option was quite convenient. I ran this two different ways, with a powered inexpensive subwoofer (a rescued Klipsch KSW-12) and with my reference subwoofers, driven by a pair of Sunfire 500 W plate amps. The latter was obviously higher quality, though it seemed incongruous to use them as accessories for a pair of speakers that cost a fraction of what one driver alone in the subwoofers cost. Nonetheless, in both cases, integration was easy thanks to the sharp and well-defined bass cutoff of the Transparent Zeros DSP crossover. With the subwoofers in place, rocking commenced, with the music of my youth transporting my imagination back to the live venues.

Now lest this come off as an unqualified rave, allow me to qualify. When compared to the reference system, the Transparent Zeros fell short in clarity, transparency, and a difficult-to-describe

quality of refinement. And of course, they don't have the bottom 1-1/2 octaves, nor will they play as loudly. But to be fair, there's about a 20:1 ratio in price, and the Transparent Zeros certainly get the basics (tonality, dynamics, and imaging) right. If you want headbanging levels, they may not be the first choice. When I cranked up some Primus, the protection circuit shut things down before my ears gave out. But for reasonable volume out of acoustic sources in a medium-size room, the Transparent Zeros will easily reach a realistic volume without overt signs of stress.

Measurements

As usual, the measurement system that I used was an Audio Precision APx515, with the speakers driven via the optical output at 24 bits and a 48 kHz sample rate. The AP1701 transducer interface was used to supply phantom power to PCB Piezotronics 376A33 (0.5") and 376A31 (0.25") condenser microphones for the acoustic measurements.

Figure 1 shows the unsmoothed quasi-anechoic frequency response on axis at 1 m with all tone controls set to Flat. This was obtained by running a chirp signal, deriving the impulse response,

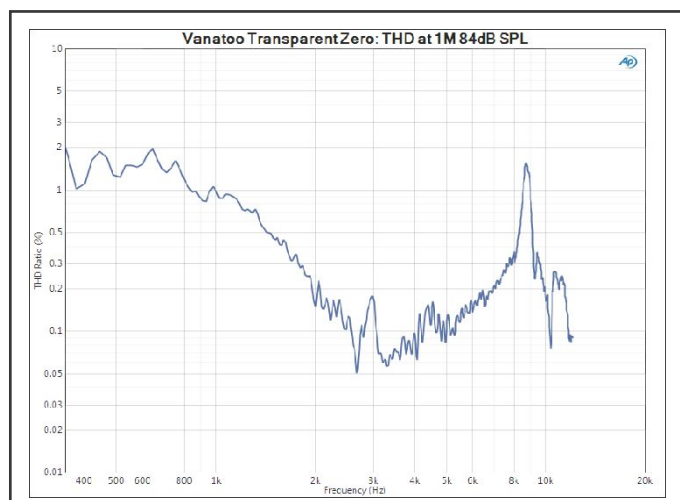


Figure 4: The distortion vs. frequency at 84 dB SPL and 1 m distance is moderately low.

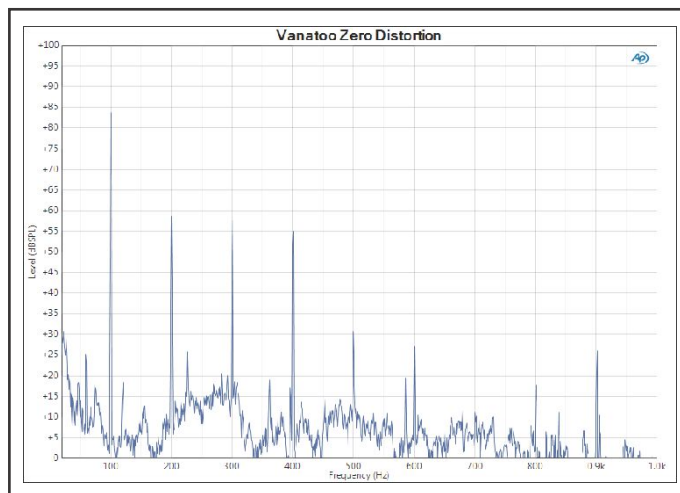


Figure 5: The spectrum of a 100 Hz tone shows that the distortion is dominated by low-order (second and third harmonic) components.

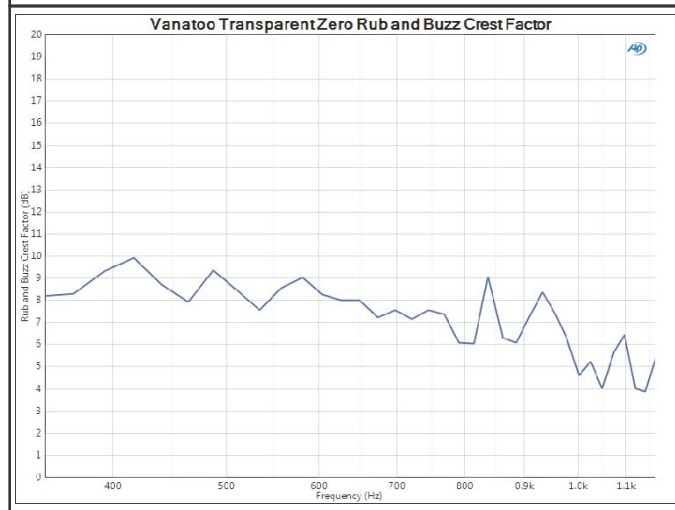
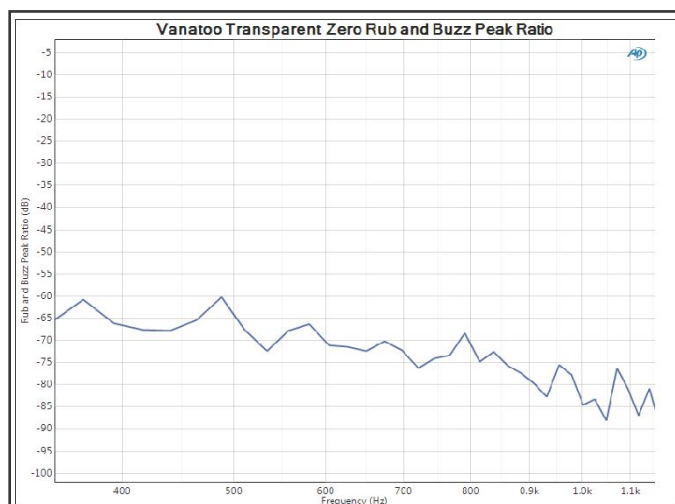


Figure 6: Rub and Buzz testing shows an absence of sonically annoying components.

then gating out the first reflection. As can be seen, the response is reasonably flat, other than a step-down shelf in the treble, which correlates with my impression of the sound being slightly soft. There's a notch at about 8.5 kHz, which persists regardless of microphone position, suggesting that it's an artifact of the tweeter rather than a diffraction notch. Other than these two observations, the frequency response is quite flat, which is unsurprising given the subjective impression of good tonal neutrality. **Figure 2** is a near-field measurement of both the woofer and the passive radiator, which confirms the specified 55 Hz woofer cutoff, and is free of ripple and shows good damping, again confirming the subjective observation of clean and well-defined bass.

The quasi-anechoic frequency response with varying horizontal angle is shown in **Figure 3**, and is very even, with treble rolling off smoothly with increasing angle, and no signs of the midrange

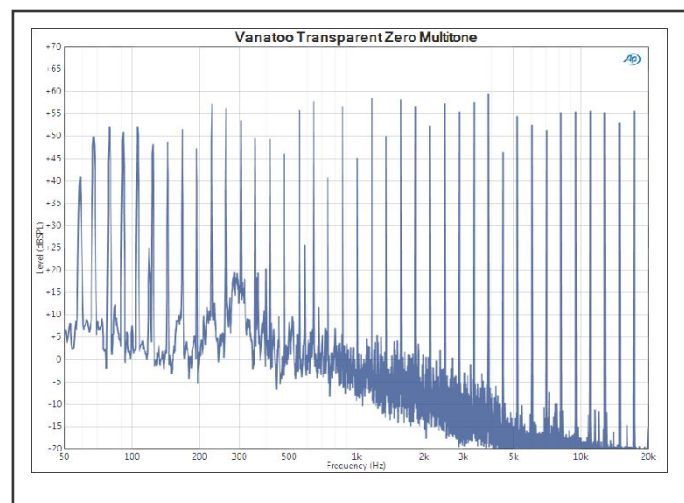


Figure 7: A 42-tone spectrum is relatively clean but does show some anharmonic noise in the lower midrange. This is a very extreme test!



Photo 4: The Transparent Zeros are a fully integrated system, offering built-in bi-amplification and digital signal processing from analog and digital inputs, including Bluetooth.

dip and flare seen in the off-axis response of systems with less capable crossover design. At extreme angles, an interference notch appears at about 10 kHz, which may also contribute to the slightly soft aspect of the in-room sound. In my experience, this sort of well-controlled polar behavior in a small speaker correlates with excellent imaging, which was certainly one of the Transparent Zeros' striking attributes.

Figure 4 shows total harmonic distortion (THD) vs. frequency at 84 dB SPL and 1 m distance. It's not exceptionally low, but it is dominated by second- and third-order components as indicated in the spectrum of a 100 Hz sine wave (see **Figure 5**). Note that in the THD graph there's a distortion peak at about 8.5 kHz, corresponding to the frequency response notch. This is one of those places where the low cost of the tweeter shows. There's also some small but noticeable anharmonic noise in the lower midrange.

Arguably, THD doesn't correlate well with listening tests, so I also ran Audio Precision's proprietary Run and Buzz tests (designed to highlight audibly annoying distortions) from Audio Precision, the results of which are shown in **Figure 6**. These plots show a reasonably low level of distortion with none of the peaks one sees with less-than-capable drivers.

Lately, I've also started running multitone tests on loudspeakers. **Figure 7** shows the results for the Transparent Zeros at 84 dB SPL and 1 m, on axis. The variation in tone height occurs because, unlike a standard frequency response, the measurement here includes room reflections. Interestingly, the anharmonic noise seen in the distortion spectrum of Figure 5 reappears here, and may be some of the reason I thought that the Transparent Zeros fell short on clarity compared to the much more expensive competition. As I collect more data like this from other speakers, we'll see if this measurement provides a way to objectively test for this perceived sonic quality.

Conclusion

This has been a rather long exposition on some relatively inexpensive speakers, but I think the engineering is enough to merit the high ratio of words to price. In the tradition of the killer small speakers of yore (e.g., BBC LS3/5A, NHT's Super Zeros, and Fulton FMI-80s), the Vanatoo Transparent Zeros offer users a way to get 80% of the performance of ultra-expensive systems at a fraction of their cost (see **Photo 4**).

When you consider that preamplification, power amplification, and signal processing are built in, the cost goes from "bargain" to "unbelievable." For under \$1,000, a music lover can assemble a complete system (source, amplification, speakers, stands, and subwoofers) that will provide years of satisfaction and outperform lots of the high-priced stuff sold on the basis of prestige rather than performance. Yes, I'm keeping these. 🎸

Resource

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Smart Speakers: Helping Improve User Experience

Audio Design Rules for Voice-Enabled Devices



Photo 1: Distance, reverberation, echo and noise impact a smart speaker's ability to hear.

Kevin Connor shares his insider knowledge with regard to Cirrus Logic's ability to provide audio and voice solutions to help OEMs create products that sound great and respond reliably to voice commands, regardless of backend service. These include ICs and software for mic capture, front-end processing and loudspeaker playback.

By
Kevin Connor

(Smart Home Applications,
Cirrus Logic)

Unless you have been living under a giant pair of floor-standing loudspeakers since the 1970s, you know that voice-controlled smart speakers have changed the worldwide consumer audio market. According to Global Market Insights, the smart speaker market will exceed \$13 billion (USD), with shipments of more than 100 million units by 2024. This year in the US alone, 35.6 million people will use a voice-activated device at least once a month—a 128.9% jump from last year, according to eMarketer.

The rapid success of Alexa and similar voice-

control services is actually the convergence of several trends and technologies, such as the proliferation of ubiquitous connectivity, search, and streaming audio, as well as breakthroughs in automated speech recognition (ASR) and natural language processing. New digital signal processing methods such as multi-microphone beamforming and talker tracking have made voice control feasible in noisy environments and at increasing distances—from headset to hands-free to “far field” or across the room.

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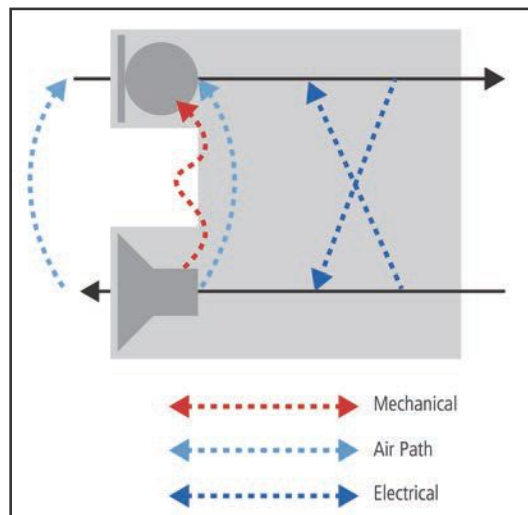
Key to improving voice interaction with voice-enabled devices is a focus on higher performance audio and acoustic technology. An ounce of prevention in the acoustic design is worth a pound of cure in the ASR backend. By reducing the influence of external noise in the environment surrounding the smart speaker, cloud-based voice services (e.g., Amazon Alexa) will be better able to hear and process commands, thus improving performance and user experience.

Turnkey solutions exist to perform “front-end signal processing” such as multi-mic beamforming, echo cancellation, and even low-power wake word detection (critical for battery-powered portable devices). The ultimate purpose of front-end processing is to improve the user experience, and critical to that experience is improving recognition accuracy of the ASR service, be it from Amazon, Google, or others. Vendors such as Cirrus Logic provide audio and voice solutions to help OEMs create products that sound great and respond reliably to voice commands, regardless of backend service. These include ICs and software for mic capture, front-end processing and loudspeaker playback.

What Affects Audio Performance?

Improving the quality of the voice signal sent to the cloud is not a question of simply adding more microphones. Three factors govern the input quality: the acoustic environment, the hardware design of the playing-and-listening device, and the digital signal processing applied to the microphone signals. The acoustical environment is a given, and largely beyond the control of the designer. The hardware design and signal processing code employed, however, are implementation choices where clever investment can pay dividends.

Figure 1: Potential coupling between loudspeaker and microphone



What Do Users Want to Do with Smart Loudspeakers?

If you’ve used a speech recognition system, you’re familiar with the challenges of using voice control in noisy environments or from a distance. These include:

- Getting the device’s attention quickly and reliably
- Waking the device in noisy environments
- Waking the device from across the room
- Interrupting the device when it is already talking or playing music
- Getting the device to understand your request and do the right thing

Smart speakers come with additional requirements:

- Fantastic sound—clear, balanced, and as loud as desired
- Good performance as player and listener in a variety of room locations
- Long usage time before re-charge, if portable or battery-powered
- Physical appeal and the best audio performance possible for the size
- Appropriate mix of voice and physical controls; audible and visual feedback
- Easy pairing or setup with other devices and cloud services
- Stereo and multi-speaker synchronization

The Acoustical Environment

A smart speaker mic signal is degraded by four main physical factors: noise, distance, reverberation, and echo (see **Photo 1**). These interact, and the ultimate result is that the microphone signal contains a mix of desired talker signal and undesired signals (distortion and additive noise) collectively called noise. A design goal is to improve speech recognition accuracy by reducing the undesired part and allowing the desired speech to pass undisturbed.

- Additive Noise: Sources can include voice-like noises such as office chatter and TVs, sharp noises (e.g., clattering dishes or dogs barking), and steady background noises such as air conditioning, road noise and even the quiet hiss of the electronics itself. Some noises come from a particular direction, like the radio in the corner, while others, like the hum of a refrigerator, are diffuse and permeate the room.
- Distance: Talker-to-device distance is critical as sound intensity diminishes with the square of the distance and since sound power radiates into the

room like an expanding sphere. Doubling the distance from 1 m to 2 m reduces the speech signal intensity by a factor of four, but the level of the diffuse room noise at the mic stays constant. Physics is unkind here. Moving from a hands-free distance of 1 m to a “far field” distance of 4 m in a living room is a reduction factor of 16, or a 12 dB SPL loss. In short, the signal-to-noise ratio (SNR) decreases very quickly as the talker moves farther from the mic.

- **Reverberation:** Sound waves are reflected by surfaces in the room like walls, floors, and furniture and arrive at the mic at slightly different times. The mic picks up a combination of the talker’s voice propagating along a direct path to the device and thousands of reflections from every surface in the room, collectively called reverberation. As talker-to-mic distance increases, the signal mix shifts from mostly direct voice plus a little reverberation, to a signal containing a good deal of reverberation and very little direct voice. This is known as the Direct-to-Reverberant energy ratio (DRR) and it decreases with increasing talker-to-mic distance. Just imagine ... a smart speaker hears the world like you hear a phone call to a person talking into a speakerphone from across the room! Anything we can do to improve this will help all voice recognition engines, whether local or cloud based.
- **Echo:** Echo is leakage of the loudspeaker output signal, the music or speech that the device is playing out into the room, back into the device’s own microphone and ultimately up to the speech recognition engine in the device and/or cloud. It occurs through the air via acoustic reflections mentioned earlier and is also conducted by mechanical vibrations through the device structure.

How do these acoustic factors impair accurate speech recognition? Recall our major use problems: “wake word detection” and “command recognition.” Respectively, getting the device’s attention from a distance, and in noisy environments, and “barge in” or getting attention when the device is already speaking or playing music.

Wake-Word Detection

Reliable wake performance is made difficult by noise, distance, and reverberation. Design cues such as LEDs and video screens may help—anything that motivates the talker to face and directly speak to the device and stand as close as possible is helpful. However, the design may require 360° pickup and we cannot control the talker distance. Therefore, we need a way to reduce noise and reverberation.

Beamforming is a spatial noise reduction method based on mic arrays that can effectively focus or point the mics toward the talker direction while diminishing signals arriving from all other directions, which we know are mostly noise or reverberation. This has the net effect of improving the SNR, far beyond what can be captured with a single omnidirectional mic. These techniques are collectively known as beamforming since an array of microphones is used to derive a single signal, which acts like a single magic microphone with a pickup pattern that is highly directional and steerable like a beam from a flashlight.

Beamforming is a term that covers a wide variety of techniques, including fixed and adaptive beams, various numbers of microphone elements (typically 2 to 8 for smart speakers), array geometries (typically points distributed on a circle or half-circle for 360° or 180° use) and processing sophistication. A related process to multi-mic beamforming is Direction

About the Author

Kevin Connor joined Cirrus Logic in 2017 in a technical marketing role where he builds prototypes and supports customers of Cirrus Logic’s Smart Home solutions, including voice processors, embedded software and audio tools. He has previously worked as a researcher and DSP developer at BlueJeans Network, Cisco Systems, and Nortel Networks specializing in voice-over-IP, conferencing, networking, voice quality measurement, and monitoring. Kevin is a member of the Audio Engineering Society (AES) and enjoys recording, electronic music, and restoring vintage synthesizers.

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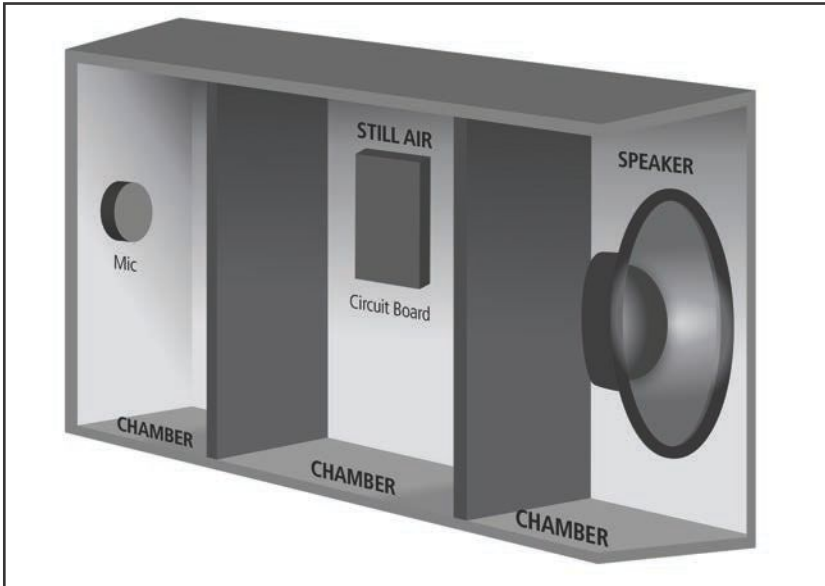


Figure 2: A three-chamber arrangement is used to minimize speaker and mic coupling.

of Arrival (DoA) estimation. The direction of the device’s “beam of attention” is relayed to the user by means of LED lights, and provides valuable feedback to the user to move closer, raise his voice or repeat the command.

There is another, more familiar type of noise reduction processing in which steady noises such as computer fans are estimated and subtracted, after the fact. However, ASR engines do not like aggressive noise reduction (NR).

Although moderate amounts sound like an improvement to human listeners, speech recognition accuracy is generally better if this type of ambient noise reduction is turned off. In fact, Google Home guidelines recommend disabling mic path NR. Since ASR engines are trained on noisy speech, ambient noise is basically invisible to the ASR and the artifacts of aggressive NR are a net hindrance to machine recognition.

Additional wake detection design issues are latency, and in the case of portable products, battery life. It is preferable to perform wake-word detection locally on the device as this gives a much faster response to the user, rather than endure a needless up-and-down interaction with the cloud, which could raise privacy concerns. However, this requires intelligence on the device, which has a power-consumption cost for portable devices. A solution is to use dedicated silicon for wake detection (and beamforming) that can function in a low-power mode, avoiding the need to run the main processor at full pace in the long breaks between control interactions.

Barge-In Performance

To interrupt a device when it is playing music, for example, all the previous difficulty of waking from a distance in a noisy environment is now compounded by echo. Echo refers to the problem of loudspeaker output signal leaking into the device’s own microphone. This leakage is acoustic as the sound waves bounce off the walls in the room and are reflected back into the microphone, and also are transmitted directly to the mic as the housing of the device vibrates with the output signal (see **Figure 1**). This is an old problem with speakerphones, but the intensity is worse here because music playback can be much louder. Have you ever tried to interrupt a person who is shouting?

Echo is mitigated by a DSP technique called Acoustic Echo Cancellation (AEC). Without AEC, there is little hope for the device to understand commands, as the user’s voice signal is impossibly corrupted with the outgoing music signal. The AEC task grows more complex and expensive as the number of loudspeakers increases (e.g., mono to stereo to surround) and the number of microphones

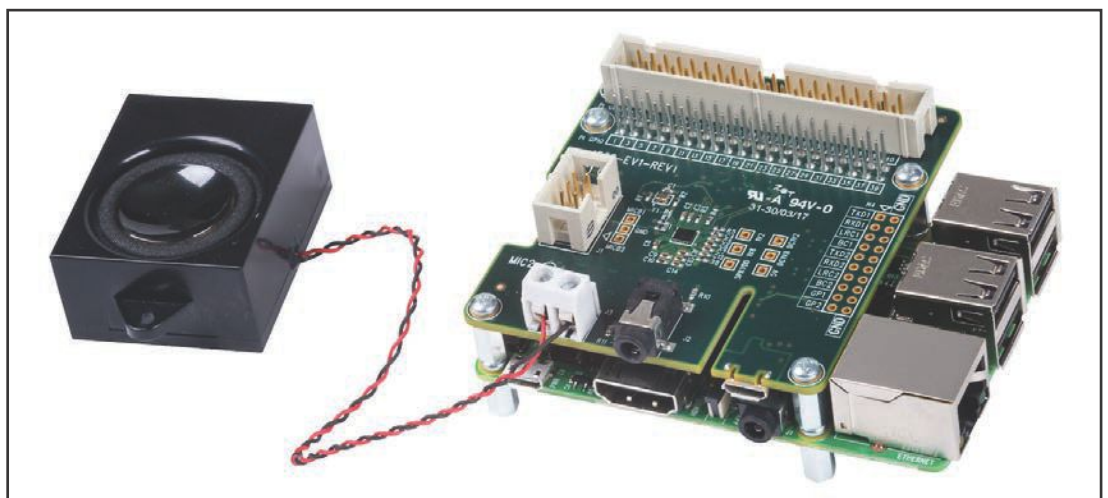


Photo 2: This is Cirrus Logic’s Voice Capture Development Kit for Alexa Voice Service.

increases. It becomes more difficult as the volume of the music is increased as there is simply more echo to start with. Louder music can cause various nonlinear distortions in the loudspeaker device itself, which creates an unsolvable math problem in the AEC.

The effectiveness of AEC, and hence barge-in performance, is improved by minimizing mechanical coupling between loudspeaker transducers and microphones in the hardware design (see **Figure 2**). Improving isolation and choosing high-quality transducers and proper mechanical design prevents buzzing and nonlinear output distortion. Modern amplifier chips such as those from Cirrus Logic include smart techniques for limiting distortion in smaller transducers by limiting the excursion of the diaphragm.

Sounds Good, Good Listener

There is a direct correlation with how “good” a system sounds (audio quality) to its ability to respond. The ability to interrupt a voice-enabled device during playback depends on the AEC capabilities, as well as the playback system. If loud playback has low distortion, it will not only sound good to you, but to the AEC, which must cancel out the playback so it does not interfere with the device hearing you call out the wake word.

Therefore, how well a digital device works and plays music will also be a direct function of how much distortion comes from the speakers. Cheaper speakers tend to result in poorer quality audio and higher distortion playback when the volume is turned up. Technically, the playback system can never be better than its digital-to-analog converters (DACs). For example, Cirrus Logic includes a smart codec with integrated high-fidelity DAC in its Voice Capture Development Kit for Alexa Voice Service devices (see **Photo 2**). This, along with embedded voice control algorithms, will enable OEMs to create products that both sound great and respond reliably to voice commands.

Lower Power Designs Will Drive New Portable Devices

As the use of voice-response smart speakers grows, new audio and acoustic technology will improve performance and pave the way for new, innovative applications. Savvy buyers will look beyond the number of microphones. They will look for high-quality audio and portability with long battery life as determining factors when purchasing next-generation voice-enabled smart speaker devices. Better acoustic technology, such as AEC, noise reduction, and low-distortion playback, will improve barge-in performance so that voice services can hear and understand you better. Another key benefit to the advanced audio technology will be higher quality audio that will also improve the overall listening experience.

New development kits with advanced audio functions are making it easier for smart speaker OEMs to develop this next generation of innovative designs and devices. Device OEMs won't have to be acoustic experts in audio design to gain the improved functionality, performance, and features that come from improved audio capabilities. OEMs will also benefit from new lower power high-performance semiconductor design. For example, Cirrus Logic solutions use as much as 80% less power than others. This will free OEMs to design innovative new portable applications for smart speakers that provide amazing new user experiences. 

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The 6922 Project

By
Robert Nance Dee

(United States)



Photo 1: I built two cases for the 6922 hybrid. It is shown here in its “presentation” case.

This is the final design in my series on buffered preamps. “Willow,” which ran in *audioXpress* (December 2011), described the first buffered preamp to use this typology. I updated the design, which was detailed in “Willow Revisited” (*audioXpress*, February 2016), and I’ve saved the best for last in this tube hybrid design.

Unlike many of my previous designs, the 6922 project (see **Photo 1**) is a complete project that readers can build from boards I’ll be supplying. I’ve also paid special attention to the bill of materials (BOM). It is an area I have been lax on in the past often because by the time articles are published, designs have changed, sometimes significantly, which can make the BOMs misleading.

Also, many of those designs were hands-on projects and I expected readers to come up with many of their own solutions to parts and modifications. Again, this is not the case here, this is a polished, complete design with relevant BOMs and accurate highly revised and tweaked circuits and diagrams. You won’t see any MLCC caps in the tube power supply or line stage.

The Design

I have gone back to the HA-5002 series for the output buffer stage. It’s just a good chip and I’ve refined the design to remove any previous problems

integrating it into the design it may have presented. The 8-pin dip is obsolete (you may be able to find them online) but the 968-HA9P5002 (Mouser) is still available (a dip adapter board is required to make a DIP 8).

While the DS1882 might seem dated to some of you, I suggest you build the 6922 and listen to it. I chose it for its qualities—it works exceedingly well, adds no coloration, uses no op-amps, and is easily soldered onto the board being a relatively large surface-mount design (SMD) package. Again, each part has been carefully chosen so please put any biases aside and listen to the total package.

I have listened to many tubes and built many tube amps over the years. The tube I chose for this design is the 6922. The 6SN7 is an excellent tube but I’ve yet to find a tube with the soundstage, depth, and sweetness in a single tube that matches the 6922. The specific tube I chose after testing quite few from new-old stock (NOS) to reissues is the reissued Genalex Gold Lion but please have a good

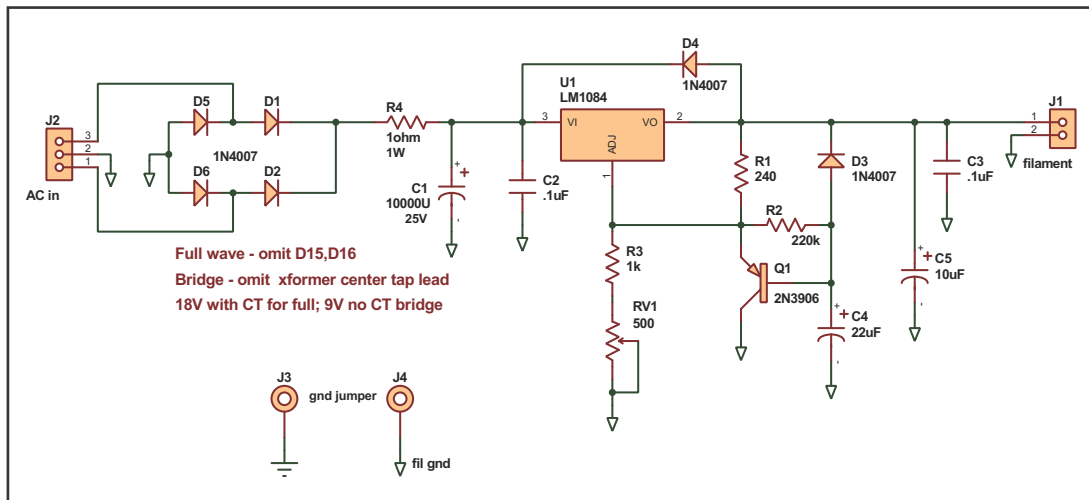


Figure 1: This schematic shows the filament ground scheme and regulating circuit.

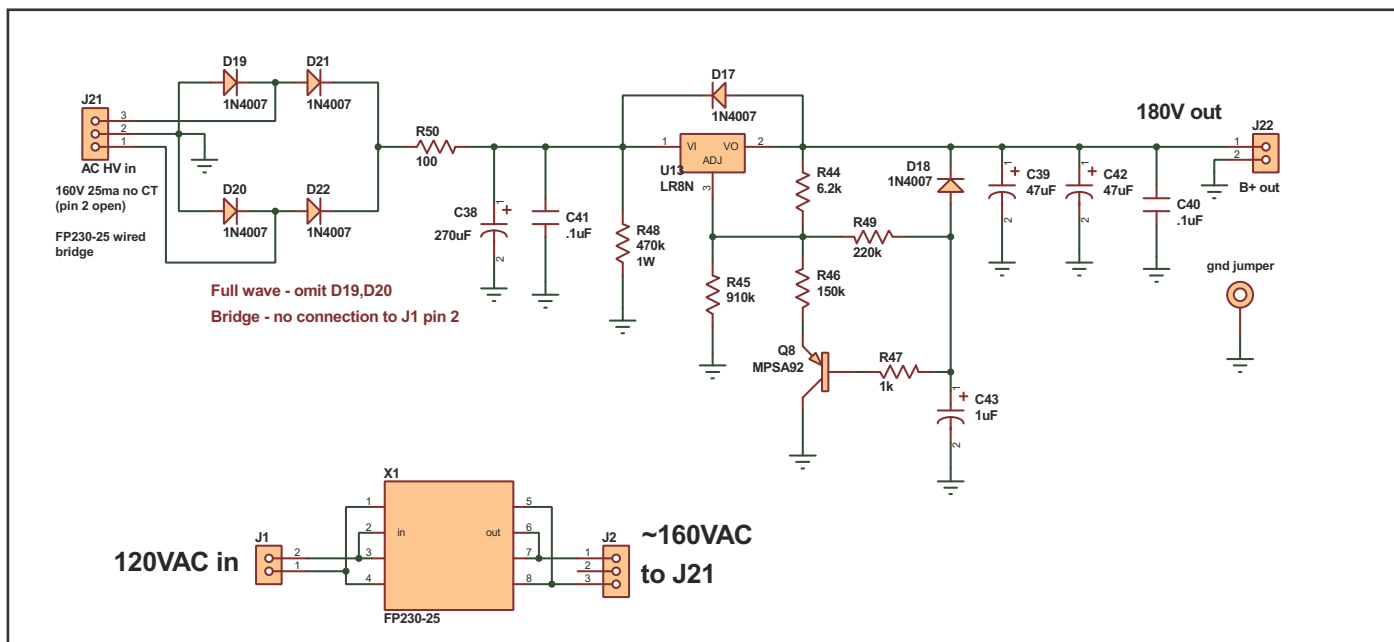


Figure 2: The high voltage supply uses the HV LR8N regulator.

time rolling your own. One last thing, the 6922 has the reputation for being a noisy tube, let's see if we can remedy that!

I view every circuit that supports a tube as integral to its sound and life. SRPP is an understandably popular option. Its low noise and a relatively constant plate current make it appealing, but I love single-ended tube circuits! I started this design at the filament because I think noise from this area is often underestimated.

Filaments have no mechanical attachment to other tube parts (directly heated being the exception) and because of this I completely isolated the filament supply from other supplies and only made one ground connection directly to the high voltage (HV) ground, not to the amp board itself.

Figure 1 shows the filament ground scheme and regulating circuit. Three terminal regulators are an excellent choice for tube power. Old school tube rectifiers simply cannot compare with respect to



Figure 3: This graph shows filament and the high voltage rise gradually over about 10 seconds on turn on.



Figure 4: Although the high voltage supply contains 180 V, it emits very low noise (70 mV).

their low noise, versatility, and cost. When I have presented all the supplies for this amp I think you'll agree.

The Topology

Filament regulation is paramount to tube life, and I tried several typologies before settling on this simple but highly effective timing circuit. Employing a current limiting lab power supply set to the filament current specs of the 6922, I timed how long it took to get to operating voltage then used this data to set the regulator's timing to approximately reach operating voltage—simple but effective.

The 2N3906 PnP transistor slowly cuts off as the voltage of C4 rises. You can change the timing capacitor C4 to increase the timing if you like as not all 6922 run at the same current—although timing is not all that critical. Checking the maximum current showed a little more than 400 mA. With a tube rated ~ 375 mA. RV1 enables fine adjustment of

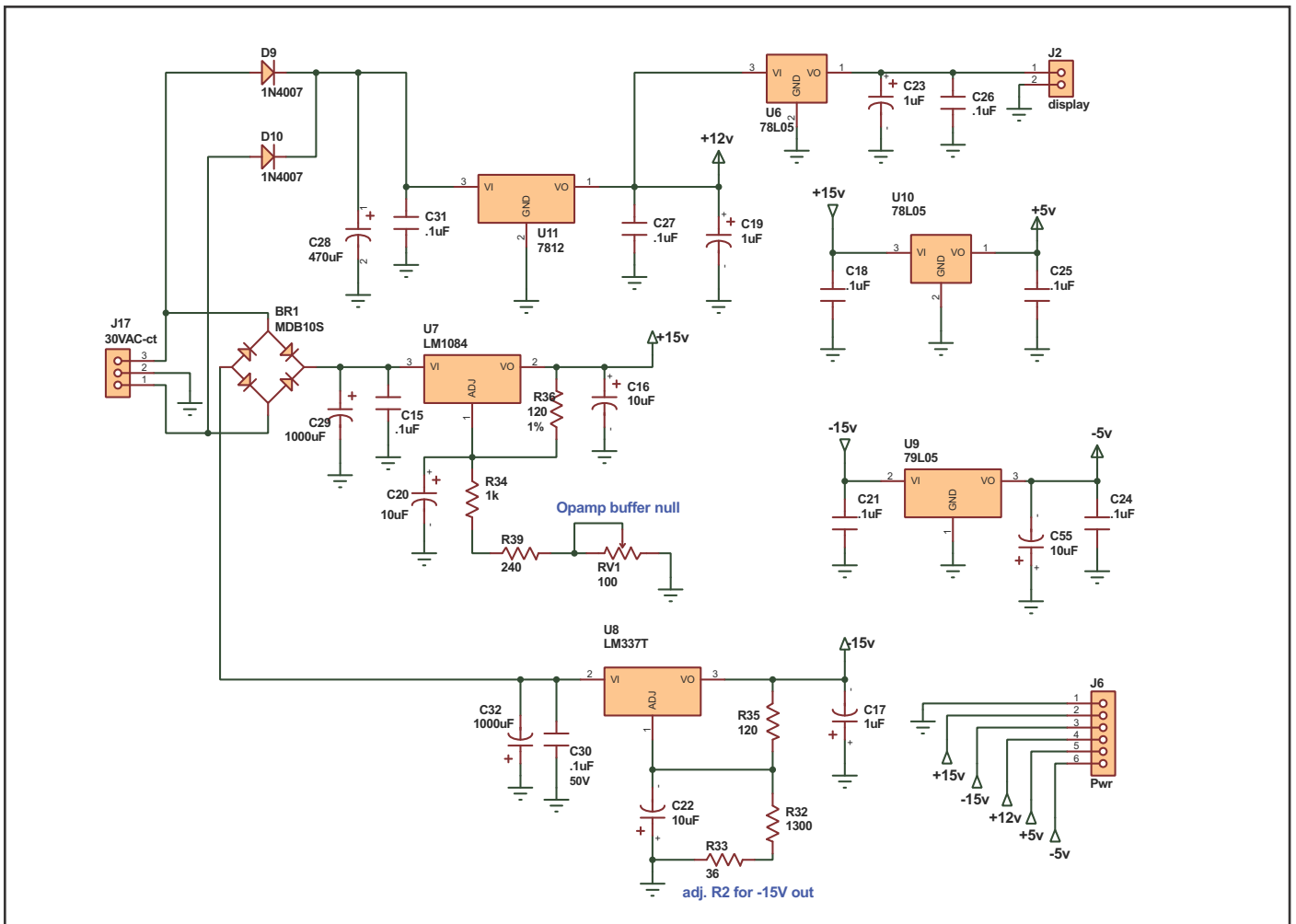


Figure 5: The schematic for the buffer supply also shows the digital pot, display, relays, and more.

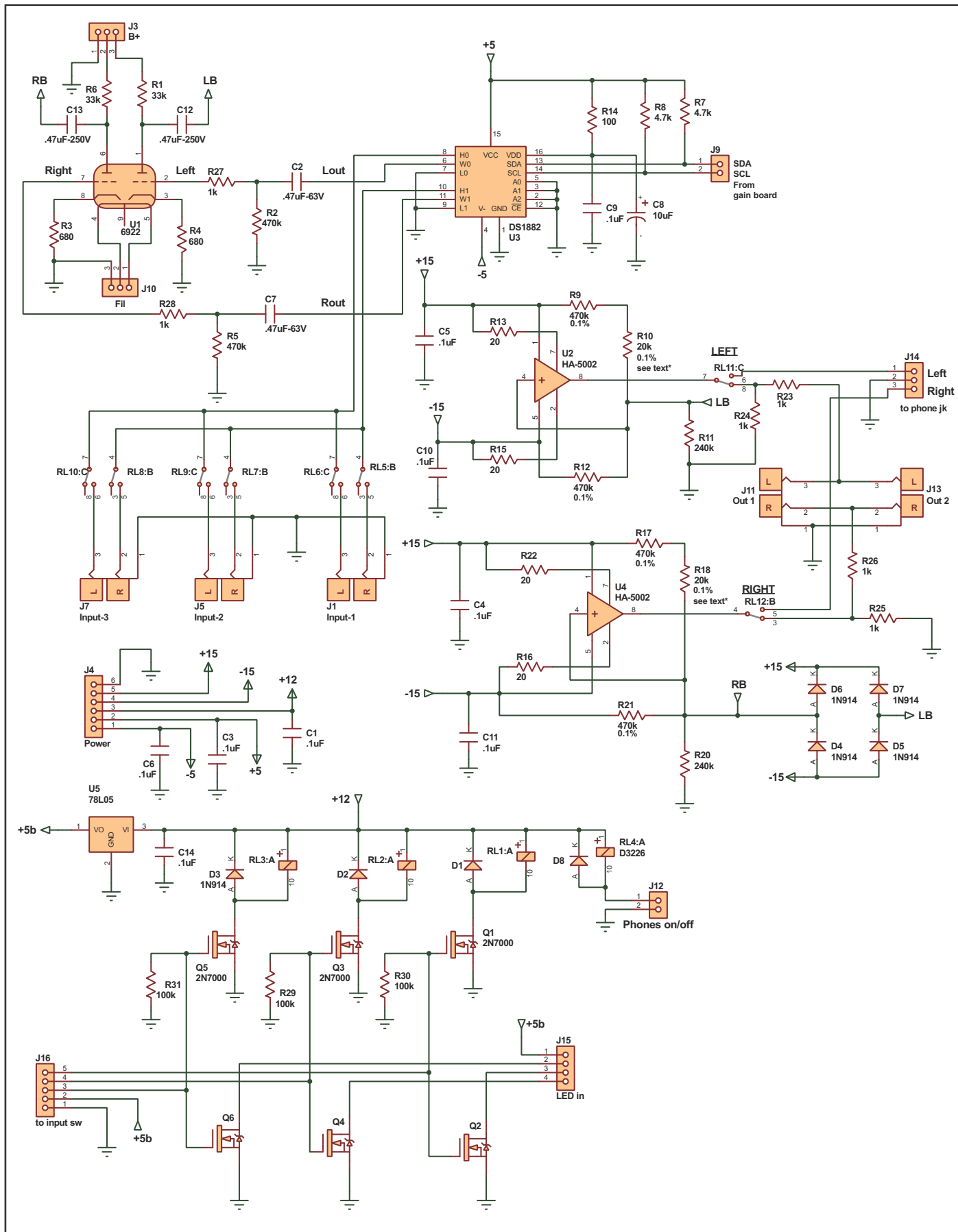


Figure 6: This schematic details the tube amp configuration.

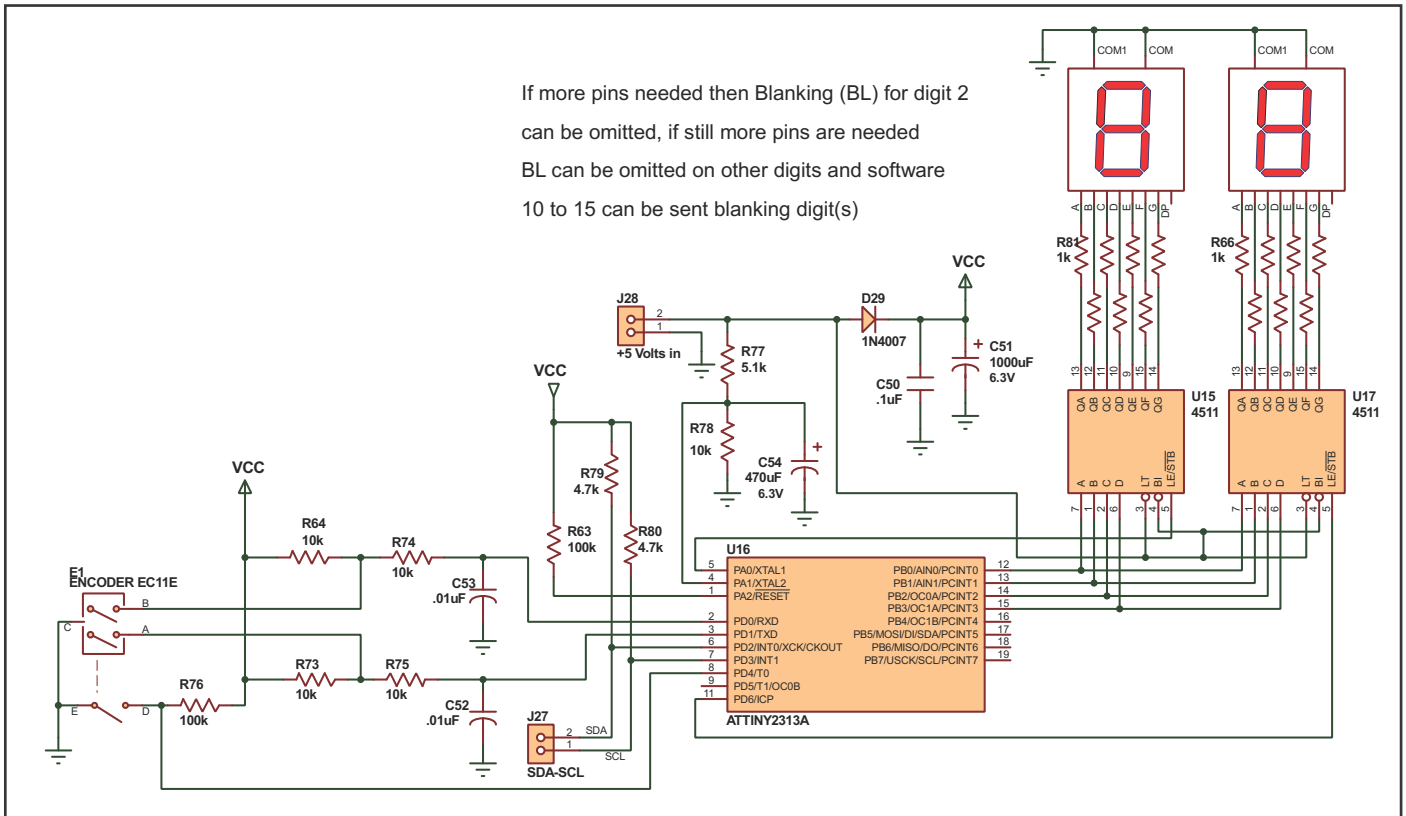


Figure 7: The display section was designed so the microcontroller shuts off after sending data to the DS18B22 so no stray noises have any chance of feeding into the amp.



Photo 2: The 6922's more traditional case was much easier to build with the benefit of having all the boards separated by a good distance.

About the Author

Robert Nance Dee is a retired electronics engineer. He received his BS from the State University of NY, where he was nominated for the Chancellor's Award for Student Excellence. He has worked on large frame military computers and has several medical instrument patents. He enjoys electronics, mechanics, clock and watch making and precision machining. He and his wife Nancy live in the Western Catskill Mountains of NY where he is presently restoring a massive E. Howard Tower Clock in the Delhi, NY village square. He can be reached at robert@dsgnspec.com

the filament voltage, I usually run 6.2 V for a spec 6.3 V filament like the 6922. I used an inexpensive transformer for this isolated supply and isolated the supplies of the HV and miscellaneous voltages. I used three transformers in all, with a total cost of about \$30. High-cost transformers are not required. If you can't hear it, you don't need it!

The HV supply shown in **Figure 2** uses the same approach and for this I used the HV LR8N regulator with a similar timing circuit as the filament to slowly raise the HV in step with the filament voltage. I did not use or need relays to switch the HV on after the filament reached optimum voltage. Relays make an abrupt change in voltage from zero to maximum and that voltage surge reflects in the output with a loud pop.

Figure 3 shows that the filament and high voltage rise gradually over about 10 seconds on turn on. **Figure 4** shows the very low noise (70 mV) of the 180 V high voltage supply. The LR8N regulator is both current limited and short circuit protected, but this is a moot point. The output capacitors C39 and C42 never reflect a low impedance back to the regulator over the several seconds it takes to reach full voltage, R50 also further protects the regulator. The HV regulator

has one more important function, it keeps the voltage constant for different plate currents. Even at full plate load, the HV never varies. Also, for this design I used two identical HV supplies—one for each tube segment. Figure 2 also shows the wiring of the HV transformer. (See the Bill of Materials for transformer ordering info. The BOM is available in the Supplementary Material section of the *audioXpress* website, www.audioXpress.com.)

Figure 5 shows the supply for the buffers, digital pot, display, relays, and so forth. I did not use special regulators here for several reasons. These regulators are very robust, the tube is very quiet, and the regulation is well within any component specs.

If you build this amp correctly, using good grounding techniques, you will not have problems. For grounding I have supplied specific ground pins on the HV and filament supplies. I found that the quietest topology is with the filament supply ground tied to just the HV ground pin with the two HV pins tied together and one ground wire from the HV ground point to the amp board. You can experiment if you like. But after building two amps with different

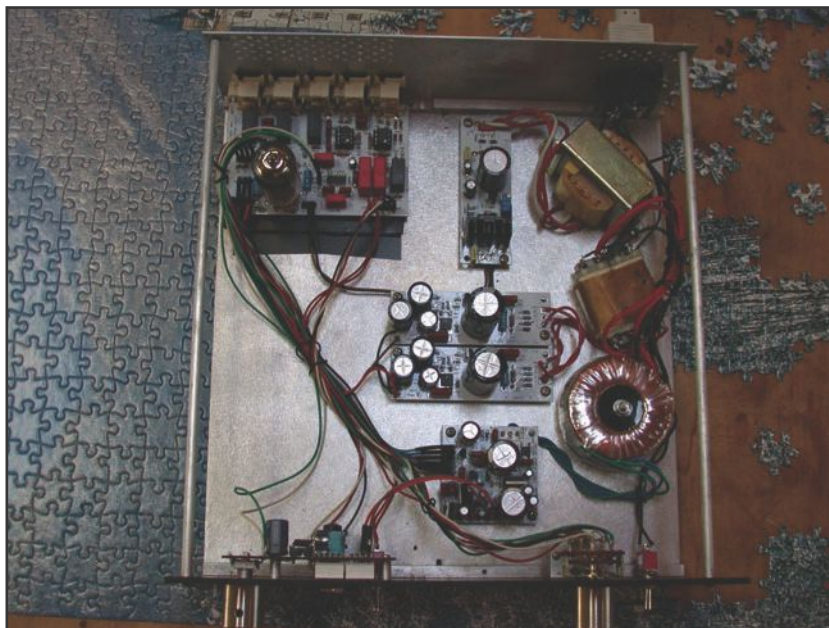


Photo 3: There is plenty of room for the transformers to have their own area far away from any signal lines.

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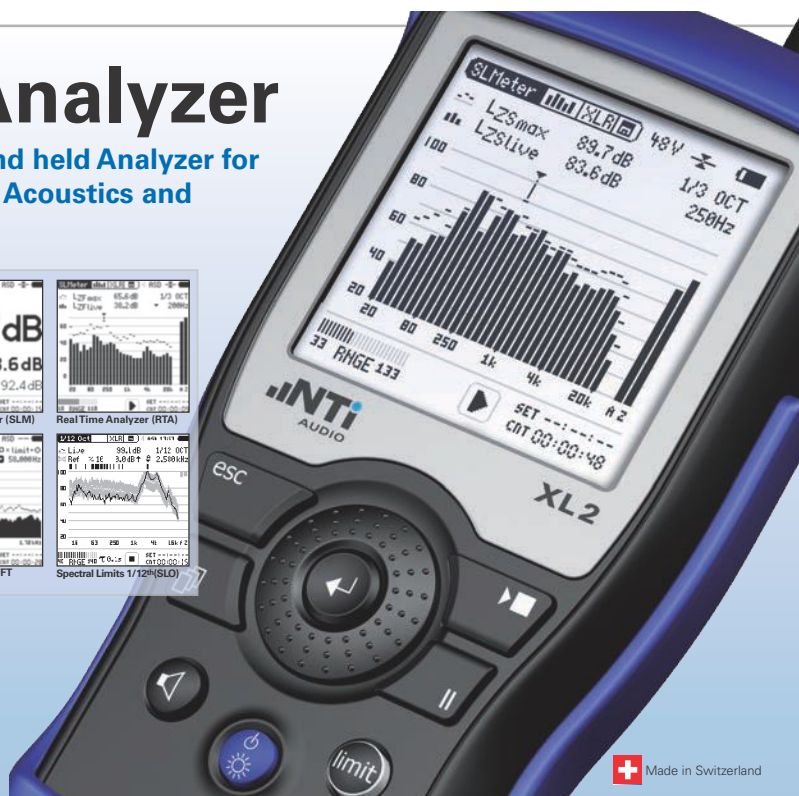
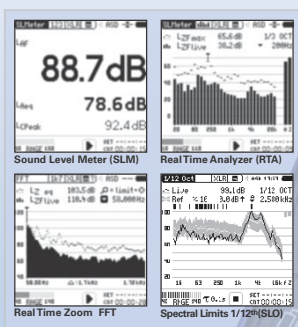
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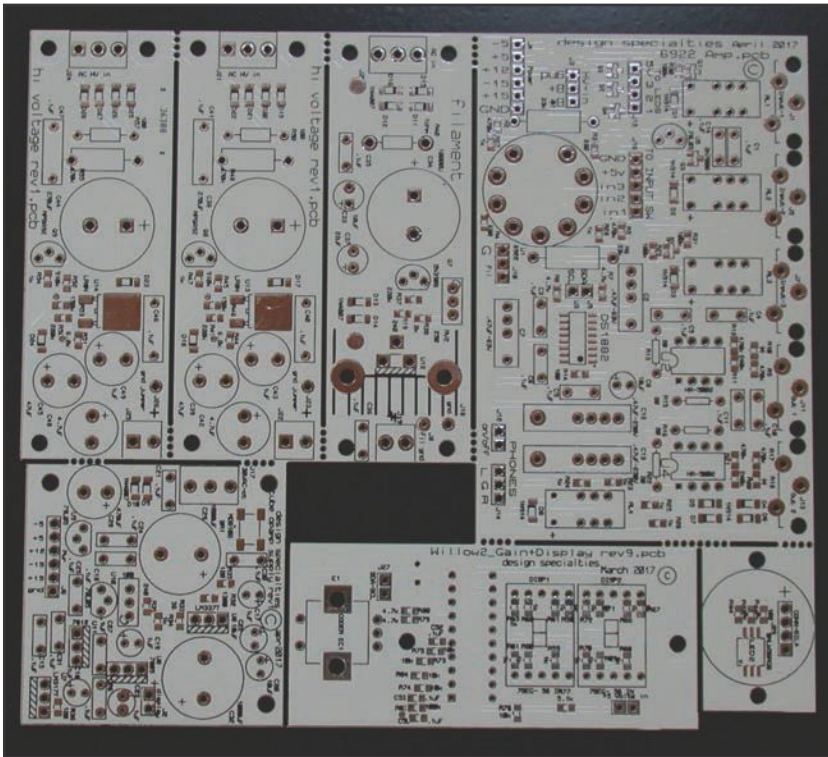


Photo 4: The circuit board includes all the individual boards in a easy-to-break-apart package.

chassis layouts, I have found both work best with the above configuration. The tube has no hums, buzzes, or noises with several headphones from electrostatics to low impedance AKGs. Even my high sound pressure level (SPL) phones are quiet. You will not have noise issues with this tube in this amplifier.

RV1, shown in Figure 5, is used to set buffer null. To set the null ground the inputs (only right and left of the working input need be selected), set the audio level to 90 (mute) and measure the voltage

on pin 4 of U2, U4 shown in **Figure 6**, which details the schematic for the tube amp. The goal is to have the two levels close to equal—they do not have to be zero or exact. Change resistor R10 or R12 making both levels close to the same value. Once this is done, null the inputs with RV1. Again the inputs do not have to be zero nor do they have to be exactly the same level. Although this is possible, it's sufficient if you get the levels to within, say, 20 mV of each other. Then, adjust RV1 so that one level is 10 mA and the other -10 mA.

The Display

Figure 7 details the display section—the microcontroller shuts off after sending data to the DS1882 so no stray noises have any chance of feeding into the amp. The readout is in decibels with mute reading as “90” and max reading as “0.” There are 64 volume positions in all 90 then 62 to 0 dB. The display stays on without noise as the 4511's latch through the output of the microcontroller before it goes to sleep. The DS1882 has non-volatile memory backup, on amp power up the microcontroller requests, over the I²C bus, the last pot setting then goes into sleep mode shutting off the microcontroller clock. With the amp on, microcontroller wake up is triggered by a rotary encoder level change. The microcontroller updates the display 4511 drivers, which lock the two LED digits and updates the DS1882. Then, it returns to sleep. The microcontroller clock is internal running at 8 MHz divided by 8 or 1 MHz, low enough for the I²C routines but well above audible.

The Completed Project

I built two cases for the 6922, one is a “presentation” case (see Photo 1), which sits on my reading table and has a relatively small footprint. It's a difficult case to build with several hours of machining and CNC work. The more traditional case (see **Photo 2**) is much easier to build with the benefit of having all the boards separated by a good distance. There is plenty of room for the transformers to have their own area far away from any signal lines (see **Photo 3**). The circuit board includes all the individual boards in an easy to break apart package (see **Photo 4**).

The schematic shown in Figure 6 details the tube and adjoining circuitry. You can play with different plate resistors, but if you use the values I've specified you'll have an exceptional amplifier. Single-ended tubes tied to buffers like the 5002 are hard to beat. First, the simplicity of one tube means you can roll different tubes at half to one quarter the cost of SRPP. The benefits of SRPP

Project Files

To download the Bill of Materials, visit <http://audioXpress.com/page/audioXpress-Supplementary-Material.html>

Resources

R. N. Dee, “Willow,” *audioXpress*, December 2011, reprinted online January 2015 (www.audioXpress.com/article/The-Willow-Pre-amp-A-high-slew-rate-JFET-amplifier).

—“Willow Revisited: A Design Celebrating the Enthusiasm and the Creativity of the Builder,” *audioXpress*, February 2016, or online at: www.audioXpress.com/article/willow-revisited-a-design-celebrating-the-enthusiasm-and-the-creativity-of-the-builder

(e.g., constant plate voltage) have been resolved in this amp along with any noise issues. SRPP just can't compete with this tube buffer combo for output impedance matching either. **Figure 8** shows a 32 Ω headphone at 10 kHz.

Overall Impressions

Building the 6922 hybrid was easier than many of my other designs and while the input output jacks may not appeal to the high-end community, they make the shortest possible path and have never given me any trouble or noise problems in all the designs I've incorporated them in.

Build this amp and listen for yourself. It just shines. It's low cost, very refined, and smooth as silk with an exceptional sound stage and depth. It's my go-to amp now. As much as I love the Willow series, this one wins me over every time I turn it on. With the detail I've given the power stages, your precious tubes should last a very long time. And, the distortion and noise is less than 0.1%. After 50 years I'm still trying to figure out how much significance all the specs printed in the back of brochures really have! 

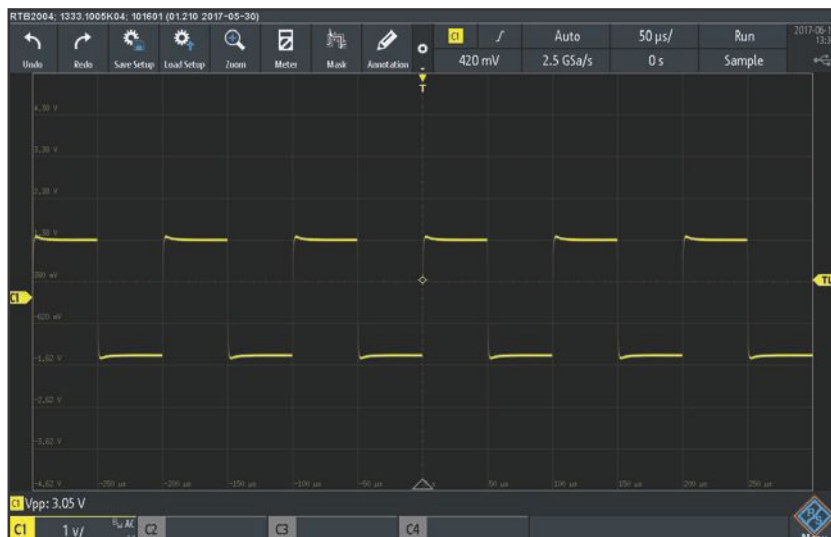


Figure 8: Single-ended tubes tied to buffers like the 5002 are hard to beat. Here are my 32 Ω headphone at 10 kHz.

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Repairing Switching Mode Power Supplies

There are a lot of books and articles out there about how to design a switching supply, but not many on fixing them. As switching supplies become ubiquitous in electronic devices today, it becomes that much more important to understand how they work and just as importantly how they fail.

By
Scott Dorsey
(United States)

The entire point of the switcher is that it rectifies the AC power line to DC, then chops the DC with a variable-duty-cycle oscillator at very high frequency so that a tiny step down transformer can be used. Transformers at high frequencies don't need large cores or many windings for a lot of power, so they can be made tiny and at little cost. The oscillator duty cycle can be adjusted with feedback so that the regulation can be done without losing any power in the process. Thus, you can get good regulation and good efficiency at the same time.

This article will focus on line-operated flyback supplies. Other topology converters exist and are popular when line isolation isn't needed, but when you look at what is going on between the AC input and the DC rails on a piece of electronic equipment today, this is the basic topology used because it gives good efficiency and line isolation.

How Switchers Work

Figure 1 shows a sample switching supply design (courtesy of Texas Instruments). This comes from the UC2842 datasheet and uses the common UC2842 PWM controller chip. (The datasheet can be found in the Supplementary Materials section of the *audioXpress* website, see Project Files for the link.)

Note that this design, as is typical, has complete isolation between the primary and secondary sides of the circuit. You can draw a line in your head through the transformer core and through the optocoupler and break the circuit up into two electrically isolated halves. This is an important point and you will see

this in almost all supplies of any size since isolation from the power line is a primary safety concern.

AC power comes off the line, and is rectified through the bridge rectifier, D_{BRIDGE} . The output charges up a big filter capacitor on the primary side C_{IN} , which provides a filtered (but hardly ripple-free) DC voltage to the primary of the transformer, N_P , as well as voltage to start the pulse width modulation (PWM) chip through resistor R_{START} .

R_{START} only supplies a small amount of current to start the device, so once the first pulse makes it through the field-effect transistor (FET), current from a third winding on the transformer is used to provide power to run the oscillator. This is what N_A and D_{BIAS} are all about. You might not see that third winding, you might just see all the running power being drawn through a higher power dropping resistor in place of R_{START} . But, using the third winding improves efficiency a lot.

When the PWM oscillator is running, it sends constant pulses from the output pin. This turns on the big switching FET, Q_{SW} , which pulses the current going through the transformer. As this happens, current is induced in the transformer secondary, rectified and filtered by D_{OUT} and C_{OUT} , and current flows from the output.

Because the PWM oscillator is so fast, the transformer and the filter capacitor on the secondary side can be very small. Although that 2200 μF cap may seem large, if the oscillator is running at 60 kHz, it's a thousand times as effective as the same value off the 60 Hz line.

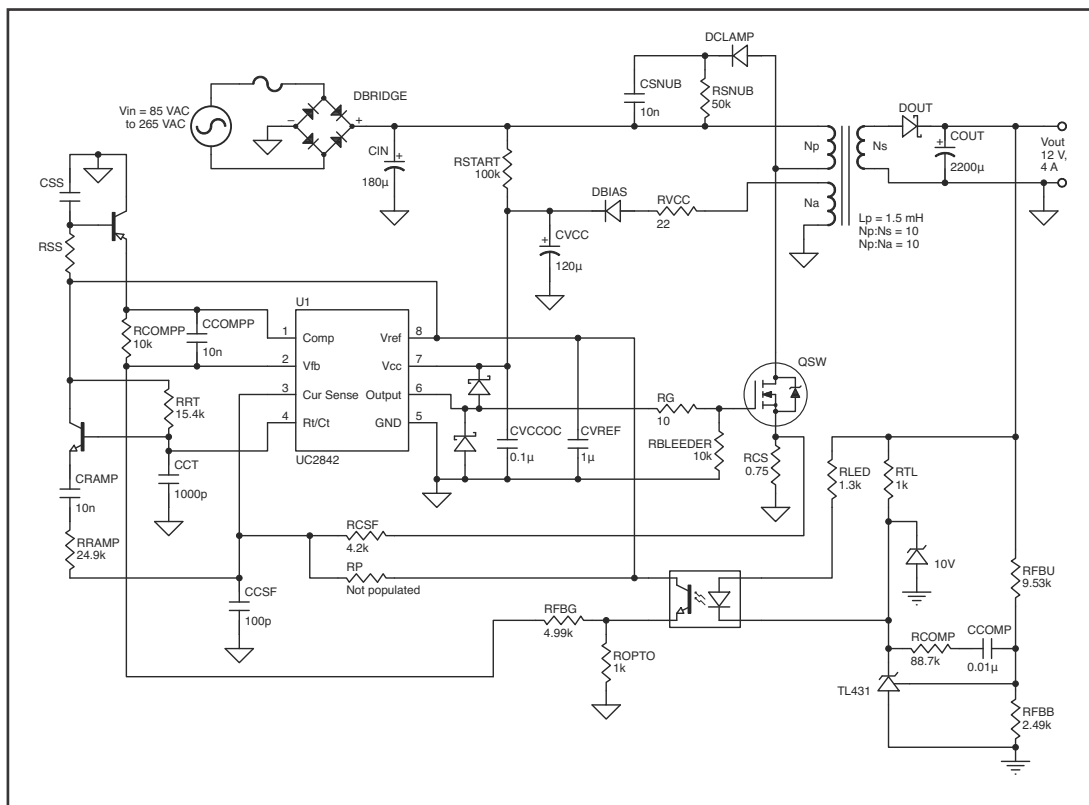


Figure 1: This sample switching supply design comes from the UC2842 datasheet and uses the common UC2842 PWM controller chip. (Original schematic courtesy of Texas Instruments)

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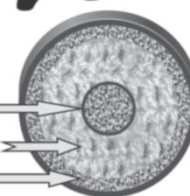
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Regulating the Power Supply

So, how does the regulation work? All that other stuff on the secondary causes the LED in the optoisolator to come on when the output voltage exceeds 12 V. The UC2842 provides a small amount of regulated 5 V (made with an internal linear regulator) and that voltage at V_{REF} is used to power the output stage of the optoisolator. It provides a varying voltage to the V_{FB} input to provide feedback to the UC2842 that the voltage is correct and to back off the duty cycle of the output waveform a little bit.

The optoisolator doesn't have to be very linear for the UC2842 duty cycle to be kept right on the edge so the output voltage is always perfect.

The I_{SENSE} input is measuring the voltage drop across R_{CS} , which is to say it's measuring the current draw through that switching FET. The UC2842 is designed so that if it exceeds 1 V, it shuts the PWM circuit down. So this is a current protection circuit.

Now, normally we would see a resistor and capacitor, R_{RT} and C_{CT} , connected to the RT/CT pin and providing a time constant for the PWM oscillator. In this case we also are amplifying the PWM ramp signal off of that with a transistor and applying it to the I_{SENSE} input through C_{RAMP} and I_{RAMP} so that the circuit is stable for very long duty cycles. This is called "slope compensation" and the trick for doing it is explained briefly in TI datasheet for the UC2842 chip, but not in the datasheets for any other manufacturers.

And what about that other transistor, with C_{SS} and R_{SS} ? That's a little circuit to narrow the pulse width when the device is first turned on and slow down the startup slightly so there is less shock to components.

Now, you will see other variants on this basic

circuit. You'll see an additional transformer winding being used to provide feedback, instead of the optoisolator. You'll see the PWM IC being driven directly off the AC line instead of with that NA winding. You'll see multiple secondaries and crowbar circuits. But this is the basic design that you will see inside any switcher and so your job is to figure out exactly what changes from this basic design exist in your circuit.

How to Determine What You Have

The bad news is that most of the time you won't have any documentation for the switcher. The good news is that most of the time the switcher will be very close to identical to the sample circuit on the PWM chip datasheet (see **Figure 2**). Not always, and not for higher end supplies, but much of the time getting the chip datasheet will tell you 90% of what is going on with the circuit.

The vast majority of better quality Chinese-made supplies seem to use the C2842/UC2843/UC3842/UC3843 series of PWM controllers. These are made by a dozen different companies including Fairchild Semiconductor, ON Semiconductor, TI, and STMicroelectronics, and each of those companies has a slightly different datasheet with slightly different sample circuits. So if you don't see the circuit you have encountered on the datasheet, get another datasheet from another manufacturer and likely you will (see **Figure 3**).

The Fairchild KA7552 shows up in a number of devices (see **Photo 1**). This was a Samsung design, now sold by Fairchild since they took over the Samsung facilities and product line. It is vaguely similar to the UC2842 although with a different pinout.

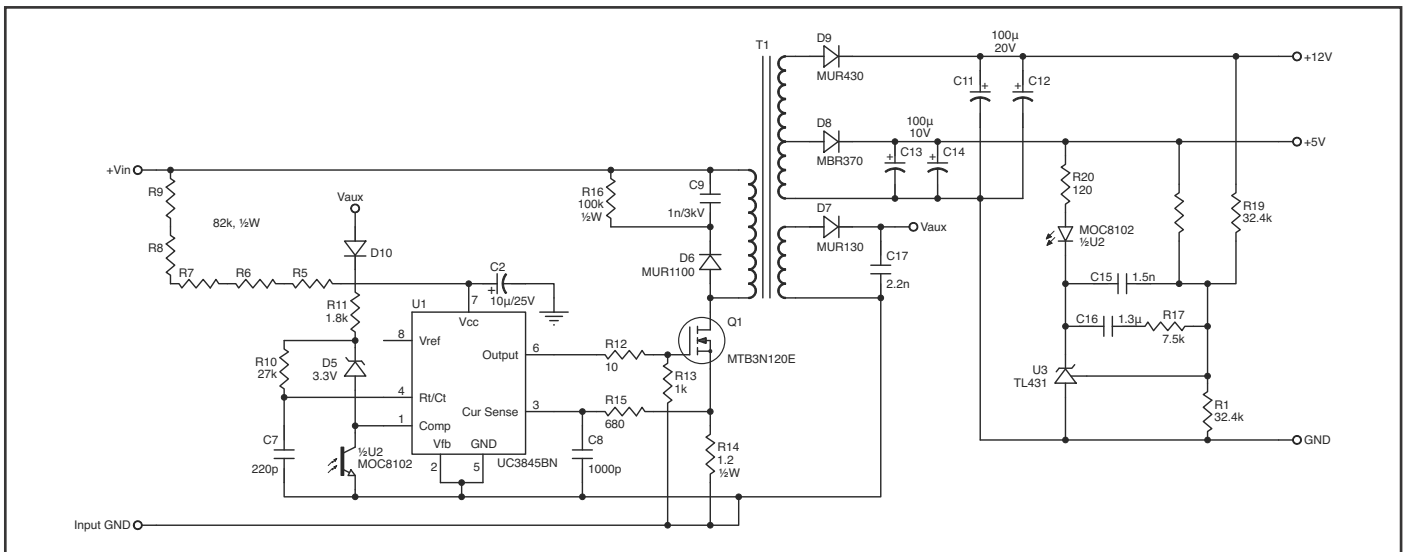


Figure 2: This schematics shows a typical small switching supply using a 3845 PWM IC. Notice the Vaux output is referenced to the input ground. The optoisolator, U2, is drawn in two individual halves. U3 provides a reference with which to compare the 5 V line.

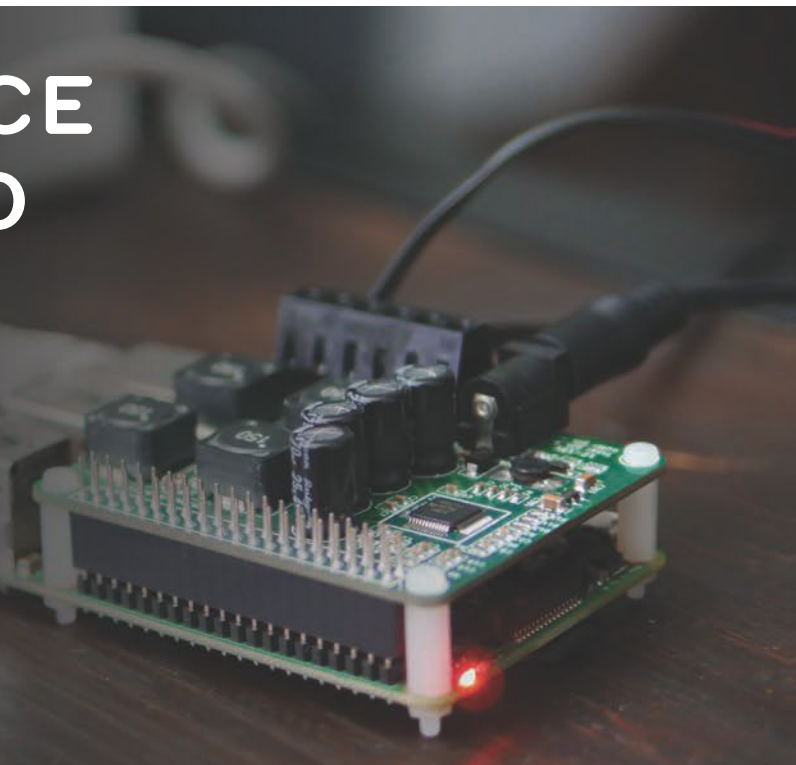


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About the Author

Scott Dorsey has a degree in electrical engineering, during the pursuit of which he worked in the broadcast and recording industries. After several years working at a major studio, he took a job with a defense contractor. This left him time to do live concert recording for acoustical music and to design and build audio devices for personal use and on contract to several audio manufacturers and importers. Scott is a regular contributor to several audio magazines. He has been publishing equipment reviews and DIY projects since the mid-1980s. He is probably best known in the general audio community for his retrofit electronics designs in inexpensive Oktava, AKG, and Feilo microphones.

Sometimes you will see the TL594 PWM controller from ON Semiconductor. Again, there are a couple other vendors for this so you should check for multiple datasheets.

One very popular IC that you will find in lower powered single output devices is the TOP242 series of chips made by Power Integrations. These are integrated PWM oscillators on the same substrate with a big power FET. Add a transformer, a couple rectifiers, and an optoisolator and you have a complete switching supply in a box. Of course, they fail frequently but are fairly easy to diagnose. However, there are dozens of power and package options on these chips so you can't always keep them all on hand. A similar but less popular device is the MC33374.

A lot of less expensive Chinese-made products will use the AP3021 control IC, and this chip is

manufactured and sold under dozens of different names by dozens of different companies in China. The documentation on it is poor, but if you ever encounter a cryptic-looking PWM controller where pin #6 is not being used, it is likely to be an AP3021 or a copy. The English language datasheets for this product are skimpy at best but once you have some idea of the pinout and how it works you should be able to figure out what is going on.

Encountering the Unexpected

Not every supply is a single switching supply in a box. Sometimes you will encounter systems with multiple switchers in the same box providing multiple output voltages, each regulated. It's more common to find multiple voltages off one transformer with a single output voltage used for the control loop but some applications require good regulation with highly varying loads.

Sometimes there is a second "always on" power supply that provides a standby voltage used to run the processor that controls the main power. This is very common for things like video monitors and computers. Often this supply is on a small daughterboard since it needs good electrical isolation from the rest of the electronics but doesn't need to produce a lot of power.

If you see a lot of small discrete transistors all over the place, a good guess is that they are involved with automatic shutdown systems, to shut down in case of

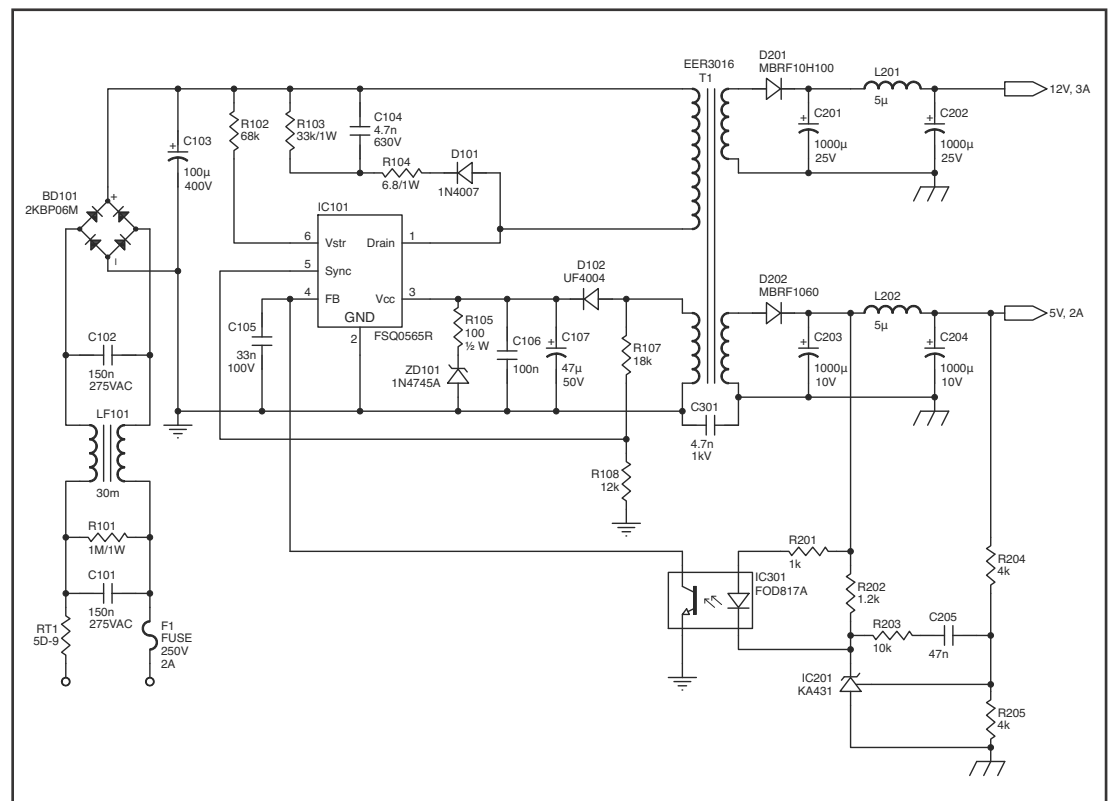


Figure 3: Here is another variation of a small switching supply design. This switcher uses regulation on the 5 V rail and the 12 V rail is regulated only in that it tracks the 5 V rail. A fourth winding powers the PWM chip.

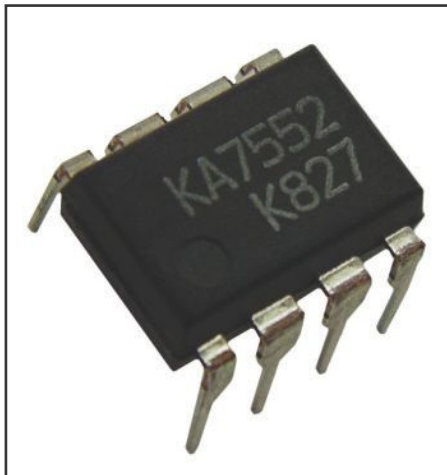


Photo 1: The Fairchild KA7552 has been used in several devices.

high or low voltages or currents at one or more place. Troubleshooting these circuits without a manual can be a real nightmare since it can be difficult to figure out at what voltages individual parts trigger.

Every once in a while for audio or other low noise applications you will see linear series regulators for a little additional smoothing, located after the switching supply. Since these can run hot, they are a common source of trouble but fairly easy to diagnose since you can see power coming into and out of them.

Fixing the Problem

If you have documentation on the power supply, half the work is done for you. If not, you know the basic block diagram and you can work out the individual parts within each block by hand. Getting the datasheet for the PWM chip will tell you a huge amount since most PWM circuits and sometimes entire supplies are just copied from the manufacturers' datasheets. Often the PWM chip will have multiple sources. For example, you can get the common 2842 PWM controller from at least four different vendors. All have different datasheets and if your circuit isn't on one, it might be on another.

If the power supply comes on but immediately crowbars, the first thing to do is check or replace all the filter capacitors on the secondary side of the transformer. Other things can cause this like a leaky rectifier on the secondary or a bad resistor in a current sense circuit, but they are far less common.

Sometimes the caps will be just leaky enough that the supply will start up with no load but won't run with any load on it. Your inclination is to blame the load for pulling too much current, but it's not always the load. When in doubt, change the caps and then take the diagnosis from there.

Many power supplies use a "kickstart capacitor" to supply current to start them up. This isn't shown in the example given above, but it is a fairly common configuration. If the power supply was working, was shut off, but then would not restart at all, replace the kickstart capacitor. If there is no documentation, this is likely to be a 25 V to 50 V electrolytic of very small value (1 μ f or 2 μ f), located near the PWM chip.

The high voltage capacitor (sometimes two capacitors) on the primary supply, which filters the line directly seldom is seen to be failed in the US. However, in Europe where the line voltage is twice as much and where the same multi-input power supplies are used, those capacitors are frequently found to be bad. European supplies whose behavior changes with the load should first have these checked.

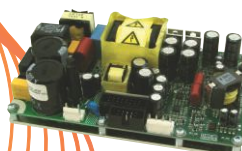
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overview

The NCxxxMP amplifier module incorporates a low power standby power supply (meets 2013 ERP Lot 6 0.5W requirements), a highly efficient switch mode power supply and a high-performance Class D amplifier in one compact and easily applicable power brick.



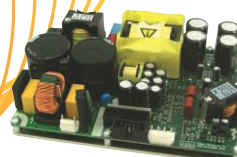
NC122MP



NC250MP



NC252MP



NC500MP



NC502MP

Add-on Module NC100HF
The NC100HF is a dedicated tweeter-amplifier which fits the NCxxxMP series.



Highlights

- High efficiency
- Universal mains operation
- Flat, fully load-independent frequency response
- Low output impedance
- Very low, frequency independent THD
- Very low noise

Features

- One or two channel amplifier
- 5W standby SMPS
- Advanced over current protection
- External controlled operation
- Auto-switching (115/230V)
- Low weight
- Compact

Applications

- Monitor loudspeakers for recording and mastering studios
- Audiophile power amplifiers for professional and consumer use.
- Public address systems
- Active loudspeakers

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Capacitors located near or under heatsinks tend to bake out very quickly and are common sources of failure. In fact, because the vast majority of failures that you encounter will be capacitor-related, having an equivalent series resistance (ESR) tester to make quick tests in-circuit is very handy. However, I am often inclined just to replace all electrolytics from doubtful manufacturers even if they test well, just because I want a longer lifetime from the supply than the likely design lifetime was.

If the problem isn't the capacitor, a very common failure is the power transistor or FET (see Q_{SW} in Figure 1). Usually these can be easily located by large holes in the board where the FET used to be, by all three of the pins of the FET having continuity between them, or by obvious diode or resistor failures in the circuit near the FET. If the FET is not "wiped" (meaning all three pins have continuity and beep on a continuity tester), it may be worthwhile to test it out of circuit.

If a FET is "wiped," however, whatever drives the gate of that FET has likely been destroyed as a consequence of the failure. That is often the PWM chip and it's good to have common PWM chips available in the spares bin.

A good rule is that if the switching transistor or FET has failed, you should replace the protection diode on the base or gate of the transistor. Even if it checks well, it may not be. The damping diode D_{CLAMP} is also one to check. FETs do fail for no apparent reason but more often they are driven to fail by overvoltage (from bad clamping diodes) or overcurrent (from bad and leaky capacitors) or high temperatures (from bad designers).

If these simple things don't fix your problem, it's time to start actually doing real diagnosis. Get out the meter and start looking at the pins of the PWM chip. Do you see a reasonable input voltage on V_{CC} ? Do you see a 5 V reference voltage from V_{REF} ? Do you see less than a volt at I_{SENSE} or more? Is the oscillator oscillating at all? Start making sure that the inputs to the PWM chip are good and then that the outputs to the PWM chip are good. If you have a waveform on the output pin but you don't have any output, start looking at the switching FET or transistor, the damping

diode around it, and so forth. If the oscillator isn't oscillating, what is it missing?

The exact values will vary depending on the PWM chip being used, but the recommended operating conditions table on the PWM chip datasheet will tell you about what they should be.

Capacitor Rules

Rule 1: Most switching supply failures are due to bad electrolytic capacitors. Even FET failures are often long-term consequences of an initial capacitor problem.

Rule 2: Nobody ever went wrong replacing cheaply made consumer electrolytic capacitors with higher grade 105C industrial ones. It might not fix the immediate problem, but it will likely improve the long-term reliability of the supply. So don't spend a lot of time trying to decide if a capacitor is bad, just replace it. Your time is worth more than an electrolytic.

Rule 3: Buy capacitors from a legitimate supplier like Digi-Key, Newark/element14, Allied/RS, Mouser, and so forth. There are a lot of counterfeit capacitors out on the market, capacitors that didn't come from the manufacturer on the can.

Rule 4: Electrolytic capacitors fail from age and poor engineering margins, but when other capacitor types have failed, it's because something else has caused them to fail.

Rule 5: Tantalum capacitors are actually electrolytics. The chemistry is a little different than that of aluminum electrolytic caps, but the long-term reliability and temperature-related issues are the same. Note that the more common "dry slug" tantalums (those epoxy-dipped types) tend to fail into shorts and this can make them easier to identify when failed. Unfortunately it also means that a failure can result in major collateral damage.

Peroration

Don't be afraid to work on equipment with integral switching supplies. It can take a long time to get the hang of how they work and the more common failure modes, but once you do, they are usually not difficult to fix.

If you want to learn how to design switching supplies (and you should, because that is also a useful skill), permit me to recommend the "Linear Technology Application Note 25: Switching Regulators for Poets," written 30 years ago by the great Jim Williams. Back then, switching supplies were fancy new things that designers were just getting a handle on, and available ICs were much more limited and crude, so Williams' description had to be detailed. It is a fine document that is available in many places on the web. 

Project Files

To download the Texas Instruments UC2842 datasheet, visit <http://audioxpress.com/page/audioXpress-Supplementary-Material.html>

Resource

J. Williams, "Linear Technology Application Note 25: Switching Regulators for Poets," September 1987.

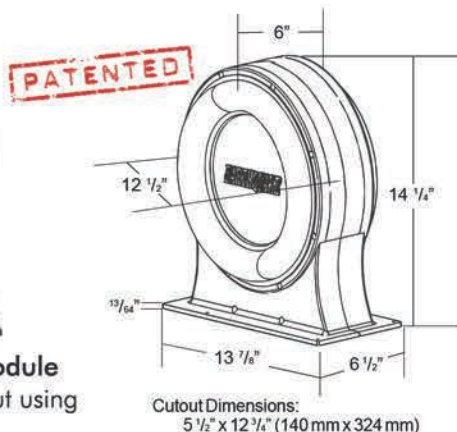
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Layout and Grounding of Hollow-State Circuits

In this article, Richard Honeycutt discusses the importance of a device's wiring and component layout. He also offers a few suggestions as to how to improve the layout.

By
Richard Honeycutt
(United States)

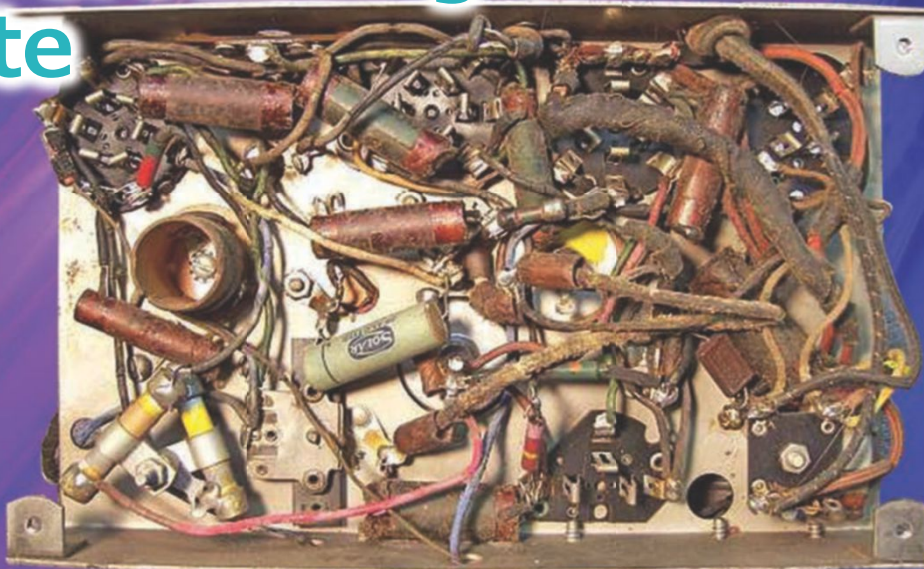


Photo 1: Although the result looks messy, the underchassis wiring of this antique Atwater Kent radio does minimize the length of wires.

The performance of all audio equipment depends significantly upon the layout of the components on the chassis and the routing of the wiring. Proper grounding

is also essential to keep noise low and prevent parasitic oscillations. The lower the level of signals in the system and the higher the AC mains currents, the more care is needed in layout and grounding.

Keeping the Wires Short

From the earliest days of electronic design, engineers understood that unnecessarily long wires could pick up electrically or electromagnetically induced noise and hum. Thus, the first rule of good layout is to keep the wires as short as possible.

Photo 1 shows the underchassis wiring of an old Atwater Kent radio. You can see that the wiring from component to component could not have been any shorter or more direct. However, you can also see that this approach, which works well electrically, is not aesthetically attractive—not that aesthetics matter in this instance.

Photo 2 shows the underchassis of a Philco radio manufactured about a decade later than the Atwater Kent shown in Photo 1. The Philco has been restored by replacing the old waxed-paper capacitors with newer Sprague Orange Drop film caps. The direct point-to-point wiring has been retained.

By the 1950s, TVs were being mass-produced, and they were all built using direct point-to-point wiring, as illustrated in **Photo 3**. Once again, this method

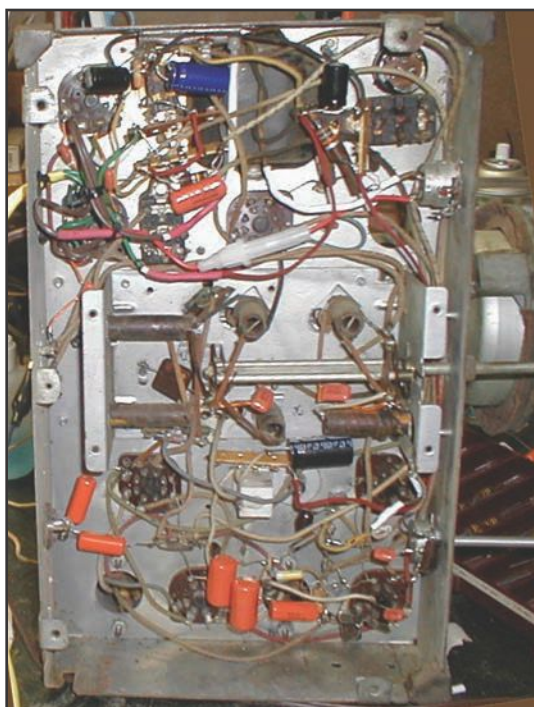


Photo 2: Direct point-to-point wiring was retained during the restoration of this old Philco radio.

of wiring results in minimum-length leads between components.

In the 1960s and 1970s, point-to-point wiring was supplemented—or in many cases replaced—by circuit boards. This approach made for a much neater appearance, although as you can see from the Ampeg SVT power amplifier's underchassis shown in **Photo 4**, the price of neatness is that some of the wires are longer than they would have been in an all-point-to-point-wired chassis. In this case, the wires that were routed in fairly straight lines with pretty sharp corners are the ones for which the engineers decided the lengths were not critical. The "minimum-length" rule is most important for leads carrying low-level signals, especially if the signal has strong high-frequency content.

Another example of a hollow-state audio design incorporating both circuit boards and point-to-point wiring is the underchassis of the PA amplifier shown in **Photo 5**. Note that the long wires are mostly DC power (B+) leads. Also, notice the twisted green wires. These are filament leads for the tubes. They carry AC, and the current in one wire is out of phase with the

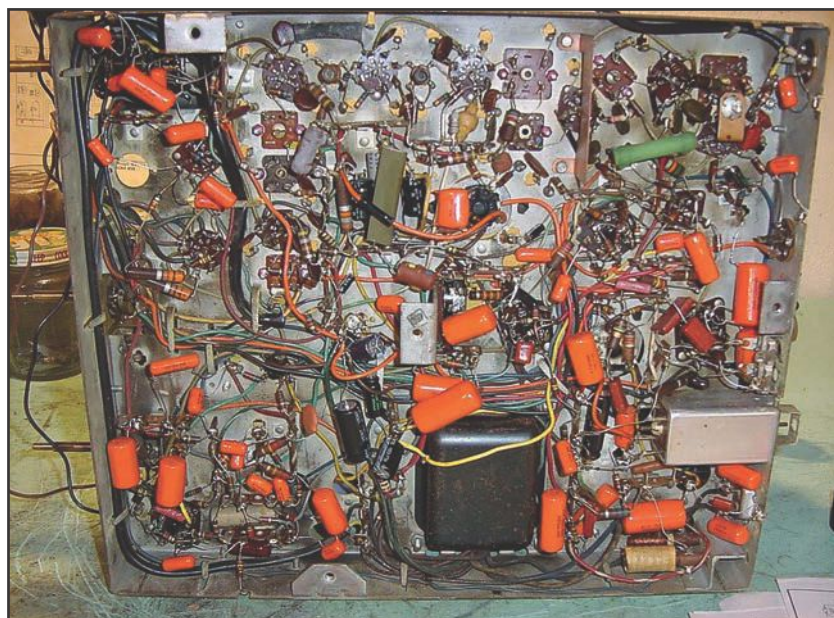


Photo 3: This 1950s-vintage TV has been restored, with the original capacitors replaced by Orange Drop film caps.

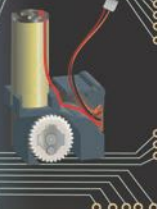
current in the other wire. Twisting them keeps them close together, so that the electric and magnetic hum fields they produce cancel out at points an inch or more from the wires. Twisting wires that conduct AC—such as filament leads—is always a good practice.

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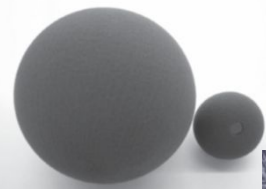
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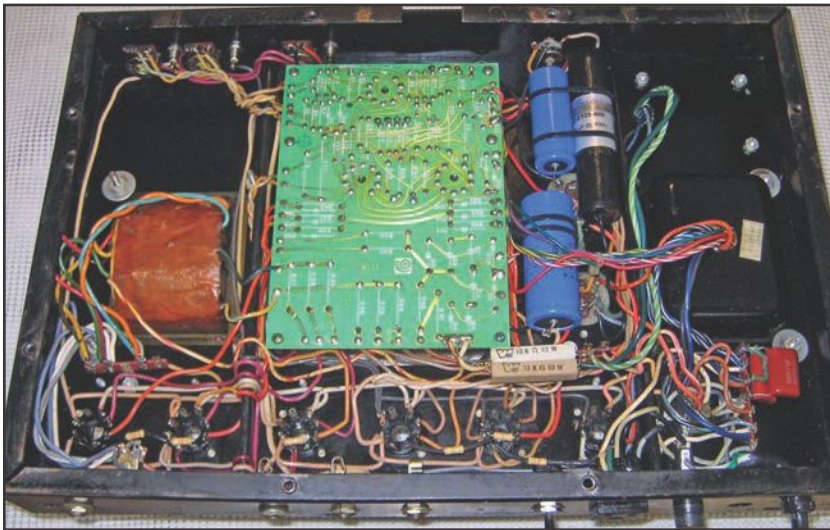


Photo 4: This underchassis of an Ampeg SVT bass amplifier's power amp incorporates a mix of point-to-point wiring and a circuit board.

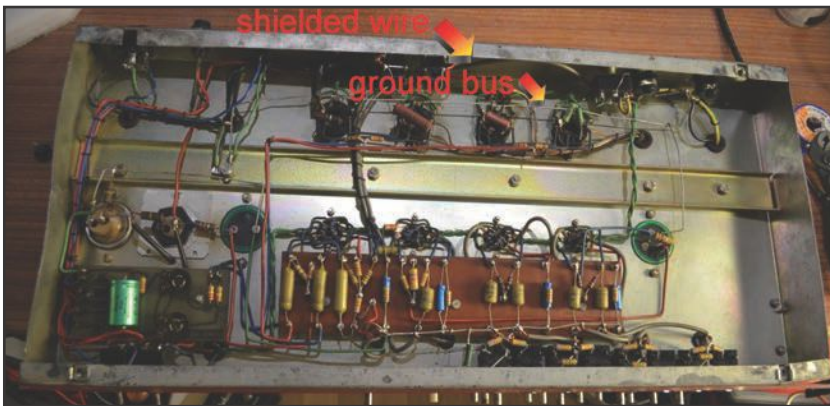


Photo 5: This Australian Simms-Watts PA amplifier uses both point-to-point wiring and circuit boards.

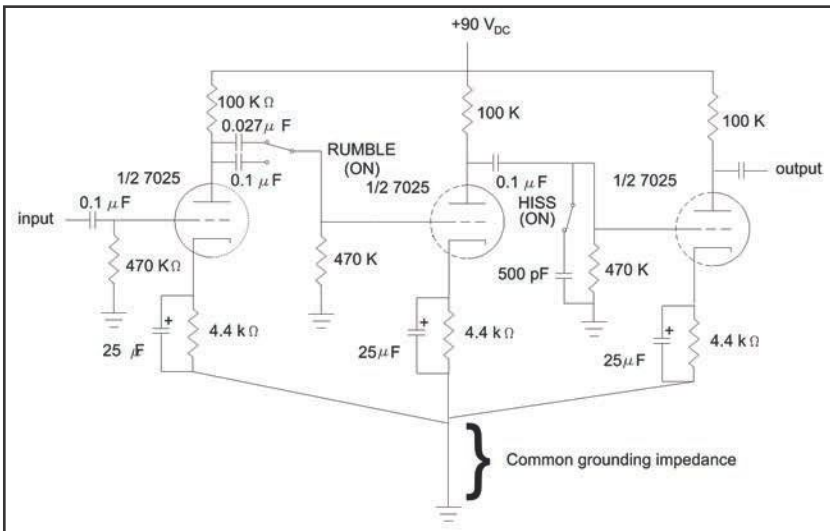


Figure 1: Common grounding impedances are one of the most common sources of hum and noise in audio equipment.

Avoid Noise and Hum

Two other good practices can also be seen in Photo 5. The components connected to the controls (which handle low-amplitude AC signals) are mounted directly on the controls themselves, avoiding leads that could pick up noise and hum. Also notice that the power supply is at the opposite end (left side in the illustration) from the inputs and preamp tubes. The preamp tube bases are seen at the top of photo (near the back of the chassis), at some distance from the power supply and power tubes. (The power tube bases are nearer the front of the chassis.) All these layout decisions bring us to two more rules:

- Keep the smallest-amplitude signal wires as far as possible from the large-signal wires, in order to prevent parasitic oscillations in the circuit.
- Keep the input wiring as far as possible from the hum fields produced by the power transformer, and by the wiring between the rectifier and the filter capacitors.

Note that the only shielded wire used in this PA amplifier is one small link between the inputs and a switch located on the back panel.

Proper Grounding

Keeping hum and noise low in any audio electronics equipment requires proper grounding. Proper grounding practices are not always well-understood. In particular, approximations that apply to most parts of electronic circuitry may not apply to grounding. We usually completely neglect the resistance of wires and cables in an electronic circuit, and so we naturally assume that any point on a chassis, or any point that is wired to ground, will automatically be at zero volts with respect to ground. However, the small signal voltages in many types of electronic equipment can render this assumption unreliable. The input voltage to a phonograph preamplifier or a mic preamp is of the order of millivolts.

In a typical listening room, the ratio of the audio program sound pressure level (SPL) to that of the background noise may be about 60 dB. Thus, noise in the amplifier that is—say—55 dB below the signal level will be reproduced in the speakers at a level above the background noise, and thus, will be audible. A noise voltage 55 dB below the 3 mV output of a phono cartridge has an amplitude of about 5.33 μ V. If the current to actual ground (the ground point of the circuit's input) is substantial, very little resistance is required to produce this tiny noise voltage.

The most common grounding error is called a "common grounding impedance." **Figure 1** shows a

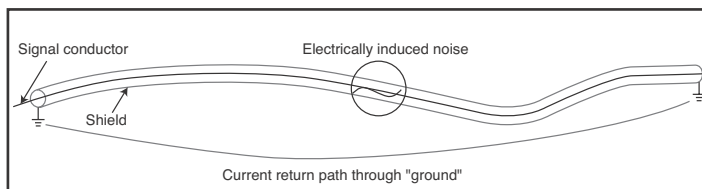


Figure 2: A signal conductor's shield that is grounded at both ends can create a ground loop.

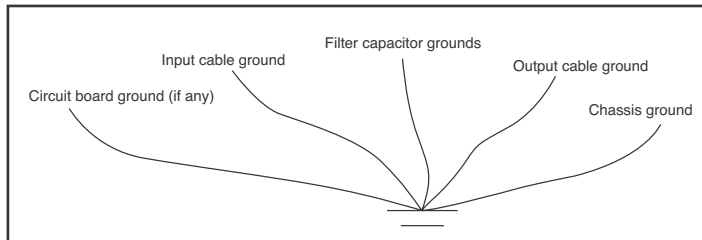


Figure 3: This "star" grounding method prevents noise from common grounding impedance and ground loops.

three-stage amplifier whose cathode resistors are all grounded, but not directly. If each is grounded directly to the circuit's common ground point, all will be well. But, in the photo, the bottom of each cathode resistor is connected to a separate terminal, and subsequently that terminal is grounded through a wire. The wire acts as a common grounding impedance. Its connection point to the cathode resistors will not be a true ground, but will have a small signal voltage created by the current from each tube stage passing through the wire's resistance. Common grounding impedances can cause hum or oscillation, depending upon where they are located in a circuit. This problem is greatest in low-impedance portions of a circuit (power supplies and output stages).

Although we seldom find digital circuits in hollow-state equipment, there are occasional exceptions. Since the ground currents from digital equipment often contain switching spikes, the results of a common grounding impedance involving digital circuitry are likely to be quite irritating for a listener. For this reason, it is essential to use separate digital and analog grounds in such equipment. In many instances, there is no reason that the digital and analog grounds must be connected at all, since although analog grounds should be connected to the chassis, digital grounds usually do not need to be—they can simply be common connections that are floating with respect to the chassis.

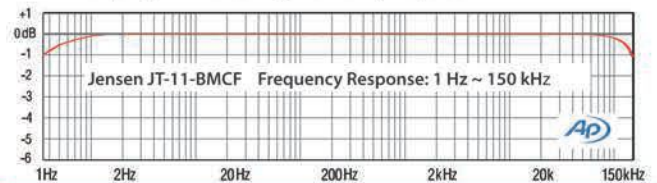
Power supply filter capacitors have fairly high ground currents that are rich in noise, and thus, special care needs to be taken in grounding them, to avoid common grounding impedances.

The third instance of common grounding impedance problems occurs when two pieces of audio electronic equipment are connected together, and their chassis are grounded to different points in the building electrical circuit. An example would be a live-sound mixer feeding a power amplifier, with the mixer and amp plugged into different power circuits. In this case, the common grounding impedance is the resistance between the power-amp ground and the mixer ground.

This problem does not occur inside the audio equipment, so its solution is not to be sought by correcting the grounding inside the chassis. Instead, the solution is to make sure all audio equipment that is interconnected is plugged into the same electrical power circuit. Hum and noise from this cause can be temporarily alleviated by using a ground isolator on one piece of equipment, breaking the path to the electrical ground for that device. However, this approach should only be used for troubleshooting, since leaving electrical equipment with



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the safety ground disconnected is unsafe and contrary to electrical codes.

The other common noise source involving grounding is the so-called “ground loop.” Ground loops usually involve a signal conductor in which noise voltages or currents are induced by alternating electric or magnetic fields. Although electromagnetically induced currents are a problem in circuits involving long exposures of the signal conductor to the electromagnetic field (telephone lines and very long mic cables), this type of induced noise is seldom a problem in properly laid out electronic circuits. An unshielded wire from a phono input to the grid of the first preamp tube, routed too close to an unshielded power transformer would be an exception. But usually, electrical induction (unintended capacitor action) is the bigger problem inside electronic equipment chassis.

Figure 2 shows a common ground loop problem. A shielded signal cable is routed in a region where a noisy electric field is present. The field induces AC noise into the shield, and since the shield is grounded at both ends, noise current flows in a loop through the shield and ground. The shield surrounds the signal conductor,

so it induces noise voltage into that conductor. The way to avoid this problem is to ground shields only at the end closest to the input of the amplifier fed by the signal.

In general, the “true” ground point of an amplifier is the ground point at the input tube, transistor, or IC. Each other ground point (except the digital ground, if present) should be connected with a single low-resistance wire to that “true” ground, as shown in **Figure 3**. Sometimes when there are multiple grounds involving low-current signals in a hollow-state circuit, these are connected to a large-diameter (perhaps AWG 14) bare wire “ground bus,” which is then grounded to the circuit’s input ground.

A careful look at Photo 5 will reveal the ground bus in the Simms-Watts amplifier. Strictly, this practice does involve a common grounding impedance, but since the bus has very low resistance, and all the ground currents are small, it usually does not create noise problems. However, filter capacitor grounds should not be connected to a ground bus.

Good layout and grounding practices are key parts to building low-noise audio circuits!

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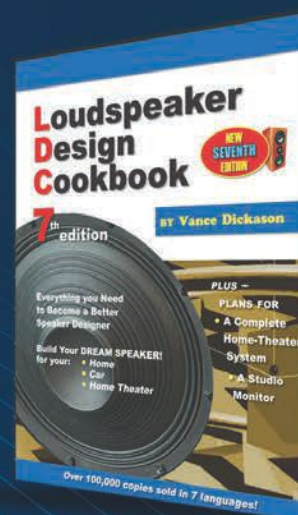
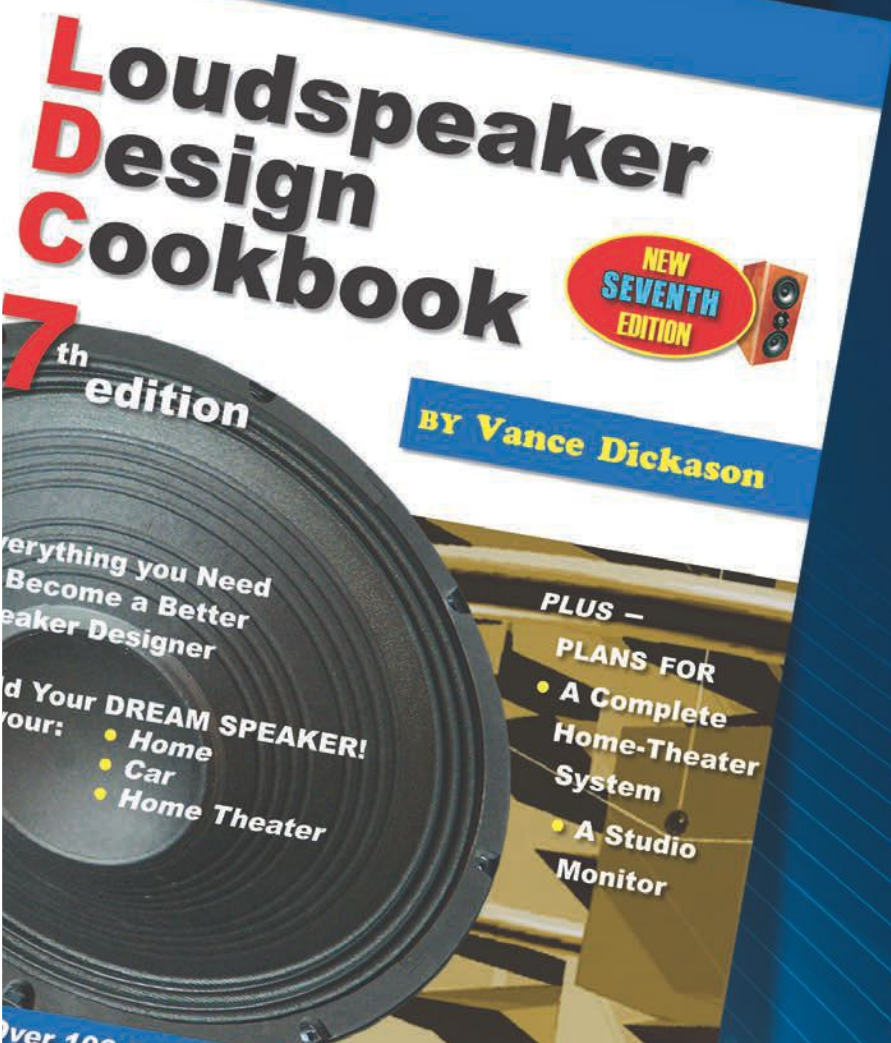
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