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November 2015

ISSN 1548-0628

www.audioXpress.com

audioXpress (US ISSN 1548-0628) is published monthly, at \$50 per year for the US, at \$65 per year for Canada, and at \$75 per year Foreign/ROW, by Segment LLC, Dan Rodrigues, publisher, at 111 Founders Plaza, Suite 904, East Hartford, CT 06108, US Periodical Postage paid at East Hartford, CT, and additional offices.

Head Office:

Segment LLC

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audioXpress

P.O. Box 462256

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E-mail: audioXpress@pcspublink.com

Internet: www.audioXpress.com

Postmaster: send address changes to:

audioXpress

P.O. Box 462256

Escondido, CA 92046, US

US Advertising:

Strategic Media Marketing, Inc.

2 Main Street

Gloucester, MA 01930, US

Phone: 978-281-7708

Fax: 978-281-7706

E-mail: audioXpress@smmarketing.us

Advertising rates and terms available on request.

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East Hartford, CT 06108, US

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Printed in the US

Oh, The Things We Could Do

At the International Broadcasting Convention (IBC) 2015 in Amsterdam, discussions about video standards and new technology introductions were almost mind-blowing. Product development cycles are shorter because companies need to stay ahead and—most importantly—because technology is being pushed by the IT, telecom, and semiconductor industries, and not just the video or broadcast industries anymore. And yet, the standardization process is much more fragmented and complicated than anything the audio industry has ever seen.

In comparison, the pace of innovation among audio companies is slower. Sometimes it even feels like they forgot what was already introduced, for no plausible reason. Let's take one of the most exciting technologies in the audio industry like AES42—the "Digital Microphone" Audio Engineering Society (AES) standard.

In these days of "digital-everywhere," the benefits of the AES42 standard are clear. The technology enables you to directly convert the signal from the microphone capsule to digital, eliminating the need for analog gain stages, improving signal transfer, and allowing for direct control of microphone features (e.g., variable polar-patterns with direct adjustment of the signal directivity and sensitivity). This means it is possible to capture the perfect signal even when physical access to the microphone is not possible (or advisable), such as during a live performance. The technology also allows for direct digital signal processing of the captured signal, including applying filters and limiting.

Sennheiser, Neumann, beyerdynamic, Schoeps, Microtech Gefell, and other companies introduced AES42 Digital Microphones in the last 15 years. Neumann, Marian, RME, StageteC, Lawo, and others have dedicated interfaces. Several digital field recorders are available with AES42 inputs from Nagra, Sound Devices, Zaxcom, Aaton, Sonosax, and more. To enable extended use of AES42 microphones with digital wireless transmitters, Zaxcom even introduced the STA042 digital stereo adapter in 2011.

All those systems support the standard originally published in 2001, updated in 2006, and revised in March 2011 (AES42-2010). Was this technology ignored for a reason? Maybe the standard allowed too much too soon? No doubt, one of the problems is that there are two modes in the specification, with only the more complete Mode-2 allowing for external synchronization, keeping all microphones in phase and using the same sampling-rate, compensating for different cable lengths and actually controlling the microphone from an external equipment—which is the whole point!

AES42 systems are expensive, which keeps the technology within a restricted application domain. Still the technology is widely used on classic recordings, and movie and television production. The main problem is that the larger market—the commercial, project, and home studios—doesn't have an affordable way to directly record from AES42 microphones to a DAW system. Also, there are no audio mixers with AES42 inputs.

Anyone who tried the Sennheiser/Neumann solutions and experimented with the software to control AES42 microphones understands how convenient and robust the whole solution is. If AES42 was widely available on popular recording interfaces with direct DAW support, the technology could be extremely popular. The problem seems to be AES42 was only pushed by the audio industry for technology's sake. Not as the disruptive winning concept it could be, if it had been introduced as a complete solution from the start.

Another current "Holy Grail" of the audio industry is the IP speaker and networked audio. At IBC 2015, two Finnish companies—Genelec and Jutel—discussed how they implemented the world's first all audio-over-IP (AoIP) system in a restaurant! Not a million dollar facility but the refurbishment of Restaurant Nallikari in the Finnish city of Oulu. The two companies designed a high-quality, flexible, multi-zoned AV network system using 34 of Genelec's first AoIP-enabled active loudspeakers and managed using Jutel's HIPman Hierarchical IP Audio Management system, which encodes the audio sources to RAVENNA/AES67 streams using ALC NetworX Virtual Sound Card technology.

The solution has low latency, any audio source can be routed to any loudspeaker, system control via tablets is simple and, most importantly—overall, it's less expensive. Multiple loudspeaker manufacturers are also implementing loudspeaker network connectivity using AoIP technologies, from Dante to AVB and RAVENNA/AES67. Everyone agrees it's the future.

In the next few months, the companies with well-designed solutions leveraging these (largely IT-inspired) technologies have a chance to dominate the market. I hope that they remember AES42.

João Martins
Editor-in-Chief



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COLUMNISTS

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

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

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.

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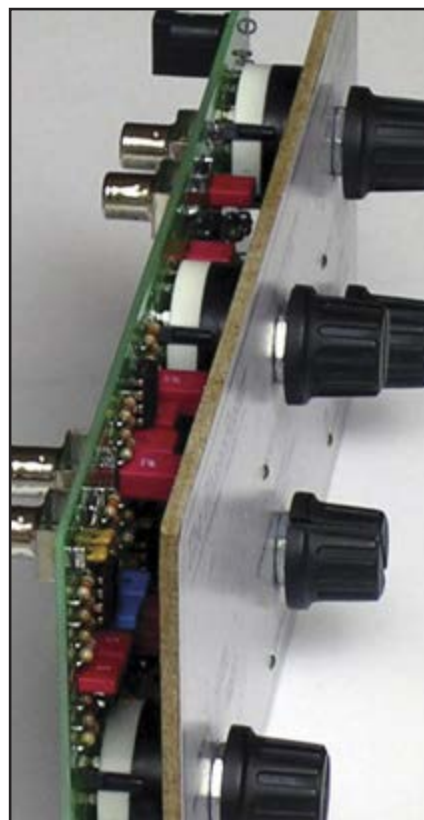


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High-Performance Audio from Smartphones?

The Testing Showdown

This month, we return to the topic of smartphone audio. Since our last article on this subject (*audioXpress*, March 2015), we've been busy in the test lab measuring the performance of eight high-end smartphones. The following is a discussion of the current state of smartphone audio playback as well as our test results.

By
Mike Klasco and Steve Tatarunis
(United States)

The market for boutique high-performance smartphones is growing. For years, there have been a limited number of designer phones that were more show than go—all of which were technologically dated even when first introduced. These products were merely glorified feature phones or smartphones with a fancy crust.

In fact, early smartphones with MP3 stereo music playback as well as the original standalone iPods and MP3 players really set playback standards back a couple of notches from CD quality with marginal bit rates and mediocre audio processors. Their primary goals were portability and affordability.

To achieve these goals, compromises were made, particularly in the area of audio playback quality. However, over the last few years decent to even awesome on-the-go earphones with microphones for phone calls have been introduced to the market. The earphones were one of the missing pieces and now we are seeing some superb on-the-go audio delivery systems bundled into feature-filled smartphones.

Apple, Samsung, LG, and HTC are in a constant battle to offer the best high-end smartphones powered by multi-core processors, loaded with tons of RAM, higher-definition displays, and the latest 4G connectivity. Thankfully, the market has

finally expanded to include audio performance and the winner is the consumer. Now, even Vertu has come out with its Signature series, offering audio "tuning" by Bang & Olufsen for a technically credible smartphone.

Playback Audio Quality

There are many features to consider when making a smartphone purchase but for now, we will focus on smartphone music playback audio quality. The most important aspect of smartphone audio quality is audio playback through earphones and headphones.

Almost all smartphone users have some or all of their music collection on their smartphones and place extraordinary value on this feature. Headphone listening is still predominantly done through wired connectivity and as such we will focus our efforts on evaluating what's coming out of that tiny headphone jack and leave discussions of the emerging Bluetooth audio market for another day.

Another consideration of music playback performance outside of headset playback quality is that the smartphone is often used as a music server for home audio systems. Many consumers connect their phones to their home systems via the headphone jack.



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So, our challenge for this month's article is to compare the audio quality of eight portable devices. These products include one non-smartphone (the iconic iPod classic), several smartphones known for high-quality audio including the iPhone 5, the iPhone 6, and two recent designs from newcomer Vivo—the Xplay 3S and the X5 Max—whose main design goals are to bring superior stereo audio playback quality to the Android smartphone platform.

Perhaps HD audio and HD Voice have been the impetus for smartphone manufacturers to think not just about the camera pixels and screen resolution, but also that the phone might be used for voice communications (sorry for the sarcasm) and this could be an opportunity for product differentiation. With its HTC One M8, HTC has raised the audio standard among the Big 3 (i.e., Apple, Samsung, and HTC).

HD Audio

HD Audio comes into play on a few fronts: stereo music playback through earphones or headphones, using your smartphone as a music server, and audio through the smartphone's speakers. Of course, we can simplistically limit our focus on the bit rate and clock speed—24 bit/96 kHz is better than CD quality (16 bit/44 kHz)—and memory capacity in smartphones is more than enough to accommodate more music than you can handle in high definition. Lossless formats such as FLAC (or ALAC for Apple users), can be either CD quality (16 bit/44 kHz) or master quality (24 bit/96 kHz). One by one, smartphone makers have been jumping on the “HD Audio” bandwagon with 24 bit/96 kHz-compatible conversion chips. HD Audio can go further, but first there needs to be more widespread availability of program delivery services. And then, there is the performance of the playback DAC.

Why Do We Care About the DAC?

When Philips introduced the first CD player in early 1980s, the sound was miserable. At first listen, the lack of hiss compared to cassettes was refreshing. But, after a while the sound was fatiguing. What was wrong could fill a book, but the last 30 years of digital audio design have addressed most of the issues. One dirty little secret was the intermodulation distortion (IMD) that was always missing from the early datasheets of the DACs, often 3% IMD or more.

IMD is nasty and the least musical distortion. It is also a source of listening fatigue, the stuff that makes you reach for the off switch after a while.

IMD originates primarily in the DAC (let's just say that over the years, the DAC has been the worst offender for IMD issues). Also, smartphones have so much digital and radio frequency stuff going on within such close quarters, it is impossible to wrestle the best from even great DACs. Nevertheless, recent smartphone introductions not only have top-grade DACs, but some of the smartphone designers have also managed to extract fine performances from the DACs and the rest of the smartphone's audio system.

Candidates for Test

Recently, we measured the performance of eight devices. Here's a quick summary of our test subjects.

- **Apple iPod Classic**—The iPod started the whole portable music craze when it was first introduced in October 2001 (see **Photo 1**). The current version plays video and audio. It has 160 GB of onboard storage. Supported audio formats include AAC, Protected AAC, HE-AAC, MP3, MP3 VBR, Audible (formats 2, 3, 4, Audible Enhanced Audio, AAX and AAX+), Apple lossless, AIFF, and WAV. In addition to its headphone jack, audio signal is also available via the proprietary Apple dock connector.

Photo 1: Apple iPod Classic



- **Samsung Galaxy S5**—Samsung's flagship Android phone (see **Photo 2**) was released in April 2014. (It was replaced by the S6, but we were unable to test the newest version for this article.) Its strong sales helped Samsung capture more than 32% of the global smartphone market in the second quarter of 2014. The S5 has 32 GB of onboard storage, with another 128 GB available

via micro SD. Supported audio formats include MP3, M4A, 3GA, AAC, OGG, OGA, WAV, WMA, AMR, AWB, FLAC, MID, MIDI, XMF, MXMF, IMY, RTTTL, RTX, and OTA.

Photo 2: Samsung Galaxy S5



Photo 3). The Z2 offers several standard audio enhancements and 32 GB of onboard storage with an extra 64 GB available via micro SD. Supported audio formats include MP3, 3GPP, MP4, SMF, WAV, OTA and OGGMP3, 3GPP, MP4, SMF, WAV, OTA, OGG MP3, 3GPP, MP4, SMF, WAV, OTA, OGG MP3, 3GPP, MP4, SMF, WAV, OTA, OGG MP3, 3GPP, MP4, SMF, WAV, OTA, OGG MP3, 3GPP, MP4, SMF, WAV, OTA, and OGG.

Photo 3: Sony Xperia Z2



- Sony Xperia Z2—Given Sony's audio heritage, it is not surprising to see it place an emphasis on quality audio delivery in smartphones (see

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- Apple iPhone 5s—Apple has always paid attention to smartphone audio quality and the iPhone 5s is no exception (see **Photo 4**). It comes with up to 64 GB of onboard storage and supports AAC, Protected AAC, HE-AAC, MP3, MP3 VBR, Audible (formats 2, 3, 4, Audible Enhanced Audio, AAX and AAX+), Apple lossless, AIFF, and WAV formats. In addition to the headphone jack, HD audio is also available from the proprietary Lightning connector.

Photo 4: Apple iPhone 5s



- Apple iPhone 6—The iPhone 6 (see **Photo 5**) is the successor to the iPhone 5, advertised storage and audio specs are similar to the 5S, but will they differentiate on the test bench?



Photo 5: Apple iPhone 6

Photo 6: HTC One M8



- HTC One M8—The M8 has a pair of front facing speakers (one on each side of the display) and HTC touts its on-board “Boom Sound” listening experience (see **Photo 6**). It offers 32 GB of onboard storage with up to 128 GB available via micro SD expansion and will play all popular audio formats.

- Vivo Xplay 3S—Vivo is a Chinese brand and its 3S has the distinction of being the first smartphone offered with a 2K display (see **Photo 7**). An equal emphasis is placed on audio, using the ESS Technology ES9018 DAC, the Texas Instruments OPA2604 amplifier, as well as DTS Headphones:X for surround sound support. The S3 has 32 GB of onboard storage and expansion via micro SD.



Photo 7: Vivo Xplay 3S

- Vivo X5 Max—The X5 Max is the successor to the Xplay 3S and is advertised as the “World’s Thinnest Smartphone” (5.08 mm depth) but there’s more to this phone than its dimensions (see **Photo 8**). Vivo built upon the success of the Xplay 3S to deliver a product that looks great and sounds even better.

Photo 8: Vivo X5 Max



Summary of Testing and Results

The test results for each of the eight smartphones are listed in the **Tables 1–8**. During testing, we made a few notes on the test methods. Here are some of our observations.

Except for the iPod classic, all the testing was done with 24-bit .wav files. The iPod Classic would not play back the .wav files we created until they were transcoded to an Apple lossless format.

Total harmonic distortion plus noise (THD+N) and output levels were measured by playing back a 997 Hz, 0 dBFS tone, except again for the iPod

Classic. When playing back a 0 dBFS signal, this device exhibited abundant distortion so a -1 dBFS tone was used instead. The measurement pass-band for THD+N was 20 Hz to 20 kHz, no weighting filter was used.

For all devices, noise was measured by setting the volume to maximum and playing back a file that contained just -144 dBFS dither noise. The noise was integrated for 5 seconds.

All the devices exhibited essentially flat frequency response, so reporting eight graphs of a flat line is pointless. The listed result is the maximum deviation from flat from 20 Hz to 20 kHz. In the case of the HTC One M8, "Bass Boost" was disabled to achieve this result.

We measured the interchannel phase but all the devices exhibited perfect phase alignment between left and right making this test somewhat uninteresting.

Crosstalk or channel separation was measured by playing back a 10 kHz, 0 dBFS tone in one channel at a time and measuring the leakage in

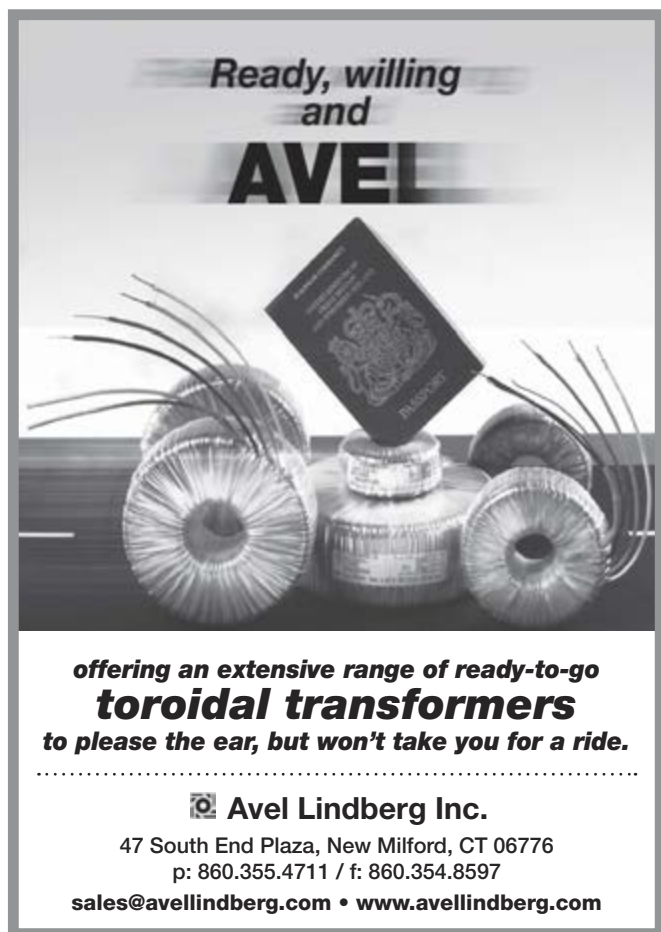
to the adjacent channel. The devices that exhibit poor separation when driving a 30 Ω load almost certainly have a resistor in the return (ground) path of the headphone jack. Some designers do this to provide a pseudo-balanced input and improve rejection of hum in the case where someone has their headphones plugged in at the same time that the phone is charging.

The High-Z configuration is driving a 100 k Ω input. Also, the 30 Ω configuration reflects the load presented by a typical headphone.

Closing Comments

There are dozens of recent smartphones on the market and well over a hundred once you account for older models still being sold. Picking the best for you can be a trying task. We won't attempt to rank the smartphones based on the test results but still, several things are clear.

The Sony Xperia Z2 comes in solidly last with high noise, low drive levels, and low THD+N.



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The HTC One, the Samsung Galaxy, and the iPod Classic are all quite reasonable middle of the pack contenders, but are all 10 dB worse

than either the iPhones or the Vivos when driving a headphone at a reasonable listening level (100 VRMS in to 30 Ω).

Vivo 3S

Vivo 3S

		Left		Right	
		Vrms	dB	Vrms	dB
High-Z	THD+N @ Max Level	2.03	-81	2.04	-81
	THD+N @ 1 Vrms	1.04	-91	1.03	-91
	THD+N @ Best Level	0.734	-93	0.74	-93
		uVrms		uVrms	
	Noise	7.97		9.92	
		dB dB +/- Deviation		dB dB +/- Deviation	
	Frequency Response	0.02		0.02	
		Deg. Rel. to L			
	Phase	0			
		dB		dB	
X-Talk	-69		-69		

		Left		Right	
		Vrms	dB	Vrms	dB
30 Ω	THD+N @ Max Level	0.51	-76	0.519	-77
	THD+N @ 100 mVrms	0.094	-90	0.096	-91
	THD+N @ Best Level	0.111	-92	0.113	-92
		uVrms		uVrms	
	Noise	2		2.1	
		dB +/- Deviation		dB +/- Deviation	
	Frequency Response	0.02		0.02	
		Deg. Rel. to L			
	Phase	0			
		dB		dB	
X-Talk	-51		-51		

Sony Experia Z2

Sony Xperia Z2

		Left		Right	
		Vrms	dB	Vrms	dB
High-Z	THD+N @ Max Level	0.29	-84	0.29	-84
	THD+N @ 1 Vrms	n/a	n/a	n/a	n/a
	THD+N @ Best Level	0.196	-84	0.196	-84
		uVrms		uVrms	
	Noise	17.7		17.7	
		dB +/- Deviation		dB +/- Deviation	
	Frequency Response	0.4		0.4	
		Deg. Rel. to L			
	Phase	0			
	dB		dB		
X-Talk	-108		-108		
		Left		Right	
		Vrms	dB	Vrms	dB
30 Ω	THD+N @ Max Level	0.237	-49	0.245	-49
	THD+N @ 100 mVrms	0.107	-64	0.111	-61
	THD+N @ Best Level	0.076	-75	0.078	-73
		uVrms		uVrms	
	Noise	15		15	
		dB +/- Deviation		dB +/- Deviation	
	Frequency Response	0.4		0.4	
		Deg. Rel. to L			
	Phase	0			
	dB		dB		
X-Talk	-35		-35		

Samsung Galaxy S5

Samsung Galaxy S5

High-Z	Left		Right		
	Vrms	dB	Vrms	dB	
	THD+N @ Max Level	0.42	-85	0.423	-85
	THD+N @ 1 Vrms	n/a	n/a	n/a	n/a
	THD+N @ Best Level	n/a	n/a	n/a	n/a
	uVrms		uVrms		
	Noise	3.76	3.24		
	dB +/- Deviation		dB +/- Deviation		
	Frequency Response	0.01	0.01		
	Deg. Rel. to L				
30 Ω	Left		Right		
	Vrms	dB	Vrms	dB	
	THD+N @ Max Level	0.394	-83	0.397	-82
	THD+N @ 100 mVrms	0.088	-83	0.088	-82
	THD+N @ Best Level	n/a	n/a	n/a	n/a
	uVrms		uVrms		
	Noise	3.56	3.08		
	dB +/- Deviation		dB +/- Deviation		
	Frequency Response	0.01	0.01		
	Deg. Rel. to L				
30 Ω	Left		Right		
	Vrms	dB	Vrms	dB	
	THD+N @ Max Level	0.394	-83	0.397	-82
	THD+N @ 100 mVrms	0.088	-83	0.088	-82
	THD+N @ Best Level	n/a	n/a	n/a	n/a
	uVrms		uVrms		
	Noise	3.56	3.08		
	dB +/- Deviation		dB +/- Deviation		
	Frequency Response	0.01	0.01		
	Deg. Rel. to L				
30 Ω	Left		Right		
	Vrms	dB	Vrms	dB	
	THD+N @ Max Level	0.394	-83	0.397	-82
	THD+N @ 100 mVrms	0.088	-83	0.088	-82
	THD+N @ Best Level	n/a	n/a	n/a	n/a
	uVrms		uVrms		
	Noise	3.56	3.08		
	dB +/- Deviation		dB +/- Deviation		
	Frequency Response	0.01	0.01		
	Deg. Rel. to L				
30 Ω	Left		Right		
	Vrms	dB	Vrms	dB	
	THD+N @ Max Level	0.394	-83	0.397	-82
	THD+N @ 100 mVrms	0.088	-83	0.088	-82
	THD+N @ Best Level	n/a	n/a	n/a	n/a

Apple iPhone 5S

Apple iPhone 5S

High-Z	Left		Right	
	Vrms	dB	Vrms	dB
	THD+N @ Max Level	1.02 -93	1.02 -93	-93
	THD+N @ 1 Vrms	n/a n/a	n/a n/a	n/a
	THD+N @ Best Level	0.481 -97	0.481 -97	-97
	uVrms		uVrms	
	Noise	6	6	
	dB +/- Deviation		dB +/- Deviation	
	Frequency Response	0.09	0.09	
30 Ω	Deg. Rel. to L			
	Phase	0		
	dB		dB	
	X-Talk	-79	-79	
	Left		Right	
	Vrms	dB	Vrms	dB
	THD+N @ Max Level	0.92 -89	0.92 -89	-89
	THD+N @ 100 mVrms	0.082 -92	0.082 -92	-92
	THD+N @ Best Level	0.29 -97	0.29 -97	-97
	uVrms		uVrms	
	Noise	3	3	
	dB +/- Deviation		dB +/- Deviation	
	Frequency Response	0.09	0.09	
	Deg. Rel. to L			
	Phase	0		
	dB		dB	
	X-Talk	-41	-41	

Tables 1–8: Each table shows the results obtained from tests on each of the eight smartphones: the Vivo 3S, the Sony Xperia Z2, the HTC One M8, the Samsung Galaxy S5, the Apple iPhone 5s, the Apple iPod Classic, the Vivo X5 Max and the Apple iPhone 6.

The Vivos and the iPhones are the cream of the crop, displaying excellent performance in our tests and all of them will provide excellent playback

quality, although the Vivos might be preferable over the iPhones when driving a high-impedance source such as an audio power amplifier. 

HTC One M8

		Left		Right	
		Vrms	dB	Vrms	dB
High-Z	THD+N @ Max Level	1.03	-79	1.03	-79
	THD+N @ 1 Vrms	n/a	n/a	n/a	n/a
	THD+N @ Best Level	0.651	-92	0.649	-92
		uVrms		uVrms	
	Noise	10.2		10.1	
		dB +/- Deviation		dB +/- Deviation	
	Frequency Response	0.7		0.7	
		Deg. Rel. to L			
	Phase	0			
		dB		dB	
		-93		-95	
30 Ω		Vrms	dB	Vrms	dB
	THD+N @ Max Level	0.96	-78	0.96	-78
	THD+N @ 100 mVrms	0.096	-81	0.096	-81
	THD+N @ Best Level	0.382	-90	0.382	-90
		uVrms		uVrms	
	Noise	9.5		9.6	
		dB +/- Deviation		dB +/- Deviation	
	Frequency Response	0.7		0.7	
		Deg. Rel. to L			
	Phase	0			
		dB		dB	
		-41		-41	

Vivo X5 Max

		Left		Right	
		Vrms	dB	Vrms	dB
High-Z	THD+N @ Max Level	0.63	-102	0.63	-102
	THD+N @ 1 Vrms	n/a	n/a	n/a	n/a
	THD+N @ Best Level	n/a	n/a	n/a	n/a
		uVrms		uVrms	
	Noise	2.4		2.4	
		dB +/- Deviation		dB +/- Deviation	
	Frequency Response	0.02		0.02	
		Deg. Rel. to L			
	Phase	0			
		dB		dB	
		-103		-88	
30 Ω		Vrms	dB	Vrms	dB
	THD+N @ Max Level	0.53	-101	0.53	-99
	THD+N @ 100 mVrms	n/a	n/a	n/a	n/a
	THD+N @ Best Level	n/a	n/a	n/a	n/a
		uVrms		uVrms	
	Noise	2		2	
		dB +/- Deviation		dB +/- Deviation	
	Frequency Response	0.02		0.02	
		Deg. Rel. to L			
	Phase	0			
		dB		dB	
		-40		-41	

Apple iPod Classic

		Left		Right	
		Vrms	dB	Vrms	dB
High-Z	THD+N @ Max Level	0.96	-88	0.95	-87
	THD+N @ 1 Vrms	n/a	n/a	n/a	n/a
	THD+N @ Best Level	0.483	-89	0.48	-88
		uVrms		uVrms	
	Noise	20.5		21.5	
		dB +/- Deviation		dB +/- Deviation	
	Frequency Response	0.1		0.1	
		Deg. Rel. to L			
	Phase	0			
		dB		dB	
		-94		-94	
30 Ω		Vrms	dB	Vrms	dB
	THD+N @ Max Level	0.77	-26	0.77	-26
	THD+N @ 100 mVrms	0.101	-83	0.102	-83
	THD+N @ Best Level	179	-86	180	-85
		uVrms		uVrms	
	Noise	17		18	
		dB +/- Deviation		dB +/- Deviation	
	Frequency Response	0.1		0.1	
		Deg. Rel. to L			
	Phase	0			
		dB		dB	
		-41		-42	

Apple iPhone 6

		Left		Right	
		Vrms	dB	Vrms	dB
High-Z	THD+N @ Max Level	0.26	-92	0.26	-92
	THD+N @ 1 Vrms	n/a	n/a	n/a	n/a
	THD+N @ Best Level	n/a	n/a	n/a	n/a
		uVrms		uVrms	
	Noise	3.4		3.3	
		dB +/- Deviation		dB +/- Deviation	
	Frequency Response	0.04		0.04	
		Deg. Rel. to L			
	Phase	0			
		dB		dB	
		-64		-64	
30 Ω		Vrms	dB	Vrms	dB
	THD+N @ Max Level	0.23	-92	0.23	-92
	THD+N @ 100 mVrms	0.11	-90	0.11	-60
	THD+N @ Best Level	n/a	n/a	n/a	n/a
		uVrms		uVrms	
	Noise	2.2		2.1	
		dB +/- Deviation		dB +/- Deviation	
	Frequency Response	0.04		0.04	
		Deg. Rel. to L			
	Phase	0			
		dB		dB	
		-41		-40	

Sound Cards for Data Acquisition in Audio Measurements (Part 6)

Examples of Sound-Card-Based Measurements

In this article series, we are examining ways to create a low-cost system for lab-grade audio electronics measurements. Now, I'll discuss a few measurement examples to illustrate the power and the versatility of sound-card-based measurement.

By
Stuart Yaniger
(United States)

The first example I will discuss is a gain block for a hybrid tube gain block intended for a Recording Industry Association of America (RIAA) preamplifier. The second example takes a brief look at the distortion characteristics of a commercial preamplifier. The third is rather more outré—a gadget that is intended to plug in between a headphone amplifier and headphones, claiming to significantly improve the sound through means about which the manufacturer is rather unclear. And finally, to illustrate the usefulness of sound card measurement in the design process, I'll explore the effects of load on power supply noise for a common IC regulator circuit. These examples just scratch the surface, but it will give you an overview of some of the more useful ways that sound card measurement can be of use in audio testing.

These measurements were all done with my M-Audio AP192 PCI card in a Hewlett-Packard PC with the "novel" mechanical modification shown in the first part of the series "Sound Cards for Data Acquisition in Audio Measurements" (*audioXpress*, June 2015).

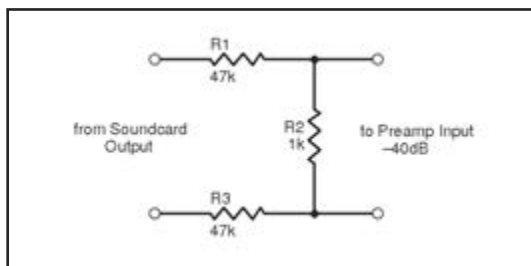


Figure 1: A balanced attenuator that goes between the sound card output and the preamplifier gain block is tested to simulate a phono cartridge impedance and level.

I used various bits and pieces of the software I mentioned before, but there is a definite slant toward the AudioTester—not because the other packages are less good (in some ways, and for some measurements, they're better), but because at the time I took these measurements, it was the one I used day in and day out. (However, these days, I am using ARTA and Virtins more and more.) Familiarity breeds efficiency! Where a particular measurement was easier with different software, I bowed to the inevitable. The Millett interface box and a few improvised bits also make an appearance.

Preamplifier Gain Block Measurement

The preamplifier under test is a gain block of about 36 dB ($A = 60$) with balanced inputs and outputs. Since this is intended for phono preamp service, the likely input voltages will be in the millivolt range. So we will have two tasks: create a balanced input signal at the millivolt level (while maintaining a good signal-to-noise ratio) and measuring the output for noise and linearity.

For the first task, the easiest way is to use the sound card output, but reduce the output level using the mixer controls or the level control in the measurement software. But "easiest" does not necessarily mean "best." The problem is noise near our "sweet spot," a nominal 1 V sound card output level, the signal-to-noise ratio (SNR) is somewhere close to 100 dB, which is excellent. But drop that output level 40 dB to 10 mV and the SNR ratio is more like 60 dB, which is likely to be noisier than



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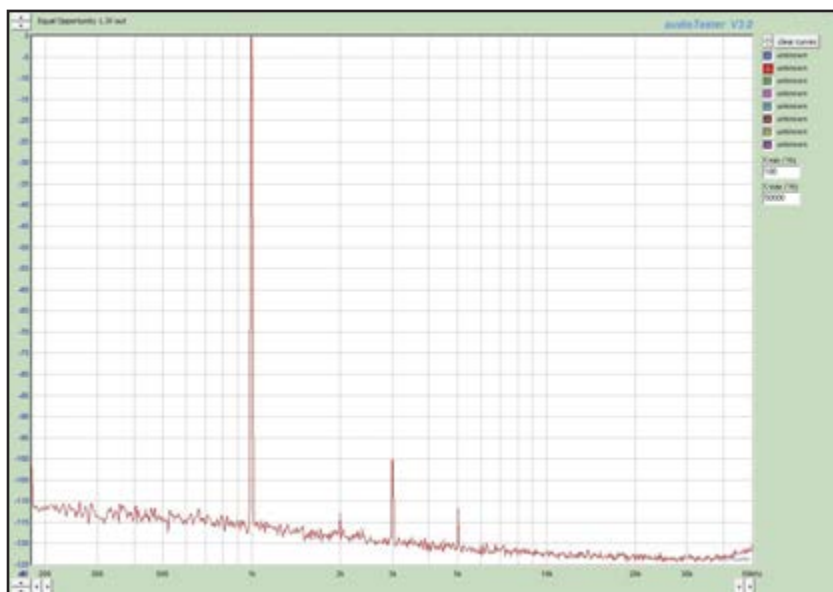


Figure 2: The harmonic distortion spectrum of a preamplifier gain block is shown at 1.3 V output.

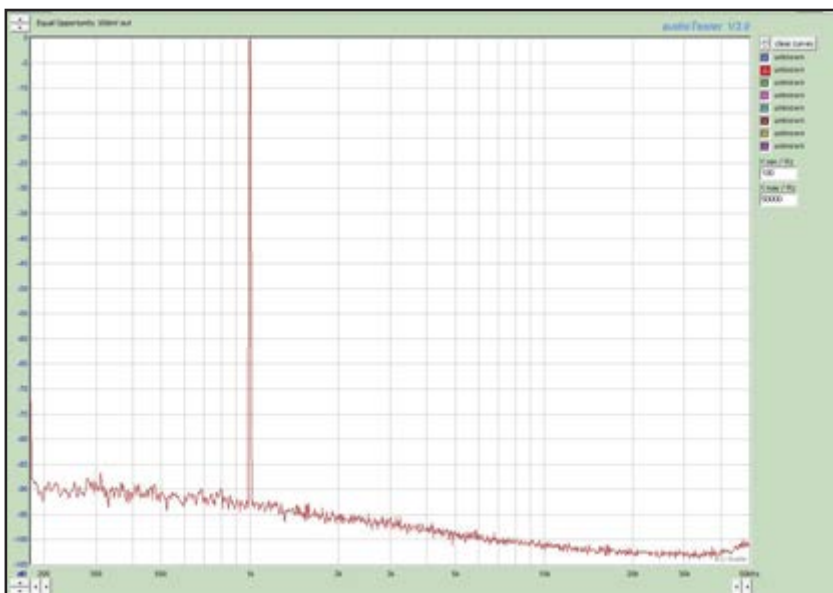


Figure 3: The harmonic distortion spectrum of the same gain block is shown, but with the output reduced to 100 mV. Since noise is constant and the signal is about 20 dB lower, the noise is about 20 dB worse.

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.

the preamp that we want to measure. The solution is to create a balanced attenuator, which can easily be fabricated from three resistors. **Figure 1** shows the one I used. The series resistors R1 and R3 were chosen to be large enough not to load down the sound card output (which is happiest with a 2 k Ω or higher load), but not so large that their parasitic capacitances become significant.

R2 was calculated by a simple voltage divider equation, but using the sum of R1 and R3 (which should be matched for best balance). To get realistic results, it's nice to have R3 somewhere near the DC resistance of the source that is expected to be driving the preamplifier. In this case, the source will be a moving magnet phono cartridge, which is likely to have a DC resistance of about 1 k Ω . Set $R1 = R3$, and $R2/(2 \times R3)$ should be approximately 0.01 to get the desired 40 dB level reduction. So I set R2 to 1 k Ω and R1 and R3 to 47 k Ω (a standard value), which is very close.

On the output end, we are taking the differential signal off the plates of a pair of tubes, so there's a lot of DC (nearly 100 V). Also, the impedances are likely to be several thousand ohms, so we'll need to buffer. In this case, I used both channels of my BUF03 box, with a channel sampling each leg of the preamp's differential output. I connected the buffer box's outputs to one channel of the sound card, taking advantage of the sound card's balanced input.

I took the basic measurements—the harmonic and intermodulation distortion, which provide a measure of the preamp's linearity. **Figure 2** shows the harmonic distortion spectrum for a 1 kHz sine wave at 1.3 V out (corresponding to 20 mV input from the attenuator and 2 V output from the sound card).

As expected for a well-balanced differential circuit, the even-order distortion is nearly canceled, and the dominant harmonic is the third at slightly under -100 dB (0.001%). The vertical axis is set to be relative to the fundamental rather than relative to a voltage standard (e.g., 1 V).

AudioTester has a provision for calibration. Once that's done, you can choose the display to show decibels relative to the highest peak in the spectrum, dBV (decibels with respect to 1 V), dBu (decibels with respect to 0.775 V), dB relative to full-scale, or dB relative to measured value at 1 kHz. Your choice.

Figure 3 shows the distortion spectrum with the preamplifier's output reduced to 100 mV. The noise floor is, as expected, 20 dB higher relative to the signal, but the distortion products seem to have largely disappeared. This measurement also indicates that the noise limitation in the measurement is not the sound card, but rather the

preamp itself, so it's a useful control measurement.

Turning to intermodulation distortion (IMD), AudioTester gives the option of a two sine wave measurement, with the ability to choose both frequency and relative amplitude of each of the two sine waves.

I arbitrarily picked 10 kHz and 11 kHz at equal levels, and adjusted the sound card output to give a preamp output of 1.3 V. **Figure 4** shows the spectrum I obtained.

Intermodulation products can be seen at the difference frequency of the two sine waves (1 kHz) and as sidebands at 1 kHz above and below the randomly selected 10 kHz and 11 kHz test signals. With the worst IMD product below -100 dB with respect to the test signals, I am not unhappy with this result. At this point, we're starting to challenge the lowest limits of the test gear, and this is well under the floor of what most of the classic dedicated IMD test sets could measure.

Line Amplifier Measurement

My doctor is an audiophile, and I occasionally do repair work for him as a favor (and in exchange, he does repair work on me). In this case, he bought a used 1980s vintage Nakamichi preamplifier that needed a bit of TLC to get it working like new. Out of curiosity, I wanted to see how its distortion looked. I set the line stage gain to unity (using the volume control) and fed in a 2 V signal from the sound card. The STEPS software from ARTA has a nifty capability that measures each distortion harmonic as a function of frequency, which is a useful way of uncovering the dominant distortion mechanisms.

Figure 5 shows a screen capture of the measurement. Several things can be inferred from the data, the first being that the overall distortion is quite low, better than 0.001% at midband. The third harmonic is dominant, suggesting that the circuit has a complementary or push-pull topology (since odd-harmonic distortion is symmetrical on the top and the bottom of the test waveform). The second harmonic rises slightly at low frequencies, suggesting that the coupling capacitors used may be polar electrolytics, which would be expected to show higher distortion at low frequencies, with the distortion being non-symmetrical on the positive and negative halves of the waveform.

The overall distortion rises with increasing frequency, which is likely a function of the preamp's compensation causing a rolloff of open loop gain (and hence reduced feedback) at higher frequencies. That's a lot of information from a quick measurement!

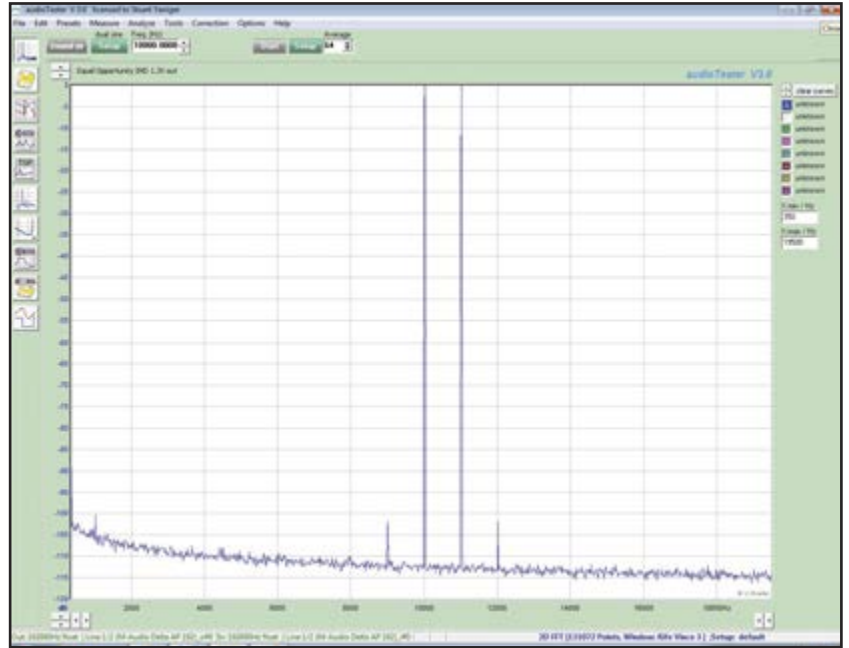


Figure 4: The intermodulation distortion (IMD) spectrum of the same gain block is shown at 1.3 V output with a 10 kHz/11 kHz 1:1 test signal.

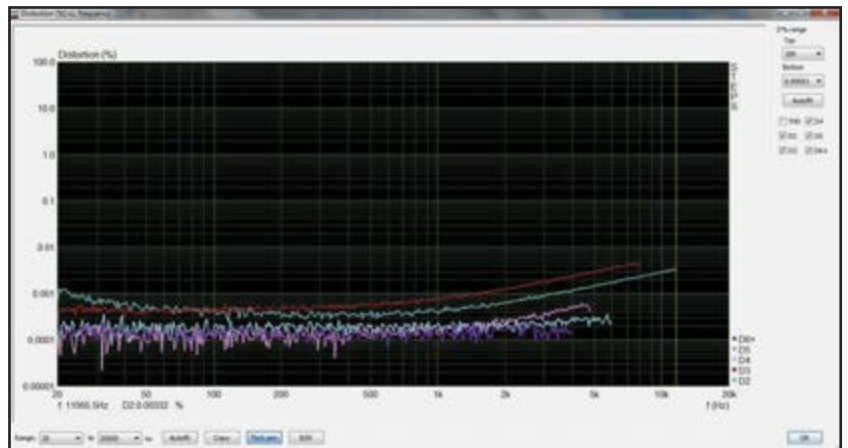


Figure 5: Distortion vs. frequency is shown for each harmonic of a vintage preamplifier.

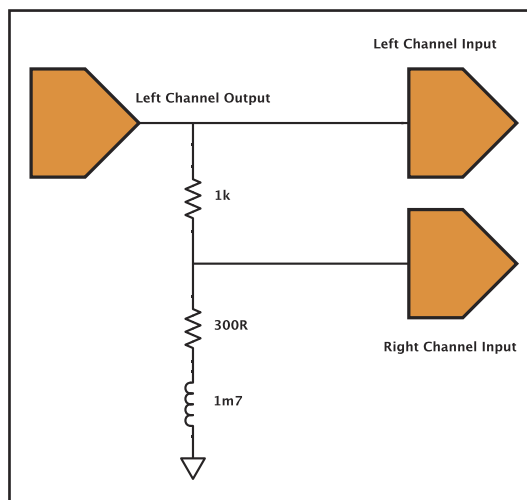


Figure 6: This is the AudioTester test setup I used to measure impedance.

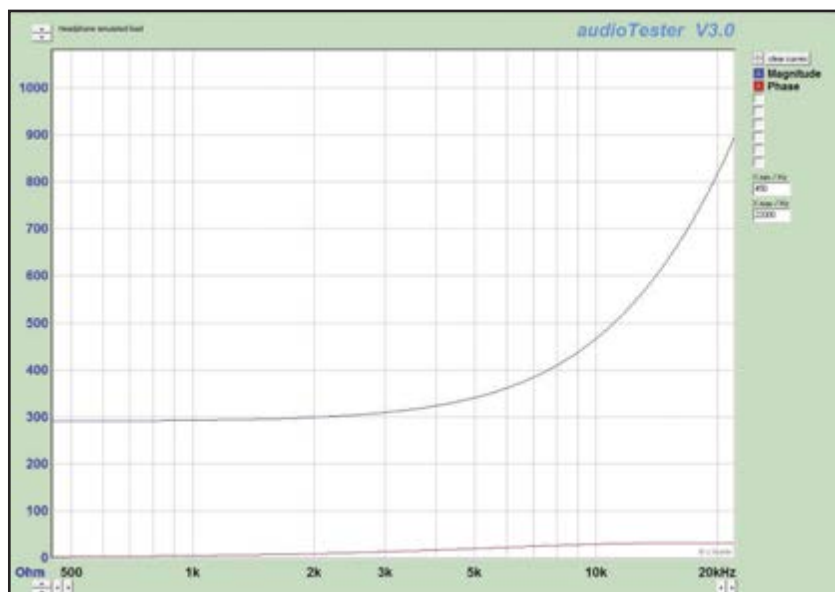


Figure 7: This is the impedance measurement of the simulated headphone load (300 Ω resistor in series with a 1.7 mH inductor).

Measuring a Tweak

The headphone gadget under test comprises a 0.25" stereo phone jack and plug situated on opposite ends of a plastic tube. The user inserts the plug end into the headphone output jack of a headphone amplifier or a similar source, and the headphones plug into the gadget's jack.

The manufacturer makes some claims about "maximizing your enjoyment with increased resolution and smoothness in the world's most holographic headphone experience." I'm not sure quite what that means, but any actual sonic changes



Figure 8: This is the frequency response across the simulated headphone load with and without the enhancement device. Note the expanded scale. The mismatch of 0.02 dB between right and left channel is well within the sound card specification and is negligible.

that the gadget might do would have to be reflected in an electrical measurement. You can't change the sound without changing the signal. It seems reasonable, given a generous reading of the claims, to measure frequency response and distortion (tonal balance), crosstalk (imaging), and noise to see if the device has any effect on these audible parameters. And, that seems like a good range of tests to show what our sound card measurement system can do.

A preliminary electrical test showed that the jack's terminals directly connect to the plug's corresponding terminals (resistance between corresponding terminals was in the milliohm range) and an ohmmeter check showed an open circuit between right hot, left hot, and common. For the crosstalk, noise, and distortion tests, the device was plugged into a unity gain headphone amplifier of my own design (basically an emitter follower with an FET input), which features very low noise and distortion.

For a load, I wanted to simulate a set of headphones. The manufacturer claims that the device was "voiced" using Sennheiser HD 800 headphones, which have a nominal impedance of 300 Ω . The published impedance curve shows about 300 Ω in the midrange, with the impedance rising in the treble because of the inevitable inductance of the headphones' voice coils.

A few minutes with a calculator suggested that the inductance was likely to be 1.5 to 2 mH. So I put a 300 Ω resistor in series with a 1.7 mH inductor to use as a dummy load to simulate the headphone load. (I chose the value using the rigorous scientific process of seeing what was in my box of coils.)

To make sure the proposed dummy load really looks like the impedance curve of the headphones, I measured it with my sound card system. Headphone jacks and plugs are single-ended rather than balanced, so I used adapters on the sound card balanced input and output to accommodate this. For convenience, the adapters I used terminated into RCA plugs. This is also convenient for measuring conventional single-ended audio equipment (which is the vast majority of consumer audio), but you could just as easily use mono 0.25" phone plugs.

Figure 6 shows the test setup with AudioTester, which I used to measure impedance. The sound card left channel output is used for applying the test signal through a 1 k Ω resistor.

AudioTester recommends using a resistor of similar value to the expected impedance. Since the impedance varies with frequency, this seemed like a reasonable value. You could probably use anything between 300 to 1,500 Ω and get a good result, but

the higher values present a friendlier load for the sound card output. In any case, the resistor value is entered into the test setup window of the software. For cases where the impedances are low (e.g., speaker testing), a small power amplifier is driven by the sound card output, the fixed resistor and the load are connected to its output, and the sound card inputs are connected as shown in **Figure 6**.

The sound card left and right inputs are connected across the excitation signal and the dummy load, respectively. The sound card generates a stepped set of tones spanning a user-specified frequency range and number of steps (also entered into the setup window of AudioTester in the impedance measurement mode). The impedance measurement is performed by measuring the voltage across the known-value series resistor and the voltage across the load. Then, use the voltage divider equation to calculate the load impedance as a function of frequency.

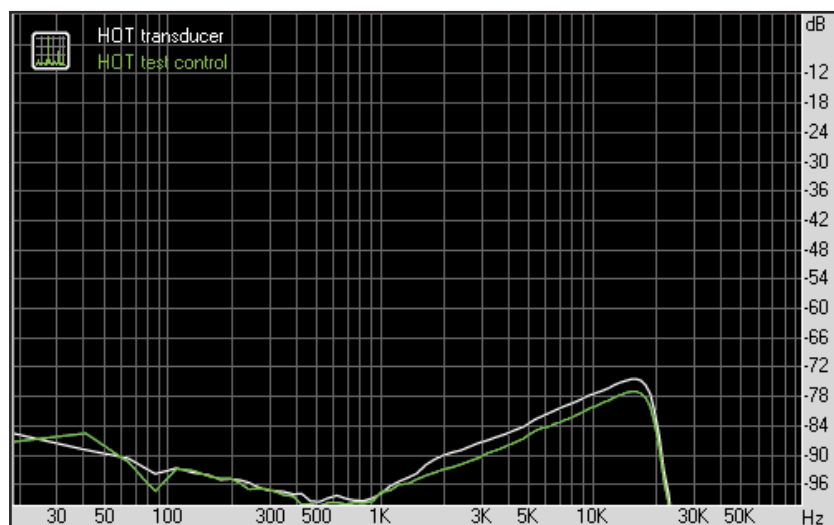
In theory, this can be done using an impulse or MLS excitation rather than swept or stepped sine waves. This is also how some other software packages perform the impedance measurement. The advantage to that approach is measurement speed, but you have to always be aware of resolution issues, especially at lower frequencies.

The stepped frequency method has the advantage of being able to choose steps that are either linear (i.e., the test frequencies are evenly spaced) or logarithmic (the test frequencies are fixed portions of an octave). One might choose the latter for a loudspeaker impedance measurement to get better resolution in the bass while not wasting a lot of measurement time with higher-than-needed frequency resolution in the treble. The difference in impedance between 20 Hz and 30 Hz is a lot more important than between 19,960 Hz and 19,980 Hz!

Tip: If your software uses Fourier Transform (FT) or maximum length sequence (MLS) for an impedance measuring signal, a nice way to increase bass resolution (at the expense of high-frequency bandwidth) is to drop the sampling rate. For example, if you set the sample rate to 32 kHz, you've lost most of your information above 15 kHz. But for a set sampling rate, the low-frequency extension and the resolution will be better by a factor of three compared to 96 kHz.

Figure 7 shows the resulting impedance vs. frequency. It resembles the impedance curve of electrodynamic headphones through midrange and treble frequencies, where we would expect to see the claimed effects of the headphone “enhancement” device.

Once I established the impedance curve was



reasonable, I began measuring frequency response with and without the device in circuit to see if it had any equalization properties. **Figure 8** shows the results of a swept sine wave frequency response measurement using AudioTester, both with and without the device inserted. As can be seen, there is no significant difference (this measurement exposes a minuscule 0.02 dB difference in gain between the channels of the sound card, but that difference is not altered by the device). So clearly, the device does not provide any significant equalization.

When one hears terms like “holographic” in the context of headphones, thoughts immediately turn to some sort of cross-feed circuit, where signals are leaked between channels, perhaps with alterations of phase or some filtering. To test this hypothesis, I ran a crosstalk measurement using the RightMark Audio Analyzer (RMAA). **Figure 9** shows the results. The slight rise in crosstalk at higher frequencies is

Figure 9: Here are the interchannel crosstalk measurements. The white line shows the results without this device inserted. The green line shows the results with the device inserted.

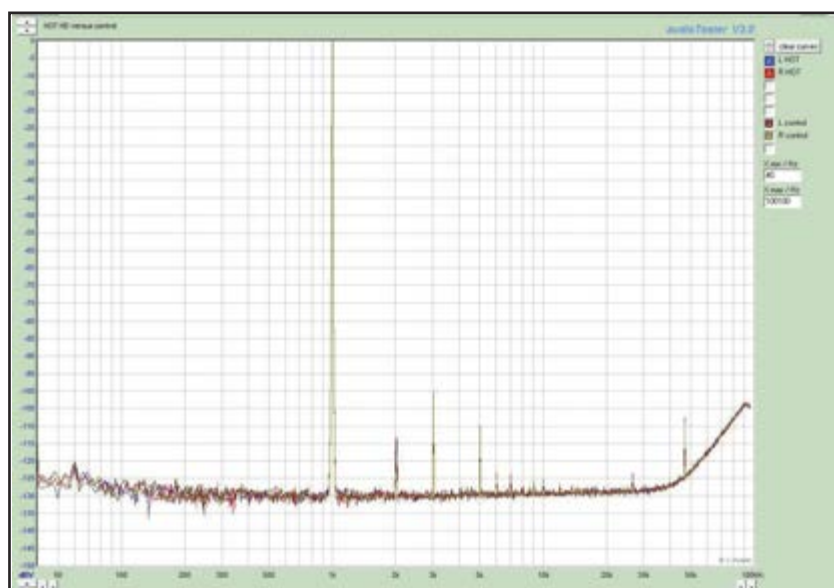


Figure 10: The harmonic distortion spectrum with 1 kHz excitation is shown across the simulated headphone load, with and without the enhancement device. There is no significant change.

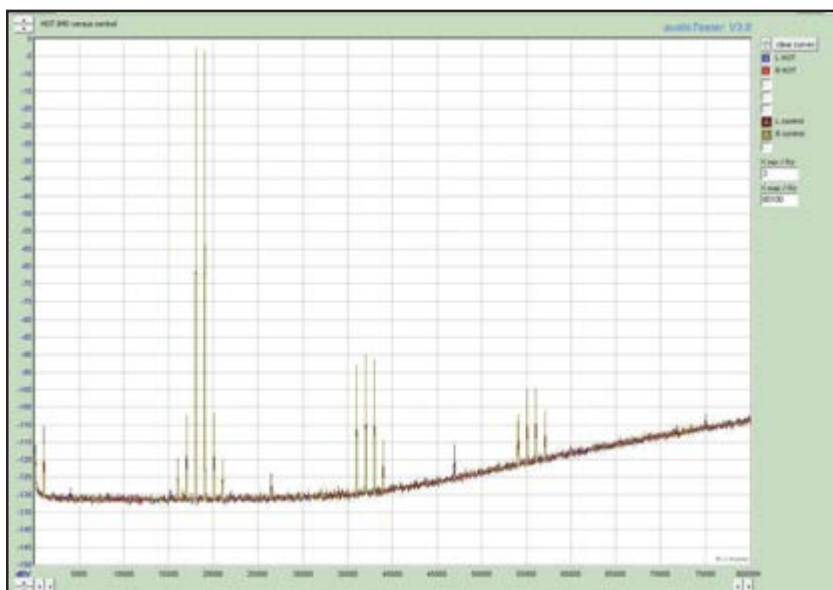


Figure 11: The intermodulation distortion is shown across the simulated headphone load, with and without the enhancement device. The test signal is 19 kHz/20 kHz dual sine wave. There is no significant change.

likely due to capacitive coupling between channels, but again, there's no real difference with or without the device inserted. So clearly, the device does not provide any significant cross-feed.

Perhaps there's some sort of nonlinear function involved? Noise reduction? There's one way to find out and that's to measure! First, I did a harmonic distortion measurement. **Figure 10** compares the 1 kHz harmonic distortion spectrum at 1 V, with and without the device inserted. The spectra overlay perfectly, and the noise floor is unaffected (and I might modestly comment that my little headphone amplifier seems to perform pretty well, with only

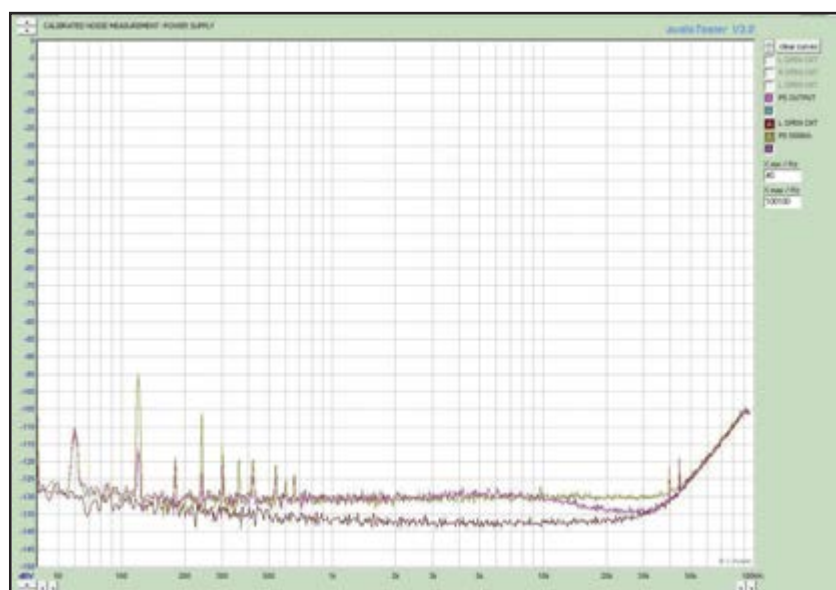


Figure 12: The noise spectrum of the power supply output is shown at 10 mA and 500 mA output current.

slightly more than 0.001% THD). Maybe there's something going on at a higher frequency? And a more complex test signal might unravel the device's secrets? For a test signal, I chose a combination of 18 kHz and 19 kHz sine waves to expose any IMD changes and the effects of any high-frequency harmonic distortion. **Figure 11** shows the results. As before, the distortion is quite low (0.005% or so), but neither the distortion nor the noise floor seem to be at all affected by the device.

With no changes in frequency response, crosstalk, noise, harmonic distortion, or IMD, the device apparently does not have any significant effect on the signal flowing through it, not atypical for audio "tweaks." But I don't feel that these measurements are wasted. Not only do they give a nice illustration of the power of sound-card-based measurement, they showed me that my little prototype headphone amplifier is a pretty high-performance unit!

Power Supply Noise

This is a lively topic of discussion among audiophiles, and having a sound-card-based measurement system will enable you to see what effect changes in a power supply design will have on output noise by examining the output noise's frequency-domain spectrum. You could also use the sound card to measure supply output impedance by injecting a signal into the supply output from a source with a known impedance and measuring the resulting supply output. If you're ambitious, you could even do this with a pulse signal, then a Fourier Transform into the frequency domain to get an impedance vs. frequency curve. I didn't do that here due to some time limitations, but it would be a great use of the sound card measurement capability.

In this measurement set, I was interested in the effects on noise and line rejection from loading the output of the power supply. Because the power supply output impedance and the DC voltage were low, I was able to run this measurement with the "bare" sound card and no interface box. The power supply under test was a simple 15 V raw supply fed into an LM317 regulator set for a 10 V output. I measured the output with no load (other than the set resistors of the LM317 circuit, which drew about 10 mA) and then with a 500 mA load. **Figure 12** shows the results.

The lower curves show the sound card without any input (control). The upper two curves show the power supply's output with and without loading. Both have a small 60 Hz peak (~110 dBV), probably due to radiative pickup in my test setup. The overall noise floor runs about 6 dB above that of the sound card

alone. With loading, two effects appear: the noise at frequencies above 13 kHz increases about 4 dB and the power supply ripple frequencies (multiples of 60 Hz) increase by about 20 dB. They're still low, but if low noise at these load currents is important in a design, I can now quantify any changes I make in the power supply circuit.

Summing Up

With a very low expenditure, the modern audio amateur or semi-professional can have a measurement capability more powerful than the best labs of the last century. With some minimal external hardware and free or inexpensive software, and recognizing bandwidth limitations, you can have a top-quality virtual spectrum analyzer, impedance bridge, oscilloscope, real-time analyzer, and signal generator capable of parts-per-million measurements. And when you get bored, you can use your test equipment center to play Solitaire! 

Author's Acknowledgments: I would like to thank Pete Millett for the boards, case, and critical parts for his

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iXBT.com | <http://audio.rightmark.org/products/rmaa.shtml>

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Virtins Multi-Instrument
Virtins Technology | www.virtins.com

excellent little interface box; the kind folks at ARTA and Virtins for generously providing me with their software; Jan Didden, Ladislav Bunta, Scott Wurcer, and Pavel Macura for advice and input; John Curl for the gift of two BUF03 ICs; and all of the wonderful people at Nicolet (especially their erstwhile Chief Scientist, Dr. Warren Vidrine) for teaching the much younger me everything I know about measurement.



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Binaural Sound from Two Loudspeakers (Part 2)

Software Options and Affordable Solutions

Last month's article in this realistic sound reproduction series discussed binaural sound from two loudspeakers, explaining the two basic concepts of crosstalk cancellation and spectral compensation. Since not everyone will be able to experiment with Dr. Edgar Choueiri's BACCH-SP processor box, we look at some software options and more affordable solutions.

By
Ron Tipton

(United States)

Hotto Ambiophonic Transcoder

First, let me mention that ambiophonic is not the same as Ambisonic (see my article "Ambiosonics,"

audioXpress, April 2015).

The free VST plug-in (shown in **Photo 1**) uses Recursive Ambiophonic Crosstalk Elimination (RACE) with the block diagram shown in **Figure 1**. The cancellation depth is not as great as can be achieved with other methods (e.g., Bacch-SP), but its cost is much less. This plug-in is available for Windows and Mac. The Windows version uses Java so the Java Runtime Environment must be installed. It's a free download from the Java website.

I am including a copy of the three-page user guide in the Supplementary Material, but I think a few words on two of the controls would be useful. If the 0 Hz to 4.5 kHz box is checked, a low-pass filter is applied. Unchecking the box increases the bandwidth to 20 kHz. The time delay can be increased to only 100 μ S, which is adequate for the 70 μ S recommended for the close loudspeaker placement of 16°. To use a standard stereo loudspeaker positioning of about 60°, a delay of around 260 μ S is needed so this plug-in is limited in that respect. Overall, it performed well in my Windows testing using the Cantabile and Reaper Digital Audio Workstations (DAWs). For some reason, Sony Audio Studio would not load it.

AmbiophonicDSP

This is an inexpensive (\$10) VST plug-in and you get both versions: 1.1 and 1.2 (see **Photo 2**). There are five built-in presets and you can easily

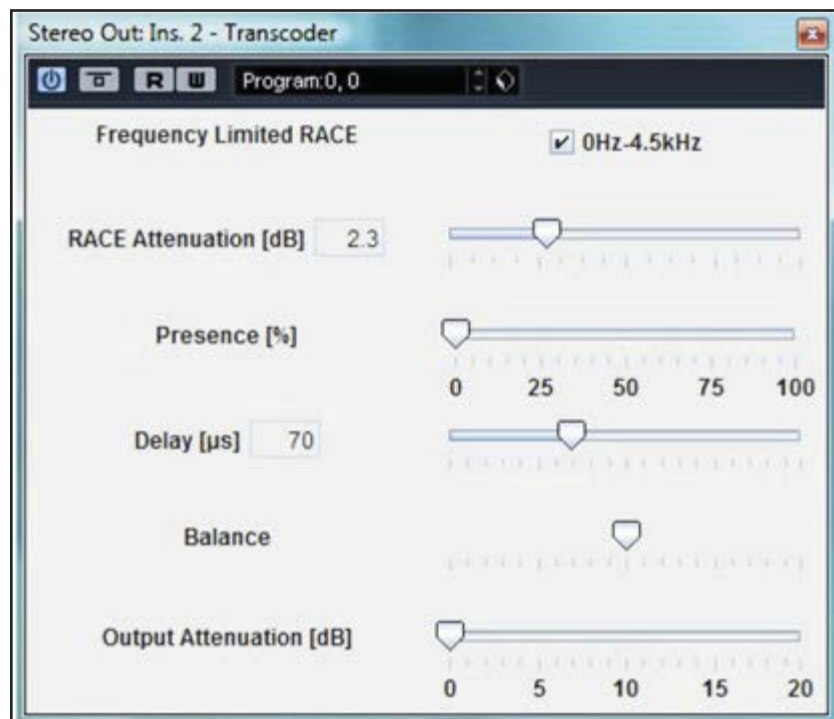


Photo 1: The Hotto Transcoder's controls are explained in its user guide included in the supplementary material. This VST plug-in is worth exploring because it has more value than its price suggests—it's free.

add others. The “Pop” preset, for popular music, emphasizes the music’s fullness. The “Jazz/Acoustic” preset has low coloration along with good soundstage widening and “Classical/Movie” combines the lowest coloration with moderate widening. “Bypass” turns off the processing so you can hear the difference.

A 16° speaker separation is recommended for best performance, but a standard 60° stereo separation can be used for good performance by increasing the delay time to about 260 μ S. Although not as impressive as the 120° soundstage from the close loudspeaker spacing, the music was pleasing from the stereo spacing with the increased time delay. A copy of the two-page user guide is included in the Supplementary Material.

SoundPimp

When I began my evaluation, I thought this was just another stereo enhancer, but I discovered the enhanced output couldn’t be recorded. I reread all the fine print and it specifically says it uses crosstalk cancellation.

SoundPimp is available as an application (virtual sound card) for Windows XP/7/8, Mac OSX, and Linux and as a VST plug-in for all versions of Windows, at the same \$52 price. The VST plug-in, which will run in trial-mode, has a graphic equalizer (see **Photo 3**).

I found it useful, but the virtual sound card applications appear to not have this feature. As I increased the width of the sound stage, I needed to boost the high-end frequencies by 5 or 6 dB to keep the sound balanced. Perhaps this should not be surprising because a low-cost plug-in would probably not have an optimal spectral compensation filter, so the EQ is helpful.

You will notice in **Photo 3** there are only a few controls besides the EQ: Speaker Angle, Soundstage, and Attenuator. It appears that Speaker Angle controls the time delay but don’t expect a good sound stage from a normal stereo loudspeaker separation.

The Attenuator is the volume control. There are two presets: Audiophile and Widescreen. They set the control sliders, but you change the settings. (The manufacturer says the internal processing is done at a sample rate of 192 kbps and a 24-bit depth.)

Processed music sounds very realistic and pleasing, but it’s best to keep the enhancement subtle. They recommend that your stereo speakers be positioned fairly close together at an angle of 10 to 20° because this increases the size of the sweet spot and the Speaker Angle control appears to work in this angle range. I have included a demo file from the website, SoundPimp-CD.wav, in the Supplementary Material.

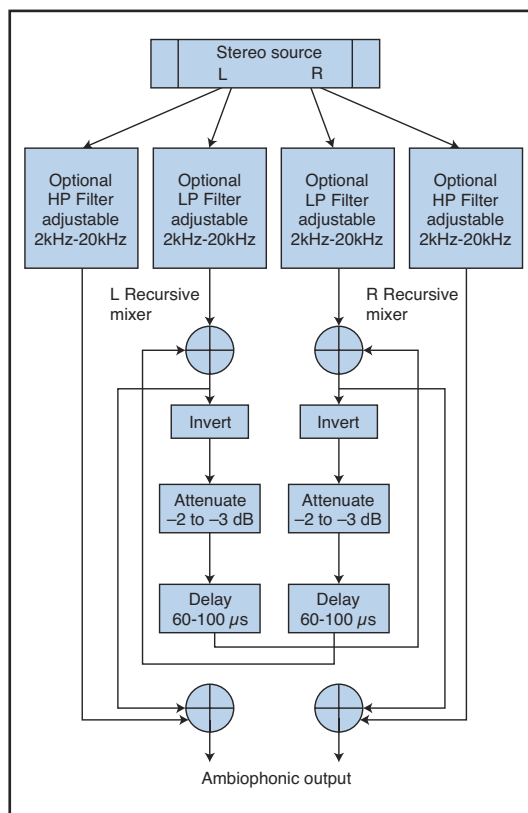


Figure 1: The Recursive Ambiphonic Crosstalk Eliminator (RACE) block diagram is apparently easier to implement than other more complex filters. This diagram indicates the input filters are optional and apparently not included in the AmbiphonicDSP or Soundpimp plug-ins. The delay time can also be longer than 100 μ S.

miniAMBIO

This is a hardware device available from Ambio4You and not a software plug-in. Its miniAMBIO version 2.0 sells for \$125, plus shipping from Hong Kong. The miniAMBIO provides a stereo input and a two-channel crosstalk canceled output (see **Photo 4**). It needs calibration using Ambio4You’s free PC or Mac setup software. Your loudspeakers need to be spaced 20° to 30°, which matches fairly well with other binaural recommendations.

I purchased one and it works extremely well with an ordinary stereo input. The sound stage is an apparent 180° wide and, if present in the music, individual instruments localize nicely. Does it sound better than the plug-ins? This is difficult to answer because it depends on the acoustics of your listening room, the type of music, and your



Photo 2: Some of these control names may seem strange but they are explained in the user guide, which is also in the supplementary material. I think its low price does not reflect the effort that went into writing it.



It's About the Sound

Photo 3: A view of the SoundPimp VST plug-in screen shows the graphic equalizer is on the left side with control points that can be dragged up or down. The control sliders for speaker angle, soundstage width, and volume are on the right.



own preferences. If you are interested in binaural sound, these solutions are inexpensive enough to try them all.

The initial setup is done by attaching your computer to the unit with the furnished USB cable and running the software. The “Quick Start” guide starts on page 7 of the 12-page user manual. It gives you the option of checking and verifying your setup choices before they are programmed into the unit. You can also choose “Restore to Default” to start over, which means your configuration is always easy to change.

You must have music playing through the unit during setup because all the changes you make are instantly heard in the output sound. In addition to the Quick Start instructions, there is a

two-part tutorial at: <http://ambiophonics.org/Tutorials/UnderstandingAmbiophonics.html> and http://ambiophonics.org/Tutorials/UnderstandingAmbiophonics_Part2.html.

When the setup has been completed, the computer connection is no longer needed. You can



Photo 4: The miniAMBIO is not very large but its performance is impressive. The shipping from Hong Kong was faster than I expected.

Project Files

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

Resources

RSB Consultants, “Point-of-View Binaural Recording with High-Definition Video,” www.immersav.com.

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Java | www.java.com

SoundPimp VST plug-in
SoundPimp | www.soundpimp.com

About the Author

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

choose any sound source: CD or DVD player, AM or FM tuner, computer's sound card, and so forth.

There are numerous technical articles and tutorials on its website. One of the more interesting, which is included in the Supplementary Material, is "Comparison of Cross-Talk Cancellation Filters." Also included are three demo .wav files starting with "ambio" that will let you preview this unit with your existing stereo system. However, you will probably need to relocate your loudspeakers to the recommended placement.

University of Southampton

The University of Southampton's Institute of Sound and Vibration Research has also been working on crosstalk cancellation. I have included two of their demo tracks in the Supplementary Material: isv-track2.wav and isv-track3.wav. The instructions for listening are explicit and generally follow the previous recommendations. With a speaker separation of about 14", sit 6' directly in front of them.

It is important that the speaker fronts are closely aligned as even a few tenths of an inch can make a noticeable difference. If possible, place the speakers near the room center to minimize acoustic reflections and slightly move your head to find the sweet spot. Actually, the sweet spot is not as small as some of the literature suggests. True, you can't walk around the room and hear the music in 3-D. But you don't have to sit motionless in your chair either. If you move your ears out of the sweet spot you will instantly know it!

Dr. Takashi Takeuchi, who is mentioned in my article "Binaural Sound from Two Loudspeakers (Part 1): Crosstalk Cancellation and Spectral Compensation" (*audioXpress*, October 2015), is an audio researcher at Southampton. You can find his biography and a list of his publications on the university website.

Lexicon CP-3

This Digital Surround Processor functions in four user-selectable modes: Panorama, Ambience, Reverberation, and Surround. In the Panorama mode, it uses crosstalk cancellation "effectively spreading the sound from the two front loudspeakers in a wide arc in front of the listener."

Apparently, this is not a current Lexicon product but it is available on the Internet at reasonable prices. I've included a copy of the owner's manual in the Supplementary Material. I have not purchased one so I can't give a "hands-on" evaluation, but it appears to be yet another reasonable approach to binaural listening.

RBS Consultants

Robert Schulein and his business partner Dan Mapes-Riordan have developed a crosstalk cancellation system they call ImmersAV Technology. Their approach is to combine point-of-view binaural recording with high-definition video. The added video provides additional visual clues to the audio's source location. They have an informative website and a large variety of video-binaural demos at www.youtube.com/user/immersav. In my experience, this demo collection is unique because it encompasses classical concerts, vocals, and jazz. Some are designed for headphone listening and they have an "(H)" in the title line. Others are for a pair of loudspeakers with an "(L)" in the title.

A high-speed Internet connection is needed to stream the high-quality audio and video. My cable connection is not fast enough to avoid an occasional stutter so I suggest you use one of the usually free browser add-ons to capture the stream and then listen and watch from your hard drive. You can freely download enough binaural music to keep you occupied for hours—Very enjoyable. 



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Smartphone Apps for Sound Measurement

Sound measurement began with basic devices. Over time, the tools have grown smaller and more complex. In this article, we explore some modern-day sound measurement apps that can be downloaded to most smartphones.

By
Richard Honeycutt
(United States)

We seldom take note of the humble beginnings from which modern science was developed. Galileo based his theory of the behavior of pendulums on observations he made of a swinging chandelier,

while he was not paying attention during Mass. His timing instrument was his own heartbeat. When W. C. Sabine made the measurements upon which he based his landmark theory of architectural acoustics, he used mouth-blown organ pipes as signal generators, his ears as detectors, and a stopwatch for timing. Early acoustics investigators using electrical instruments obtained better repeatability by using calibrated test microphones such as the Western Electric 640AA.

By the 1960s, test equipment had been “miniaturized” to the point that a complete test setup would fit into a station wagon’s cargo area. Sweep-frequency generators such as the one shown in **Photo 1** served as signal sources. More modern test microphones from several manufacturers included the detectors, and graphic level recorders somewhat larger and heavier than the sweep-frequency generators captured the sound levels. Even with all this “miniaturization,” few acoustical engineers felt a need to patronize a gym for exercise after a day’s testing!

In comparison to these behemoths of the past, today’s assortment of smartphone-based sound measurement apps outshine Dick Tracy’s fictitious wrist TV in terms of increased functionality per unit of weight! These apps comprise a vast range of tools, some intended for amateurs, and some for professionals. Their measurement capabilities



Photo 1: This General Radio sweep-frequency signal generator was only part of a “portable” acoustical test setup in use during the 1960s.

App Name	SPL Modes	Platform	Website Information
Advanced decibel Meter	SPL A	iOS	https://itunes.apple.com/us/app/advanced-decibel-meter/id595718101?mt=8
Advanced decibel Meter	SPL A	Android	https://itunes.apple.com/us/app/advanced-decibel-meter/id595718101?mt=8
dB Volume meter	SPL	iOS	https://itunes.apple.com/us/app/db-volume-meter/id592769907?mt=8
deciBel	SPL	Android	https://play.google.com/store/apps/details?id=bz.bsb.decibel&hl=en
Decibel Meter	SPL	Android	https://play.google.com/store/apps/details?id=cn.menue.decibelmeter&hl=en
Decibel Meter	SPL	Windows	www.windowsphone.com/en-us/store/app/decibel-meter-free/6c7ae110-5a64-e011-81d2-78e7d1fa76f8
Decibel 10th	SPL	iOS	https://itunes.apple.com/us/app/decibel-10th/id448155923?mt=8
Noise Meter	SPL	Windows	www.windowsphone.com/en-us/store/app/noise-meter/c68f7156-9c86-480b-a44e-87ba6b5639f1
Sound Meter PRO	SPL	Android	https://play.google.com/store/apps/details?id=com.soundmeter.app&hl=en
TooLoud?	SPL	iOS	https://itunes.apple.com/us/app/tooloud-pro/id425137981?mt=8

Table 1: In this listing of sound level meter apps, “SPL” indicates that no weighting, “fast,” or “slow” options are indicated in the website information. Not all links will be available in all app stores outside of the US.

extend from simple noise level measurements to comprehensive acoustical/electroacoustical test platforms.

Measurement Shortcomings in Smartphone Devices

Before we examine the apps themselves, a few words are in order about the devices on which they run. As you know, smartphones come in three basic varieties: Apple iOS, Android, and Windows. (Blackberry devices seem to support only sound playback.) None of these systems was created with audio/acoustical testing foremost in mind. As a result, they all share certain weaknesses for testing applications. One of these is the user’s presumed expectation that (s)he can just use the internal microphone. Since the microphone systems were primarily designed for speech, most do not have good response below about 300 Hz. This weakness can be addressed by the use of a calibrated external test microphone and preamplifier, if the mobile device can accept an input from these.

Another weakness is the sampling frequency. Some models sample at 8 kHz, leading to a Nyquist frequency (theoretically highest usable frequency) of 4 kHz. Some later models sample at frequencies up to 48 kHz, allowing for measurements slightly beyond the top of the human hearing range.

The third weakness is latency—the “waiting time” involved in internal digital processing. Latency imposes very real limits on the implementation of any measurement system using digital processing.

Apple devices use a core audio system that enables very low latency. Android has been improving its audio core in recent years, but the basic OS architecture prohibits Android from achieving the level of latency performance that Apple provides.

Worldwide, approximately 80% of smartphones



Photo 2: The “Too Loud?” app has a simple display that can be seen from some distance.



Photo 3: The “Decibel Meter” screen has a speedometer-style sound pressure level (SPL) display.



Photo 4: The “Decibel 10th” screen uses a VU-meter-style display.



Photo 5: The “dB Volume Meter” shows an LED pictorial-style SPL indication.

and 60% of tablets use Android. Yet, because of the audio limitations, by far the greatest number of sound measurement apps are ported for iOS, with fewer for Android, and very few for Windows.

Sound Measurement Apps

The greatest number of sound-measurement-related apps is sound level meters. **Table 1** shows a representative listing. Since a comprehensive listing would be a moving target, with new apps being released all the time, this listing makes no claim of covering all available apps.

I should mention that the information in this table is the best that I could find on the Internet. Many apps do not have printed specifications, although some pages do link to videos. The videos may contain further specifications, but I do not consider a video to be a professional means of conveying specifications, so I did not examine them. **Photos 2–6** show screenshots of a few of



Photo 6: The “decibel” screen can show the values and the waveforms corresponding to the minimum, average, and maximum values of the noise.

these sound level meter apps. As you can see, the displays betray the developers’ concept of the target user, from cute and simple to professional and informative. The “Too Loud?” app shows sound levels with a color-coded display that turns redder as the sound gets louder (see **Photo 2**). The “Decibel Meter” shows the sound pressure level (SPL) in the upper part of the screen, and the noise waveform in the lower part (see **Photo 3**). The “Decibel 10th” display shows the sound level on a VU-meter-style display (see **Photo 4**), where “0 VU” corresponds to 75 dB, which is about the maximum safe level for an 8-hour exposure. The noise waveform appears at the top. The “dB Volume Meter” shows the maximum, minimum, peak, and “peak-hold” values on an LED-style graphic. As illustrated in the right-hand section of **Photo 5**, this app supports the use

of external microphones and can be appropriately calibrated for microphone use. The “decibel” screen shown in **Photo 6** displays the sound level’s minimum, average, and maximum values, showing both numerical values and waveforms. A-weighting is available in this app.

App Assessments

Probably because sound level meters are the most commonly used sound measurement devices, several studies have been published in which some of these devices have been assessed. Chucru Kardous and Peter Shaw of the National Institute for Occupational Safety and Health (NIOSH) published a paper in the *Journal of the Acoustical Society of America (JASA)* Express Letters, April, 2014, in which they concluded that:

Table 2: There are only a few real-time analyzer apps available for smartphones.

App Name	SPL Modes	RTA Modes	FFT Modes	Other Modes	Included Signal Generator	Platform	Website Information
Analyzer	SPL, LEQ, A, B, C, Z wtg, response times	RTA	FFT 1k–64k		white pink sine	iOS	http://dspmobile.de/2012/10/analyzer-2-0
Audio Analyzer	SPL	1/12–1 RTA	FFT 256–64k	Spectrogram scope		iOS	https://itunes.apple.com/us/app/audio-analyzer/id681184981
Etani RTA & RTA Pro		1/6–1 RTA			pink	iOS	http://www.etani.co.jp/eng/en-product
Spectrum Analyzer	SPL all, pk	1/3 RTA	FFT		white pink sine	iOS	https://itunes.apple.com/us/app/spectrum-analyzer-real-time/id490078884

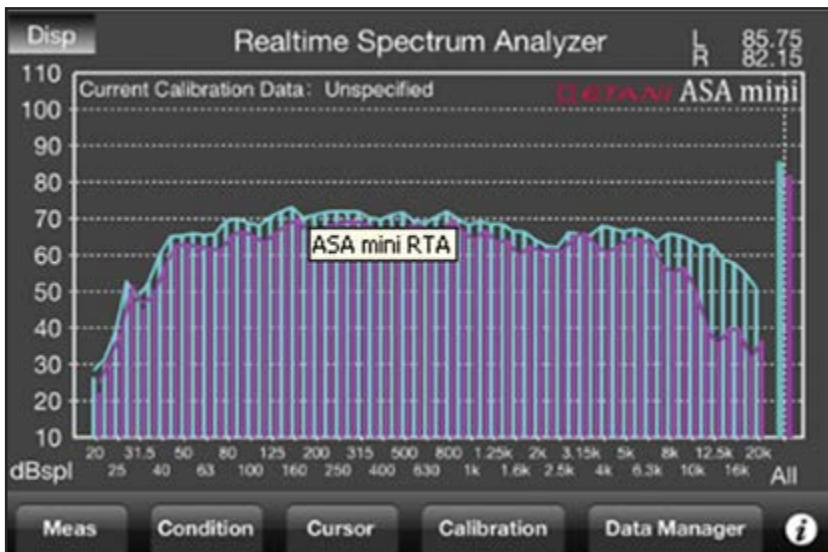


Photo 7: The Etani RTA and RTA Pro apps display decibels of sound pressure level (dB SPL) with resolutions from 1 to 1/6 octaves.

- “Certain measurement apps for Apple smartphones and tablets may be considered accurate and reliable to be used to assess occupational noise exposures.”
- “Android and Windows developers do not offer apps that meet the functionality needed for occupational noise assessments.”
- “Field measurement results may vary greatly due to the effect of temperature, humidity, long-term use, object interference, and overall stability of the microphone and electronics in these devices.”
- The same app used on different generations of iOS devices yielded somewhat different results.

The accuracy of these measurements was determined in comparison to a newly-calibrated Larsen-Davis model 831, Type 1 sound level meter using a model 2559 random-incidence microphone. Deviations varied from 0.07 dB for unweighted measurements to well over 10 dB for the worst case. It would seem to be mandatory that the accuracy of any smartphone/app combination be measured before one relies on it for critical measurements.

D. R. Nast, W. S. Speer, and C.G. Le Prell of the University of Florida compared five iOS-based sound level measurement apps with a Brüel & Kjaer 2250 SLM. They found significant variability in the accuracy, frequency response, and linearity of the apps tested, with differences in unweighted measurements reaching 15 dB, and weighting errors up to 9 dB. These authors concluded, “Unless the accuracy of an app has been determined, consumers should be hesitant to trust noise level estimates based solely on the app-based measurements.”

I should point out that in both studies, certain apps performed very well, so that users who invest the time and effort to evaluate research studies that have been performed before deciding on an app may achieve excellent results, within the unknown limits of environmental stability of the smartphone microphone and circuitry.

Real-Time Analyzer Apps

While many audio practitioners routinely use real-time analyzers (RTAs), there are far fewer RTA apps than sound level meter apps. **Table 2** shows some RTA apps.

Screenshots from two RTA apps are shown in **Photo 7** and **Photo 8**. The Etani RTA app shown in **Photo 7** has a built-in pink noise source and supports the use of an external microphone, for whose properties the app can be calibrated.

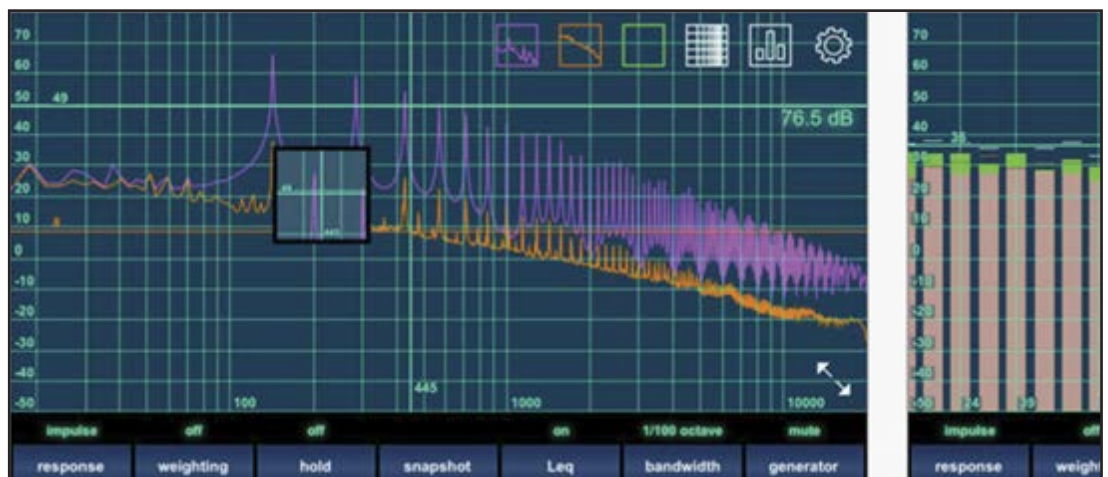


Photo 8: This photo shows both the real-time analyzer (RTA) and a partial Fast Fourier Transform (FFT) display from the Analyzer 2.5 app.

App Name	SPL Modes	RTA Modes	FFT Modes	Other Modes	Included Signal Generator	Platform	Website Information
Analyzer	SPL, LEQ, wtg, resp times	RTA	FFT 1k-64k		white/pink/sine	iOS	http://dspmobile.de/2012/10/analyzer-2-0
Audio Analyzer	SPL	1/12-1 RTA	FFT 256-64k	Spectrogram oscilloscope		iOS	https://itunes.apple.com/us/app/audio-analyzer/id681184981
Etani RTA & RTA Pro		1/6-1 RTA			pink	iOS	www.etani.co.jp/eng/en-product
Spectrum Analyzer	SPL all, pk	1/3 RTA	FFT		white/pink/sine	iOS	https://itunes.apple.com/us/app/spectrum-analyzer-real-time/id490078884
FFT Plot			FFT 16k			iOS	https://itunes.apple.com/us/app/fft-plot-real-time-sound-frequency/id569468015
Octave RTA			FFT 8k			iOS	https://itunes.apple.com/us/app/octave-rta-real-time-sound/id569156857
Audio Tools	[Detailed in <i>audioXpress</i> , December 2015]					iOS	www.studiosixdigital.com/
ClapIR			FFT	RT		iOS	https://stevetarzia.com/clapir.php
RoomSCOPE				RT	white/pink/lin&log chirp	iOS	www.faberacoustical.com/ios_apps/
iSTI	SPL A			STIPA	STIPA	iOS	www.embeddedacoustics.com/index.php/isti
SPLnFFT	SPL A, B, C, ITU-R 468; LEQ		FFT			iOS	https://itunes.apple.com/us/app/splnfft-noise-meter/id355396114
Spectrogram Pro			FFT			iOS	https://itunes.apple.com/us/app/spectrogram-pro-super-smooth/id415301721

Table 3: This is a list of Fast Fourier Transform (FFT) and special-purpose apps.

There are two versions of this app: the free one is designated as the Etani RTA, and the paid one is the Etani RTA Pro. Differences between the two versions are described on the website (www.etani.co.jp).

The Analyzer app (see **Photo 8**) includes the RTA and the Fast Fourier Transform (FFT) modes, and provides RTA bandwidths from 1/2000 octave to a full octave, as well as “fast” and “slow” response times. It also supports the use of an external microphone, and calibration can be performed manually or automatically.

Apps Intended for Professional Use

Table 3 lists some apps that are intended more for professional use than the ones discussed thus far. All of these apps except for iSTI and Spectrogram Pro provide FFT analysis, and three also display response in RTA fashion. Most of them have built-in signal generators—either pink noise or a choice of pink, white, or sine signals.

Photo 9 shows a screenshot from ClapIR, a unique app that uses a handclap as a signal source. This approach was discussed in a paper presented by Prem Seetharaman and Stephen P. Tarzia at the 132nd Audio Engineering Society (AES) Convention.

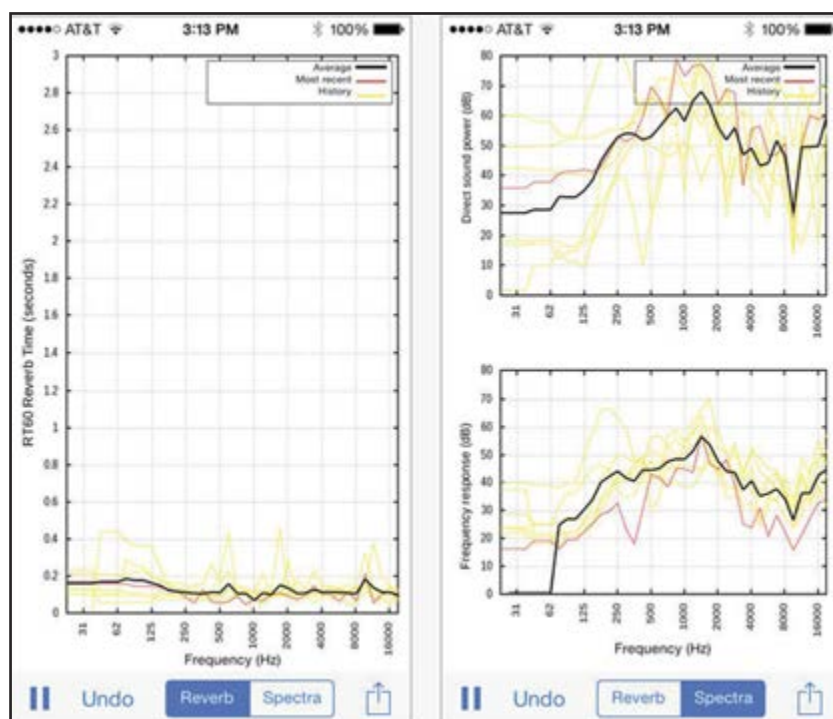


Photo 9: The ClapIR app uses a handclap as a signal source for RT and frequency response measurements in rooms.



Photo 10: SPLnFFT provides FFT analysis, plus many options for noise measurement.

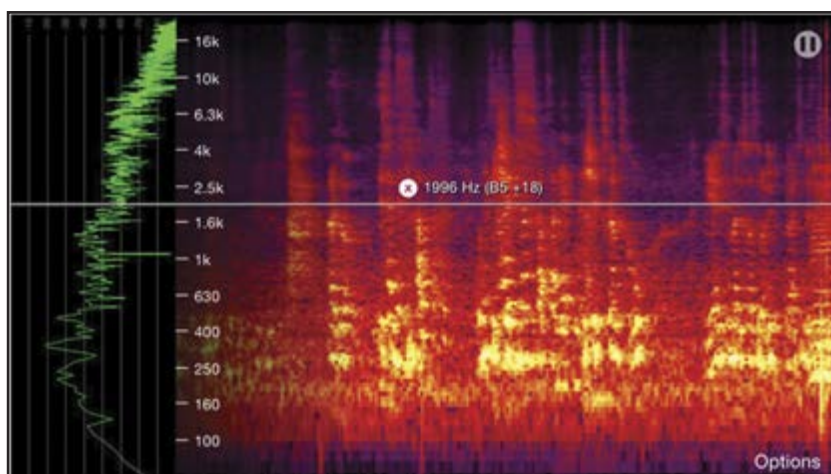


Photo 11: The Spectrogram Pro app displays its results as a spectrogram.

In the paper, they mention signal processing steps taken by the app to mitigate the effects of weak low-frequency content in handclaps. They do admit that results for frequencies below 300 Hz are not reliable, due to the lack of low-frequency energy, but point out that handclaps and balloon bursts (which suffer from the same weakness) have been used for a long time in measurement applications that do not require results at low frequencies.

Photo 10 shows two screens from SPLnFFT, which offers FFT analysis and a range of noise measurement capabilities. **Photo 11** shows a screenshot from Spectrogram Pro, another unique app that presents its display as a spectrogram. (The Audio Analyzer app can also display spectrograms.) This form of display has long been favored in certain acoustical disciplines such as musical acoustics, voice and speech acoustics, and hearing studies.

A shortcoming of most of these measurement apps is their inability to measure speech intelligibility. The iSTI app is specifically designed for speech intelligibility measurements, using the STIPA standard. It contains a signal generator providing the standard STIPA test signal, and can be used with an external microphone. It will also accept calibration information for this microphone.

Audio Tools App

Table 3 does not include any specific information for the Audio Tools app. The reason for this is that Audio Tools is actually a set of measurement apps designed by Andrew Smith of Studio Six Digital for the professional in audio and acoustics. It includes essentially the full functionality of a complete computer-based measurement platform. In addition, there is a group of modules available as separate in-app purchases, the most noteworthy of which is probably the Smaart Tools I module, which provides the single-channel measurement functions available in the Smaart measurement platform. In addition to the Audio Tools app, Studio Six Digital offers three test microphones and a stereo audio interface, comprising a complete test ensemble. In order to do justice to the Studio Six line, we will devote the December Sound Control article to it.

App Accuracy

In summary, iOS-based mobile device users can choose from a variety of apps for acoustical and electroacoustical measurements. Some of the apps can be quite accurate. However, users need to be careful as to which app they choose, and be aware that they will probably need to use an external microphone in many professional applications.

Resources

D.R. Nast, W.S. Speer, and C.G. Le Prell, PhD, "Accuracy of Sound Level Meter Applications for the iPhone®," University of Florida, http://c.ymcdn.com/sites/www.hearingconservation.org/resource/resmgr/imported/Nast_SLM%20APPS_Poster.pdf.

P. Seetharaman and S. Tarzia, "The Hand Clap as an Impulse Source for Measuring Room Acoustics," Convention Paper 8585, 132nd Audio Engineering Society (AES) Convention, 2012.

Sources

Type 2250 sound level meter
Brüel and Kjaer | www.bksv.com

Etani RTA

Etani Electronics Co., Ltd. | www.etani.co.jp

Model 831 Type 1 sound level meter

Larsen-Davis (a PCB Piezotronics Division) | www.larsondavis.com

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The SRX Plus Hybrid Amplifier (Part 1)

A Design for Electrostatic Headphones

Recently, I bought Stax SR-007 MKI headphones to accompany my old Stax SR-5s. The SR-5 requires a bias of around 230 V, whereas the SR-007 requires a bias of around 580 V. I wanted a good amplifier that would be able to drive both headphones so I built a tube amplifier with solid-state support for electrostatic headphones.

Photo 1: My SRX Plus Hybrid Amplifier is a tube amplifier that is able to drive my Stax electrostatic headphones.

By
James Lin
(United States)

Tubes are high voltage, low current devices, and a good choice for driving the capacitive load of electrostatic headphones. After all, that is what the input of another tube looks like. Since my stereo system is tubes driving Quad electrostatic speakers, it was natural to build a tube amplifier for my electrostatic headphones (see **Photo 1**).

Arguably, Kevin Gilmore is the preeminent designer of electrostatic headphone amps. His hybrid Blue Hawaii design (see Resources) has gained a reputation as the state-of-the-art amplifier for electrostatic headphones. He based this on the Stax SRM-T2, an exotic and complex “statement” amplifier.

The T2 was a highly complex, balanced, all DC coupled hybrid design with two 6DJ8 input tubes per channel, followed by two transistor stages, and an output stage using two triode-connected EL34 power tubes. The T2 could also output 630 V RMS at 1 kHz, had a slew rate in excess of 200 V/ μ sec, and consumed 200 W.

The T2’s circuit backbone is based on a two-stage design that Stax published in the 1970s—the SRX (see **Figure 1**). It is a wonderfully simple balanced design with only three tubes, one pot, four

capacitors, and 13 resistors per side. Conceptually, the T2 replaced the SRX’s coupling capacitors with the two direct-coupled transistor stages.

I chose the SRX as the basis for my amplifier, but I made several improvements that are detailed in this article—hence, I call my version the SRX Plus. This project involves potentially lethal voltages, and test equipment including a DC power supply and an accurate digital multimeter, so it is not the best project for a beginner.

The Amplifier Circuit

The Input Stage

The first stage is a cross-coupled cascode long-tailed phase splitter using 12AT7s. It has an open loop gain of 44 dB, and with the tubes dissipating 0.1 W each, they should have an excellent lifespan. A long-tailed phase splitter does not produce a perfectly symmetric output even with matched passive elements and perfectly matched tubes. There are several things that can be done to improve the output balance.

First, use a high gain circuit topology such as a cascode (see Resources). Second, cross-couple the cascodes (see Resources). Third, use a balance pot

between the two cathodes. All of these are used in the original Stax circuit. A fourth technique is to use as large a cathode tail resistor as possible.

The advantages of this input circuit are its high gain and simplicity. The disadvantages are its puny current and high output impedance.

A good rule of thumb is that the current delivered to a load should be less than a fifth the standing current of the circuit. As a worst case, 16 V peak at 20 kHz into the output tube's Miller capacitance of 80 pF requires a peak current of 0.16 mA, a quarter of the designed idle current.

Since real musical signals are less demanding, the current drive should be adequate. But, the high output impedance into the Miller capacitance of the output stage causes the open loop gain to be -3 dB at 11 kHz and -7 dB at 22 kHz.

Overall feedback is needed to flatten the frequency response. It also stabilizes the gain and decreases the distortion. Decreasing the plate resistors to 250 k Ω slightly increases the idle current and marginally increases the rolloff point to just over 12 kHz at the cost of 1 dB in gain.

A 100 k Ω resistor between ground and the negative junction of the two 100 μ F electrolytic capacitors across the 5 k Ω adjustment pot was added to bias the capacitors and improve their linearity. Finally, the coupling capacitor between input and output is a relatively low value, so I doubled it, pushing the open loop rolloff corner down to 3 Hz.

Input Stage Tubes

A cascode amplifier's gain is largely determined by the lower tube's transconductance. The 12AT7 has one of the highest transconductances of commonly used audio tubes, but it does have a significant degree of second harmonic distortion. However in a push-pull circuit such as this, even-order harmonic distortion will tend to be canceled out, and the cross-coupling of the cascodes will also help cancel out even-order harmonic distortion from the lower tubes.

The upper tube only marginally affects the gain and the output impedance. However, the upper section tube does most of the voltage swing. Although substituting different tube types in a circuit is generally not a good idea, substituting a 12AX7 or a 5751 *for the upper section only* results in negligible change in gain and output impedance, and these tubes have less second harmonic distortion than 12AT7s.

Output Stage

The output differential amplifier has a 24 dB open loop gain, and is direct-coupled to the headphones—output plate voltages within 10 to 15 V of zero are acceptable (see Resources). The closed loop gain is 54 dB, so the frequency response should be flat between about 1 Hz and 50 kHz.

In the original circuit, the positive rail voltage is lower than the negative rail voltage to allow the output plates to sit near 0 V with respect to

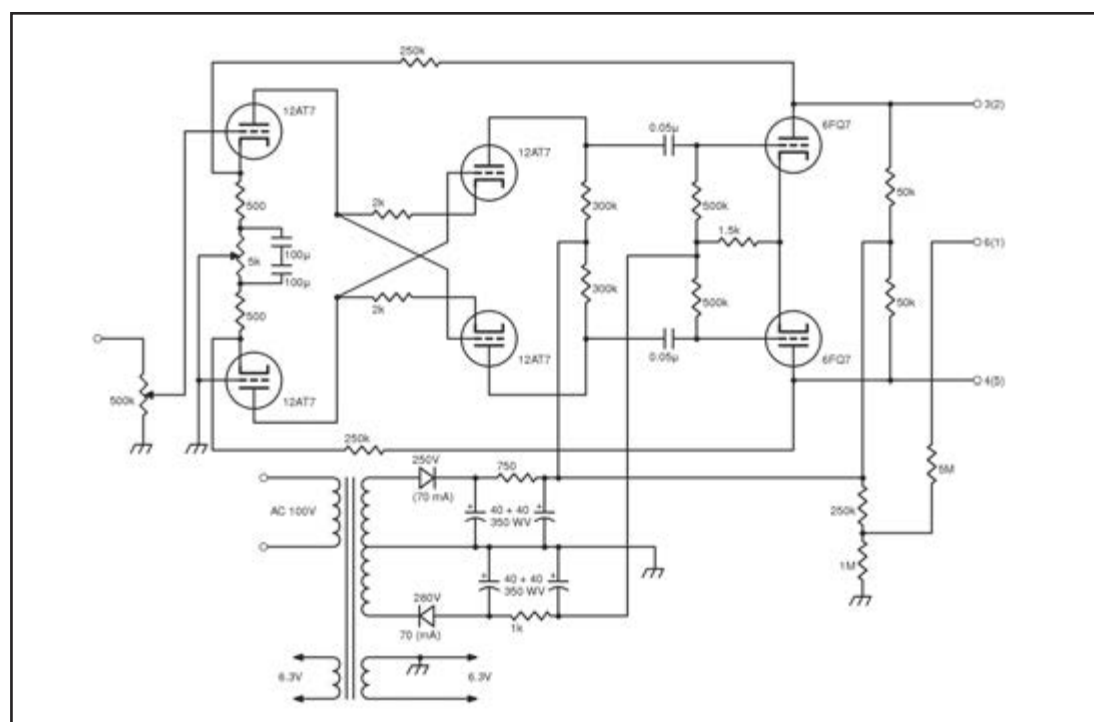
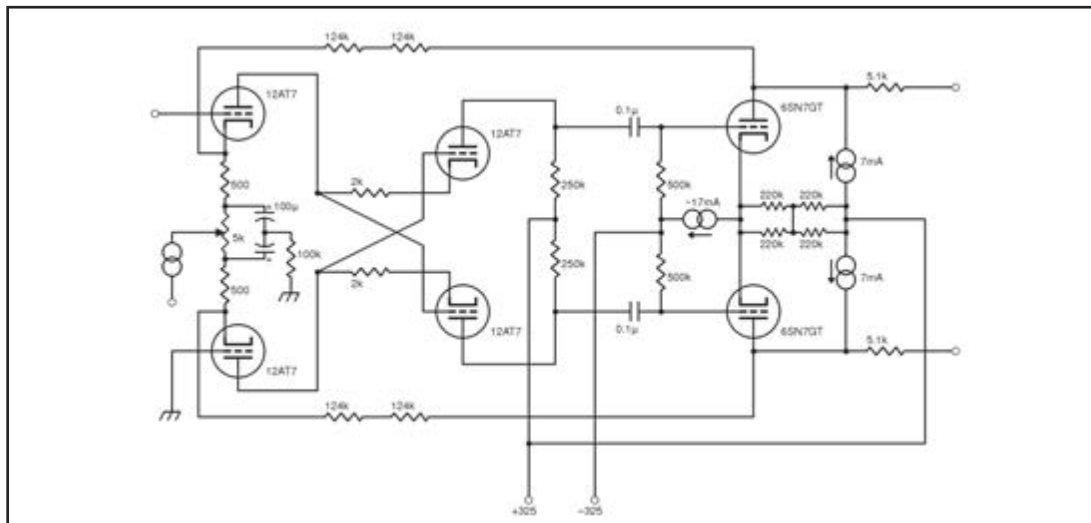


Figure 1: The backbone of my tube amplifier is based on the SRX, a two-stage design that Stax published in the 1970s.



Figure 2: I made a few modifications to the original SRX design. This is a simplified schematic showing those changes.



ground, and the output tubes run about 5 mA per side.

With better output tubes we can increase the rail voltages, with a potential peak-to-peak voltage swing of around 1,200 V. Worst case with a full voltage signal at 20 kHz into the 100 pF load, this would require a peak current of about 8 mA (4 mA per side). Music signal measurements reported by Peter Baxandall and Nelson Pass suggest that a full power bandwidth of around 6 kHz is adequate, so the peak current with music would be about 2.4 mA. Thus, the ability to provide full power to 20 kHz should provide an adequate margin—provided that all the signal current goes to driving the headphones.

Output Tubes

The 6CG7/6FQ7 tube used in the original circuit is marginal in both voltage limit and power dissipation, but there are not many tubes that have a higher maximum plate voltage rating.

6SN7GT tubes have identical electrical specifications to the 6CG7, except for a marginally higher grid-to-plate capacitance. The A and B suffix versions are higher voltage and power versions developed in part for tube TV use (see Resources) so they have a significantly higher maximums: a plate voltage of 450 V, a peak pulse plate voltage of 1,500 V, and power handling of 7.5 W for both plates combined. They are one of the most linear dual triode tubes, a significant consideration when wide voltage swings are required.

What about the triode-connected EL-34 output tubes used in the Blue Hawaii and the Stax T2? They can handle much more power and voltage, but the voltage gain in triode connection is 6 dB lower. Also, the grid-to-plate capacitance of a triode connected EL34 is 10 pF, so the closed-loop frequency response barely makes it to 20 kHz before rolling off.

Running tubes at or near maximum ratings will significantly reduce reliability and tube life, so we will design for 70% to 80% of maximum voltages and 60% to 70% of maximum power dissipation.

Modifications

Current Loads/Sinks

The original circuit was designed about 45 years ago, and there are several things that can be done to improve its performance, specifically, using current sources and sinks. A current source or sink produces a current that is independent of voltage—an ideal current source has an infinite dynamic resistance. By convention, current sources that connect to a positive supply are called sources, while those that connect to a negative power supply are called current sinks.

Allen Wright of Vacuum State Electronics cited the use of current sources as one of the major innovations that came with solid-state circuits. Stax has used current sources in its solid-state headphone amplifier designs since the 1980s.

A current sink replacing a tail resistor in a long-tailed phase splitter or differential amplifier isolates it from the negative power supply rail and improves the balance. On the anode side, it isolates the anode from the positive power supply, decreases distortion, and maximizes gain.

For the input cascode phase splitter, a cathode current sink should improve its balance. However, replacing the anode resistors with a current load will increase the output impedance and lower the open loop roll off frequency.

The output stage is a different story. A cathode current sink will improve the balance of the output stage, but the use of a current source/load on the anode side will also improve the function of the output stage.

About the Author

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

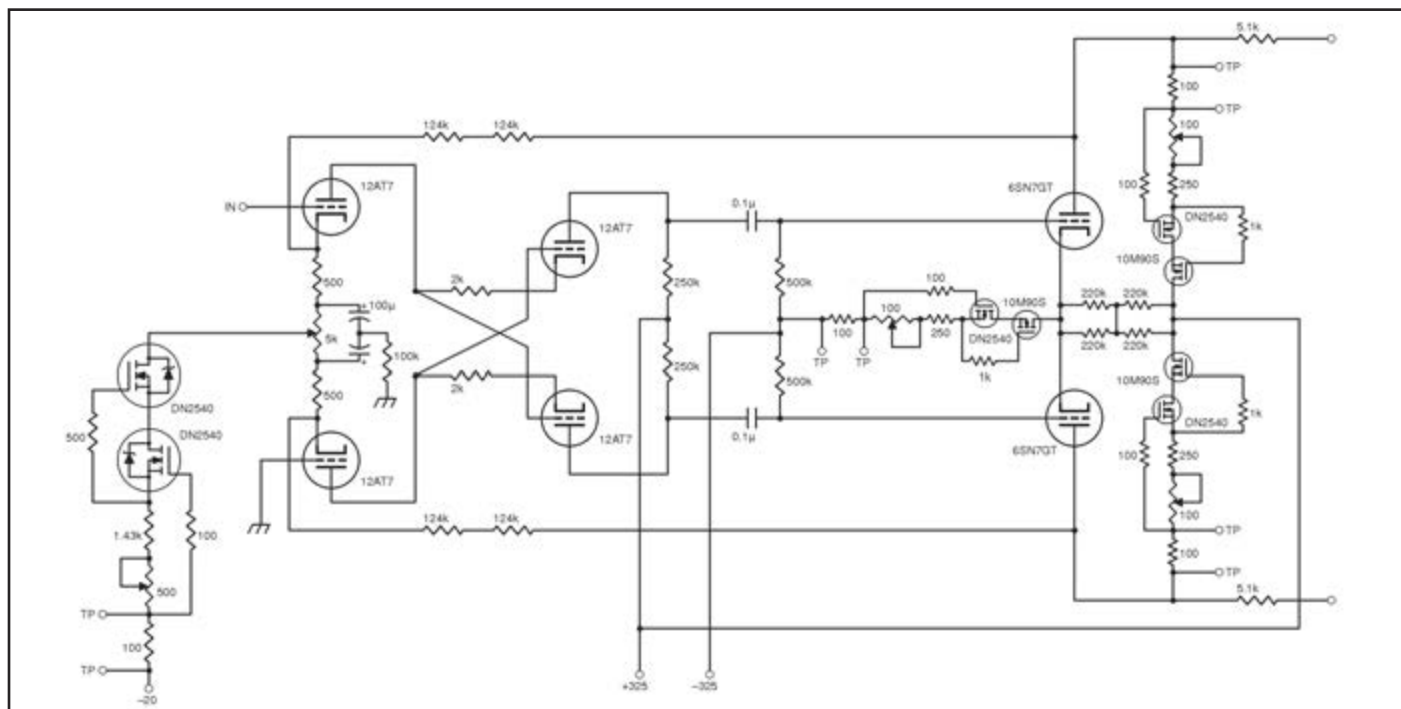


Figure 3: The complete amplifier schematic is detailed.

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The anode resistor sets the standing voltage and the current in the output tube. For the amplifier to produce its maximum voltage swing, it must also use its maximum current swing in the anode resistors. As soon as headphones are attached, the output tube's current is divided between driving the headphones (the desired result) and driving the anode resistor. On the bright side, since electrostatic headphones are so much higher in impedance than the anode resistor, the impedance of the two in parallel is nearly the same as the anode resistor alone, so very little voltage headroom is lost. On the dark side, nearly all the power is going into the anode resistor, so we are wasting most of our effort.

A perfect current load decouples the voltage and current functions, so all the signal current produced by the output tubes goes to drive the headphones. Also, since current variations in the tube increase distortion, the relatively low impedance anode

resistor is by far the most significant cause of distortion. Thus, a current load simultaneously decreases distortion and allows the output tube current to be used much more efficiently.

A passive anode resistor in an electrostatic headphone amp will typically have a value of 25 to 60 k Ω . A simple 10M90S current source, which is used in a number of electrostatic amp designs, has a typical dynamic resistance of 150 to 200 k Ω , a useful improvement.

In addition, having current sinks and current sources at both the cathodes and anodes of the output tubes isolates them from the power supply and the other channel, stabilizing circuit parameters and improving channel separation. Finally, the use of current loads also increases the output stage's open loop gain up to 26 dB.

With these changes, we have the simplified schematic shown in **Figure 2**, which looks similar to that of the original SRX. Anticipated sonic improvements include enhanced fidelity of subtle sonic detail, more stable imaging and tonality, decreased distortion, and improved bass performance—particularly at high levels.

While tubes can be used as current sources, the wide voltage swings in the output means each tube current source requires its own filament supply. Furthermore, a single triode tube doesn't make a good current source (see Resources). A pentode performs much better, but still needs a filament supply. A high voltage MOSFET makes a good, simple current source with the sole disadvantage of needing a heatsink.

For constant current sources and loads, Walt Jung's articles in past issues of *audioXpress* and Gary Pimm's DIY website are excellent sources of information. Using the 400 V Supertex DN2540 or the 900 V IXYS 10M90S as single-current sources, measured dynamic impedances ranged from 200 k Ω to over 1 M Ω at lower audio frequencies, rolling off at higher frequencies like a small capacitor between 3 to 130 pF. The numbers might sound high, but the first amplifier stage has anode resistors of 250 to 300 k Ω . The typical electrostatic headphone has a capacitance of around 100 pF, with an impedance of about 80 M Ω at 20 Hz, 1.6 M Ω at 1 kHz, and 80 k Ω at 20 kHz.

Cascoding increases dynamic impedance into the 10 to 3,000 M Ω range with equivalent capacitances in the 0.2 to 2 pF range. Regardless of the exact measurements, cascoded current sources clearly offer superb performance, achieving dynamic resistances far exceeding the other resistances in the active circuit, or the headphone load. The price for the improved performance of the cascode pair over a single device is one field-effect transistor

Resources

E. Barbour, "6SN7: Driver of Choice," *Vacuum Tube Valley*, Issue 11, Spring 1999.

D. R. Birt, "Self-Balancing Push-Pull Circuits," *Wireless World*, May 1960 and June 1960.

K. Gilmore, "Blue Hawaii Amplifier for Stax Omega II Headphones," HeadWize DIY Projects Library, headwize.com.

L. B. Hedge, "Cascode A.F. Amplifier," *Wireless World*, June 1956.

Walt Jung, "Performance Current Regulators Revisited," *audioXpress*, April, 2009.

—, "Sources 101: Audio Current Regulator Tests for High Performance (Part 1): Basics of Operation," *audioXpress*, April 2007.

—, "Sources 101: Audio Current Regulator Tests for High Performance (Part 2): Precise High Current/Voltage Operation," *audioXpress*, May 2007.

A. Kimmel, "The Mu Stage," *Glass Audio*, Volume 5, 1993.

G. Pimm, "CCS Performance Measurements," www.pimmlabs.com/web/ccs_performance/htm.

Stax SRX amplifier, www.bonavolta.ch/hobby/en/audio/hdgsta.htm.

A. Wright, *The Tube Preamp Cookbook*, Second Edition, Vacuum State Electronics, 1997.

Sources

IXCP10M90S current source
IXYS Corp. | www.ixys.com

DN2540N5-G MOSFET
Microchip Technology, Inc. | www.microchip.com

STP19NM50N MOSFET
STMicroelectronics | www.st.com

DN2540 N-Channel MOSFET
Supertex, Inc. | www.supertex.com

(FET), one resistor, and a few additional volts.

Pimm has designed more complex designs using up to three times more parts, and in some cases tubes and/or battery biasing, but the additional improvement in dynamic impedance is a relatively modest 3-to-20-fold. For the input stage, the standing current is 1.3 mA, putting the upper tube plates halfway between ground and B+. If you are using the original 300 k Ω plate resistors for the input stage the current should be set for 1.1 mA.

For the output stage, the standing plate current load is 7 mA per tube section. With ± 325 V supplies and the plates at ground voltage, each plate current load would have a quiescent dissipation of about 2.3 W. High-voltage aluminum oxide insulators were used to mount the TO220 devices to the heatsink. A 10M90S TO220 is used for the upper device and a DN2540 for the lower device. I took one cascaded current source and ran the voltage up to 400 V without any measurable change in the output, corresponding to a DC dynamic resistance greater than 160 M Ω . Even at 20 Hz, nearly 90% of the audio current is going to the headphones, and by 100 Hz at least 99% is going to the headphones. Less than 5 μ A of current will drive the output stage current loads to clipping, leaving 99.9% of the standing current available to drive the headphones.

Each 6SN7GTA/B plate dissipates 2.2 W each, or 4.4 W total, which is less than 60% of the rated maximum for both plates combined, and should result in a long tube life.

Output Stage Resistor String

When we combine current loads with a current sink, we run into a problem. The current produced by the anode loads must exactly equal the current absorbed by the cathode current sink, which is impossible.

The solution is to set the cathode sink current approximately 3 mA higher than the combined currents of the anode loads and then add a resistor string between B+ and the cathode sink to drain the excess current. This resistor string compromises the dynamic resistance of the cathode current source, but at 220 k Ω , it is still much higher than the original 1.5 k Ω cathode resistor. The cathode current sink is adjustable via a multiterm trimmer. This has the additional benefit of allowing the output plate voltages to be set to zero without needing an offset in the positive and the negative power supplies.

Finally, 5.1 k Ω resistors on each output protect the headphones in case of tube failure. The complete amplifier schematic is shown in **Figure 3**.

More to Come

In the second part of this article, we will detail the power supply, the construction, and the test results of this hybrid amplifier design. We will also include the parts lists for the amplifier, the raw power supply, the high-voltage regulator, and the bias supply—everything you will need to build this hybrid amplifier. 

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Build the PhaseBuster!



Photo 1: The PhaseBuster creates interesting and useful effects for recovering and preserving hidden or missing elements in old and contemporary recordings.

The directionality, the localization, and the ambiance of a stereo audio performance are defined by amplitude and phase differences between the left and right channels. Equalization offers some degree of adjustment over apparent levels of individual instruments or vocals in an existing recording. This simple project provides the means of manipulating the phase between stereo channels.

By
James B. Wood
(United States)

The goal of the PhaseBuster project (see **Photo 1**) was fourfold. I wanted to create a monaural mix from stereo without losing audible components; to process a stereo program for better mono compatibility, one that will sum to L+R mono with minimum degradation; to create a pseudo-stereo program from a monaural source; and to “spatialize” a stereo source, increasing the apparent soundstage width.

The Mono Compatibility Issue

Two incidents separated by more than 20 years inspired this project. The first involved an audio diagnostics product that graphically displayed the dynamic range of broadcast audio. To simplify analysis, left and right channels were combined with simple L+R summing. But the absolute energy of the sum did not agree with sound pressure level (SPL) measurements in an actual listening environment.

A respected audio guru of the era suggested displacing the phase of one channel by 90° to avoid cancellations and buildups that commonly occur when stereo recordings are mixed-down to mono. This “quadrature summing” did indeed provide a more honest representation of bass and vocal

energy in pop music. Vocal and bass tracks are nearly always recorded individually and panned to the center of the soundstage. When summed to L+R mono, these components can come across as louder than when heard in stereo.

The more recent incentive to achieve better monaural compatibility grew out of my own frustration in mixing-down certain orchestral recordings to mono. A longtime friend uses historic “Decca Tree” microphone techniques (see Ron Tipton’s article series on Realistic Sound Recording in *audioXpress*) to make lifelike binaural recordings for a local symphony. I, on the other hand, take arcane delight in cutting lacquer discs with a vintage lathe that predates stereo records.

In mixing my friend’s .wav files to mono, certain instruments, along with most of the venue ambiance, just completely disappeared! While this phenomenon can also be noted to a lesser extent in multi-tracked, artificial-stereo pop music, my friend’s live recordings would not properly mix-down to mono. So now I had an immediate desire to turn stereo into more realistic mono, and figured that imparting the 90° phase offset between channels, as suggested years ago, might just do the trick.

As an adjunct to this idea, why not use this same quadrature technique to “correct” stereo material so that it would inherently mix to better mono when the two channels were combined with simple L+R addition at a later time? Although we don’t commonly reduce stereo to mono intentionally, today’s FM receivers incorporate circuitry that automatically blends stereo to mono to minimize noise as the carrier fades and to reduce distortion from multipath effects as they skew and corrupt the multiplex signal. Of course, the true imaging of a stereo program with an intentional 90° offset between the two channels might suffer at the expense of maintaining better mono compatibility, but this could prove the lesser of two evils under some circumstances. Worth a try anyway, thus, this became the second goal of the PhaseBuster project.

Stereo Synthesis

The third project goal was the creation of a pseudo-stereo effect from monaural material. Various methods have been employed over the years to synthesize stereo from mono sources. In the early days of stereo discs, record companies used crude spectral reassignment to convert their existing monaural catalog to the new format.

“Re-channeled for stereo” was often as simple as sliding the bass to the left and the treble to the right. This was refined with more frequency divisions, assigning alternate bands to the left and the right channels. Bass frequencies were kept in the mono domain, however, mastering engineers channeled everything below 200 Hz into mono. Not only is bass more fully reproduced by two loudspeakers driven together, but avoiding low-frequency vertical groove modulation ensured better tracking of stereo records and less playback distortion.

The early years of stereo TV sound saw a comeback for stereo synthesis. Broadcasters employed purpose-built stereo synthesizers to enhance mono audio feeds. RCA even proposed building similar circuitry into the TV set itself. Both strategies split the audio spectrum into filtered frequency bands and assigned them to the left and the right loudspeakers. This does create a pseudo-stereo image of sorts and helps impart a certain amount of ambience and realism to program audio.

When using the 90° offset idea for mix-down to mono years ago, the reverse was tried as well—that is, creating a pseudo-stereo program from a monaural source with a quadrature offset between channels.

This capability has been incorporated into the PhaseBuster, along with more aggressive techniques discussed later in the article.

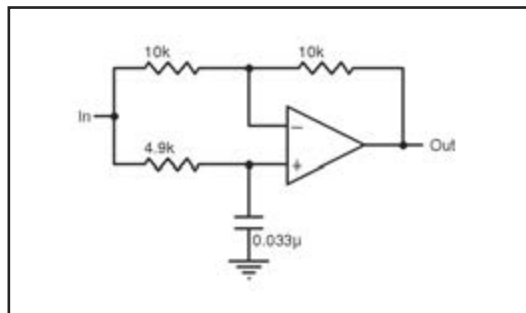


Figure 1: The final all-pass section is the single op-amp stage, IC9B.

Stereo Enhancement

The broadcast industry is generally held responsible for the “loudness wars” that began with AM radio in the 1960s. Despite claims to the contrary, not only are loudness wars still with us, but they have metastasized from top-40 programming to nearly all AM/FM radio formats, television commercials, and even CD and music download releases.

Compressing the dynamic range of music to just a few decibels is bad enough. But once the effort to make radio programming louder reached a point of diminishing return, a certain radio Program Director asked his engineer, “Well, if we can’t be louder, can we be wider?”

Most broadcast audio processing systems now offer a means of expanding (widening) the stereo image in addition to increasing the RMS power (loudness). They can even expand the soundfield beyond the perimeter defined by the placement of the loudspeakers themselves, giving the illusion of surround sound from just two speakers. Current methods employ dynamic expansion of the L-R stereo difference signal, and frequently make use of phase manipulation as well. The result can

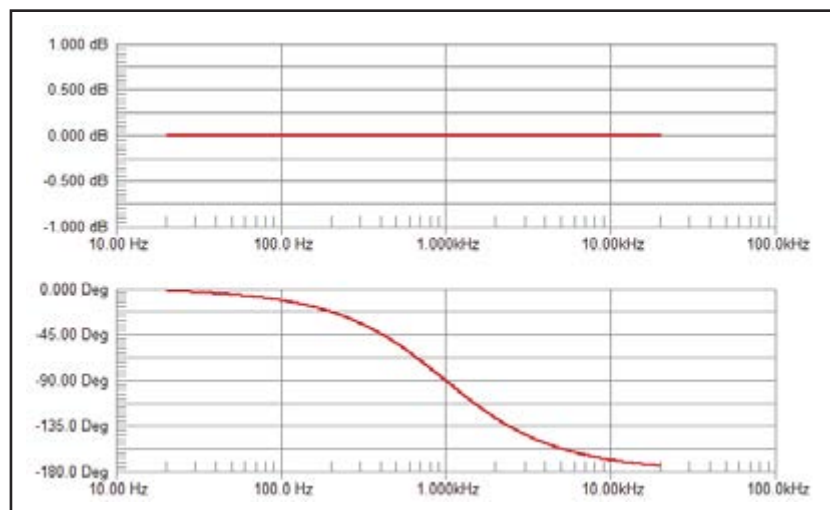


Figure 2: The circuit’s amplitude and phase response is shown.

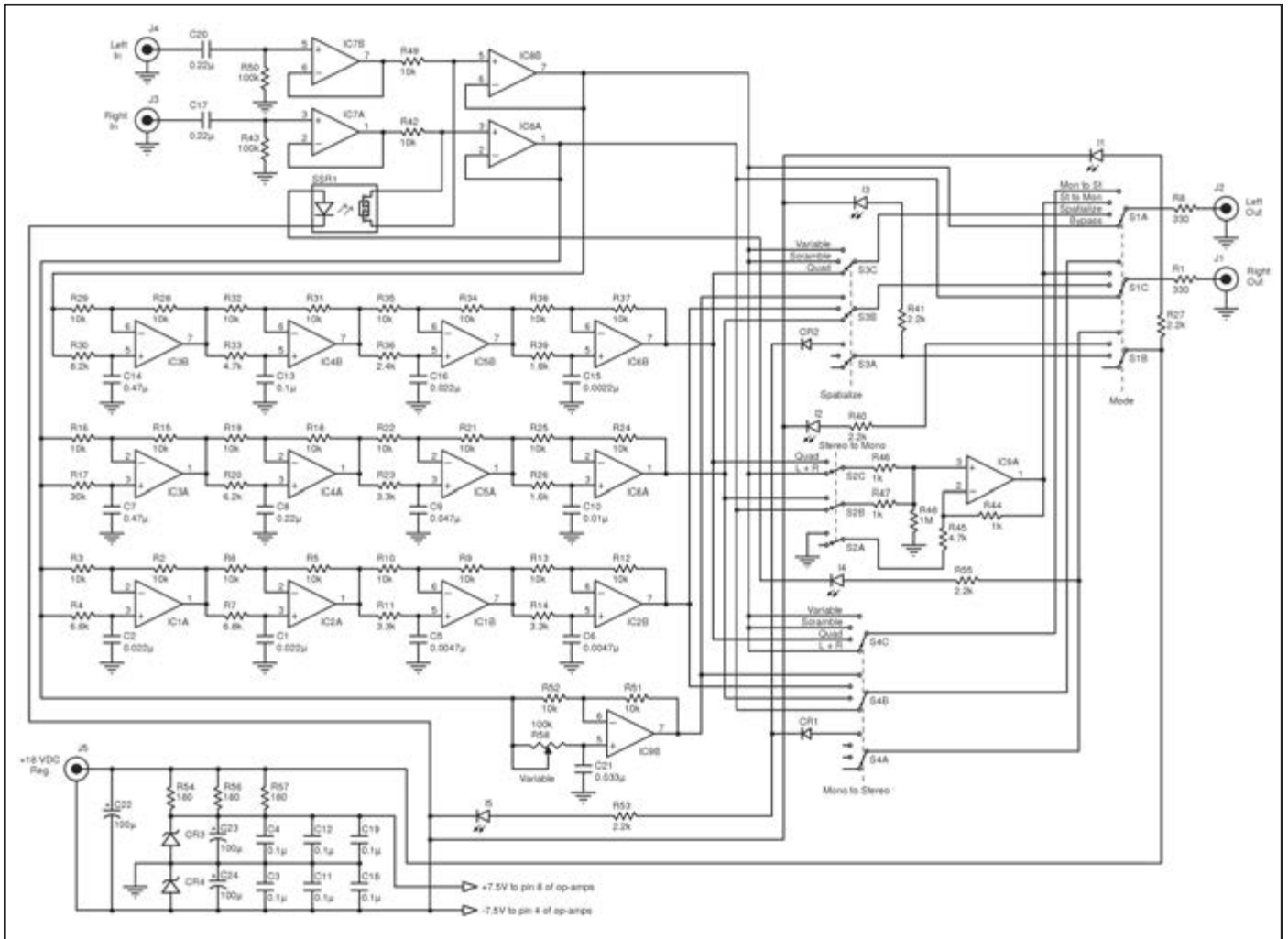


Figure 3: I used this schematic to build the PhaseBuster.

make a performance decidedly more spacious, sometimes too spacious, with an obvious “hole in the middle” and a sense that the performance is somewhat disjointed. Sadly, just as with the pursuit of loudness, this practice has also migrated into music recording and production.

Nevertheless, there are instances where a bit of stereo enhancement can improve a performance. Recordings from an earlier, more conservative era are a case in point. Consequently, a “spatializing” utility was incorporated into the PhaseBuster design.

PhaseBuster Building Blocks

Figure 1 shows a typical first-order all-pass filter section. This single op-amp stage exhibits non-inverting, unity gain for DC and low audio frequencies. At very high frequencies, the capacitor essentially grounds the non-inverting input, and the stage flips the signal by 180°, again with unity gain for the values shown. Between these extremes the output will exhibit a frequency-dependent phase offset from the input signal, but will maintain a flat amplitude response. **Figure 2** shows the amplitude and phase response of this circuit. The values yield a 90° phase lag at 1 kHz.

The PhaseBuster incorporates several such stages. The frequency-dependent phase offset shown in **Figure 2** is a simple matter, but our requirement for a wideband quadrature phase relationship (90° at all frequencies) takes a bit more doing.

Figure 3 shows the PhaseBuster’s schematic. At the center-left of the diagram, note the arrays of cascaded all-pass filter stages. There are three

About the Author

James Wood is Founder and Board Chairman of Inovonics, Inc., a 43-year manufacturer of audio and radio frequency products for broadcast. Jim has been active in audio, radio frequency, and instrumentation circuit design and construction for more than 60 years. He holds several patents in the field and is a Life Member of the Audio Engineering Society (AES) and a Life Senior Member of the Institute of Electrical and Electronics Engineers (IEEE). A perennial student of scientific history, Jim takes particular delight in developing practical applications for arcane technologies.

banks of four cascaded filters. The top pair (IC3, IC4, IC5, and IC6) is of initial interest here.

The amplitude and the phase characteristics of this pair of filter banks are shown together in **Figure 4**. With the specified components, four cascaded all-pass sections will provide a frequency-dependent phase offset that follows a reasonably straight line from 20 Hz to 20 kHz, a phase lag slope of about 180° per decade. But the phase difference between the outputs of the two banks is a constant 90° at any and all frequencies in the audible range, a true full-bandwidth quadrature relationship.

The third filter bank (IC1/IC2) is similar to the others, but has a steeper phase shift characteristic that turns over (begins) at a higher frequency (see **Figure 5**). Note that this filter has little effect on bass frequencies, but it does subject the midrange and top-end to a radical phase-lag slope. This steep filter provides the phase scrambling function discussed later in the text.

The final all-pass section is the single op-amp stage, IC9B. This is essentially the circuit shown in **Figure 1**, except that resistor R58, which determines the phase-shift turnover frequency, is a variable control. This allows the 90° phase-shift frequency to be set anywhere between 50 Hz and 3 kHz.

Operating Modes

The four phase-shift networks are the heart of the PhaseBuster. Left and right line inputs are hard-wired to these networks, and network outputs are selectively switched to PhaseBuster line outputs depending on the operating mode. Front-panel controls are four rotary switches and one potentiometer. Spare switch contacts light the front-panel LEDs beside each knob to indicate which controls are active in any particular mode.

The following information includes a description of the signal path, an explanation of what the circuit is doing in each case, notes on what you might expect to hear, and ideas for further experimentation.

Bypass

Bypass directly routes the PhaseBuster input to its output. However, power must be applied as there are buffer stages in the signal path. The PhaseBuster is designed to work at average, “zero-VU” signal levels in the -10 dBv (-8 dBu) range common to consumer audio gear. The internal and the output clipping level is about +15 dBu.

Spatialize

This is the stereo-enhancement mode for expanding the stereo image or soundstage. There

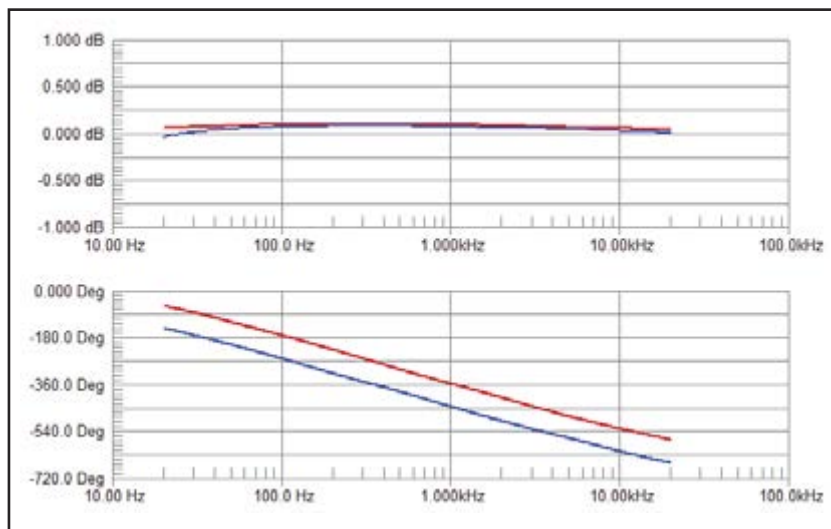


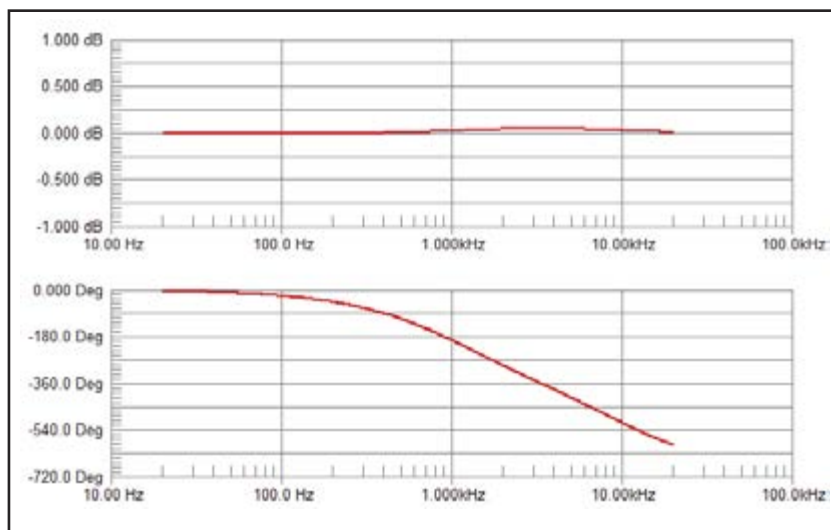
Figure 4: The amplitude and the phase characteristics of the quadrature pair of filter banks are shown together.

are three Spatialize sub-modes—quadrature, scrambled, and variable. Quadrature routes the left and the right inputs through the two all-pass filter banks that impart a 90° phase offset between the two channels. This will have the effect of spreading the virtual center-channel (L=R) program components between the two loudspeakers. Vocal and bass tracks will “spread,” no longer centered midway between the two speakers. Music may exhibit somewhat nebulous directionality, but images will be stable. In fact, you may not notice much of a difference unless you are sitting in the listening sweet spot. Someone in the next room probably won’t be aware of a change at all.

Quadrature is also the means to alter a stereo program so that it will make a better mono mix later on. This concept is further explored in the stereo to mono discussion.

Scrambled passes the left channel directly through, but routes the right channel through the

Figure 5: The third filter bank (IC1/IC2) is similar to the others, but has a steeper phase shift characteristic that begins at a higher frequency.



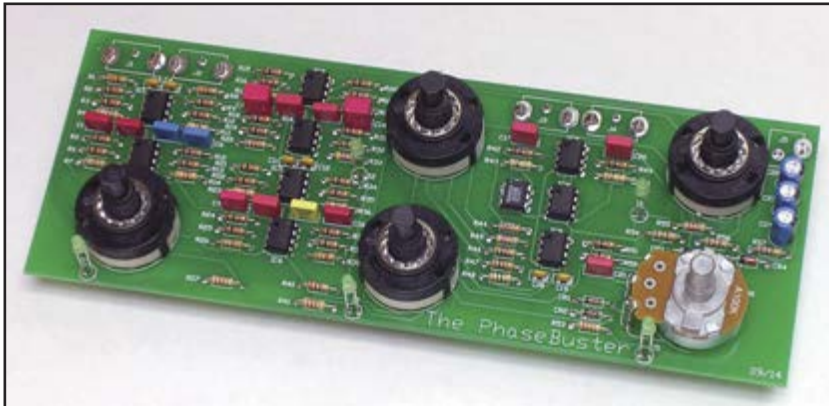
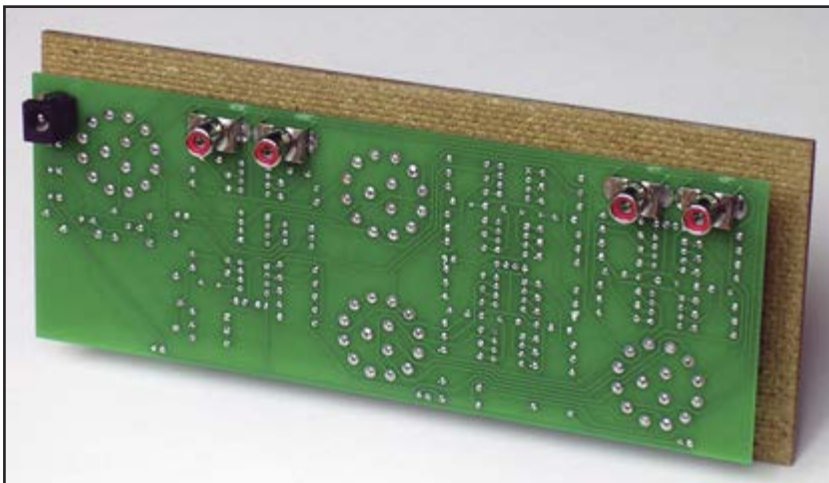


Photo 2: PhaseBuster circuitry is contained on a single circuit board.

all-pass filter bank with the steeper phase slope shown in **Figure 5**. Because this filter bank doesn't do much below 200 Hz, this mode will have little effect on bass frequencies. But, program components in higher registers will tend to move within the soundstage as a function of their pitch. For example, a piccolo going up and down the scale might move from left to right and back again, because this filter bank can put those frequencies through a complete 360° phase rotation. Vocal sibilants may likewise seem to detach from the singer.

Variable also passes the left channel directly through, and routes the right channel through the adjustable all-pass stage. Now, the variable control knob will be active to adjust the 90° phase lag

Photo 3: Almost all components, including the rotary switches, are on one side of the board, the input, the output, and the DC power connectors are on the back side.



frequency. Turned all the way down (out), the all-pass section is essentially out of the circuit and you should not hear a difference. As you turn the knob clockwise, however, the right channel will maintain a more normal (not inverted) phase below the indicated frequency and a more out-of-phase condition above it (see **Figure 2**).

Unless you set a particular music passage into a short repeat loop, you may not hear much difference between variable and scrambled as you switch between the two. The variable position may lend better stability to the soundstage, as the phase of high frequencies never lags by more than 180°.

There is the one important consideration concerning both the scrambled and the variable modes. The resultant enhanced stereo is for exclusive use in a listening environment. Audio processed this way cannot be subsequently mixed-down to monaural without creating gaping holes in the mono frequency response. These modes are just for listening, not dubbing.

Stereo to Mono

This PhaseBuster mode is helpful in creating a mono-compatible mix-down from a stereo pair of channels. The mono mix in this mode may be taken from either the left or the right PhaseBuster outputs, they both are fed.

L+R combines the stereo input channels in traditional summing style as the name implies. Quadrature sum, on the other hand, inserts the 90° phase offset between the stereo channels prior to summing them, and introduces a 2 dB gain in the output path as well. Because program components common to both channels lean toward a more out-of-phase than an in-phase condition in quadrature summation, this bit of gain helps match the apparent loudness of most program material between the two switch positions.

I found that the quadrature sum position restored at least some of the musical content that went missing when the two stereo channels were combined in simple L+R fashion. Listening to the resulting mono, quadrature sum returned ambience to the performance, sounding much the way mono LPs did in simpler times.

Mono to Stereo

This mode is the reverse of the stereo to mono mode, and has four sub-modes. In all cases the PhaseBuster left and right inputs are first combined in a straightforward L+R manner to make sure that the material is monaural to start with, even if the source is stereo. This L+R addition is performed after the input buffering stages, rather than simply

Project Files

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

Resource

R. Tipton, "Surround Sound—The Next Generation: Dolby Laboratories," *audioXpress*, February 2015.

tying the inputs together. (The stiff, low impedance voltage source outputs of audio equipment ought never to be fed into one another.) If you do feed just one input with your mono source, expect a 6 dB loss through the system. Instead, apply the mono source to both inputs to preserve overall unity gain.

L=R does not provide any stereo synthesis, it merely routes the monaural signal to both the left and the right PhaseBuster outputs. Mono played this way ought to sound as if it's coming from a point midway between your speakers with no "soundstage" at all.

In quadrature, the mono source drives both all-pass filter banks shown in **Figure 4**. The two banks are then fed to the left and right PhaseBuster outputs, yielding two sources that have a 90° phase shift between them at all frequencies.

As stated in the earlier L=R discussion, a single channel of audio fed to two loudspeakers should appear to emanate from a point midway between them. If you then reverse the wiring to one speaker, so that the two are driven out of phase, the sound should then seem to separately come from each of the speakers, the classic hole-in-the-middle effect.

But if you drive your two speakers with signals that are in quadrature (90° apart at all frequencies), the effect is a veritable "wall of sound" between the speakers. However, this is hardly pseudo-stereo. There is no sense of directionality or localization at all. Nonetheless, this effect may find some utility. For example, a simple PA system with speakers on each side at the front of a hall might give an illusion of better coverage (or not!). It is an interesting effect deserving of further experimentation.

Scrambled routes the mono source directly to the PhaseBuster left-channel output, and through the steep all-pass filter bank (see **Figure 5**) to the right-channel output. Unlike the spatialize/scrambled mode, which added an enhancement effect to a program that was already stereo, this mode synthesizes pseudo-stereo from a mono source. But unlike the traditional stereo synthesizers discussed earlier, all audio frequencies are present in both output channels. There is no frequency band division.

Instead, the apparent location of a sound in this artificial soundfield will depend on pitch, and the resulting phase relationship between channels at a given frequency. The localization effects are being generated and perceived in the listening environment, they are not amplitude differences between the signals that feed the two loudspeakers. Both speakers are being driven with full-range audio, not interleaved, defined bands of frequencies.

"Busy" might best describe this mode's sonic



Photo 4: The PhaseBuster's front panel was made from 1/8" Masonite.

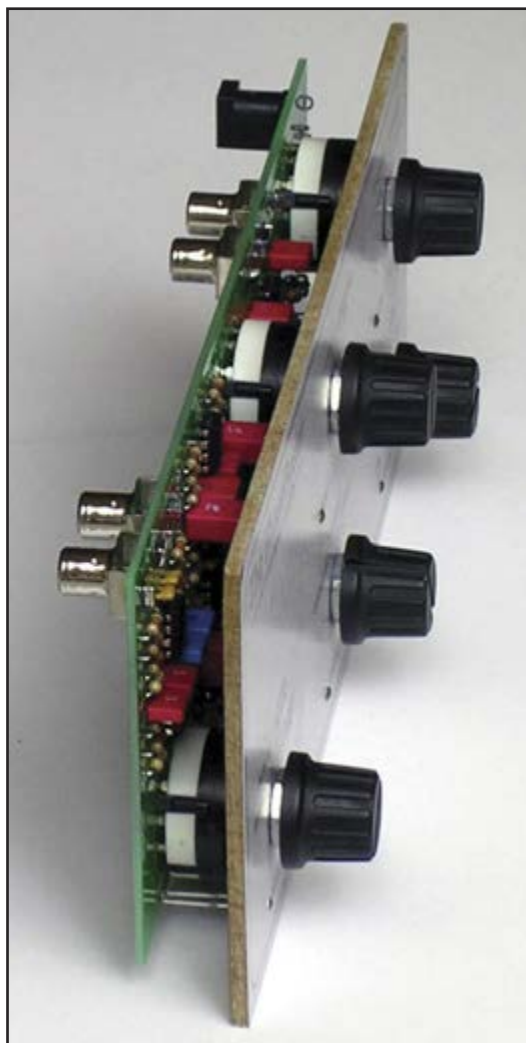


Photo 5: The board and panel "sandwich" is held together by the switches and potentiometer.



effect. Vocals and instruments alike will shift about within the soundstage, the strings, and the vocal sibilants, in particular. This soundfield can be impressive, but perhaps confusing and annoying to music purists.

The variable mode of stereo synthesis is probably

the most interesting, as the circuitry involved is so exceedingly simple. Once again, the left PhaseBuster input is routed directly to the left output, but this time just the adjustable, single op-amp phase-shift stage is inserted in the right channel signal path.

Assume the variable knob is set at 1 kHz, giving

PhaseBuster Parts List

Designation	Qty.	Description	Mouser Part No.
Capacitors			
C1, C2, C16	3	Film, 0.022 μ F, 5%	505-MKS20.022/63/5
C3, C4, C11, C12, C18, C19	6	Ceramic, 0.1 μ F	581-AR155C104K4R
C5, C6	2	Film, 0.0047 μ F, 5%	505-FKS24700/100/5
C7, C14	2	Film, 0.47 μ F, 5%	505-MKS20.47/63/5
C8, C17, C20	3	Film, 0.22 μ F, 5%	505-MKS20.22/63/5
C9	1	Film, 0.047 μ F, 5%	505-MKS20.047/63/5
C10	1	Film, 0.01 μ F, 5%	505-MKS20.01/63/5
C13	1	Film, 0.1 μ F, 5%	505-MKS2.1/63/5
C15	1	Film, 0.0022 μ F, 5%	505-FKS22200/100/5
C21	1	Film, 0.033 μ F, 5%	505-MKS20.033/250/5
C22, C23, C24	3	Electrolytic, 100 μ F, 25 V	140-REA101M1EBK0611P
Diodes			
CR1, CR2	2	Silicon, 1N4151	78-1N4151
CR3, CR4	2	Zener diode, 7.5 V, 1.3 W	8-BZX85B7V5-TAP
Miscellaneous			
I1 - I5	5	LED, T1 Green	859-LTL-4231N
IC1 - IC9	9	8-pin DIP Op-Amp, TL072	595-TL072CP
J1 - J4	4	RCA Jack, PC-mount	161-R239A1-E
J5 (small pin)*	1	DC Power Connection, 5.5 mm $\times$ 2.1 mm	806-KLDVHCX-0202A
J5 (large pin)*	1	DC Power Connection, 5.5 mm $\times$ 2.5 mm	806-KLDVHCX-0202B
S1 - S4	4	Rotary Switch, three pole, four position	105-SR2612F-34-21RN
SSR1	1	Solid-State Relay, IXYS LCA110	849-LCA110
Knob	5	0.25" diameter shaft	450-2035-GRX
Power Supply*			
Wall-Wart	1	18 V switch-mode regulated supply, 100 mA minimum (*select the proper J5 mating connector to match)	
Resistors			
R1, R8	2	0.25 W, Carbon Film, 330 Ω	660-CF1/4C331J
R2, R3, R5, R6, R9, R10, R12, R13, R15, R16, R18, R19, R21, R22, R24, R25, R28, R29, R31, R32, R34, R35, R37, R38, R42, R49, R51, R52	28	0.25 W, Carbon Film, 10 k Ω , 5%	660-CF1/4C103J
R4, R7	2	0.25 W, Carbon Film, 6.8 k Ω , 5%	660-CF1/4C682J
R11, R14, R23	3	0.25 W, Carbon Film, 3.3 k Ω , 5%	660-CF1/4C332J
R17	1	0.25 W, Carbon Film, 30 k Ω , 5%	660-CF1/4C303J
R20	1	0.25 W, Carbon Film, 6.2 k Ω , 5%	660-CF1/4C622J
R26	1	0.25 W, Carbon Film, 1.6 k Ω , 5%	660-CF1/4C162J
R27, R40, R41, R53, R55	5	0.25 W, Carbon Film, 2.2 k Ω , 5%	660-CF1/4C222J
R30	1	0.25 W, Carbon Film, 8.2 k Ω , 5%	660-CF1/4C822J
R33, R45	2	0.25 W, Carbon Film, 4.7 k Ω , 5%	660-CF1/4C472J
R36	1	0.25 W, Carbon Film, 2.4 k Ω , 5%	660-CF1/4C242J
R39	1	0.25 W, Carbon Film, 1.8 k Ω , 5%	660-CF1/4C182J
R43, R50	2	0.25 W, Carbon Film, 100 k Ω , 5%	660-CF1/4C104J
R44, R46, R47	3	0.25 W, Carbon Film, 1 k Ω , 5%	660-CF1/4C102J
R48	1	0.25 W, Carbon Film, 1 M Ω , 5%	660-CF1/4C105J
R54, R56, R57	3	0.25 W, Carbon Film, 180 Ω , 5%	660-CF1/4C181J
R58	1	Potentiometer, 100 k Ω audio taper	31VJ501-F3

the phase response shown in **Figure 2**. Low audio frequencies will remain in phase between the two output channels, which will feed system speakers in tandem for efficient bass reproduction. As program components reach the 90° phase shift frequency, the two channels will be in quadrature (and remain reasonably in quadrature over a modest range), so that midrange sounds will appear to fill the soundstage between the two speakers. High frequency material will approach, perhaps even reach an antiphase relationship (180° apart), and appear to come from the two speakers independently. The overall effect is a wide sound with a pleasant degree of soundstage dynamics, but not too confusing.

The best pseudo-stereo performance setting seems to be in the 1 kHz to 3 kHz range of the variable control. At too low a frequency setting the bass is weak as much of the audio spectrum is completely out of phase. Set too high, there is little stereo effect at all.

This single-stage stereo simulation circuit proved valuable in another project. The single op-amp phase-shift stage was easily added to a portable record player, providing a bit of pseudo-stereo “pizzazz” for old 78 and 45 RPM dance records played in a meeting hall.

Other Circuitry

There is really not much to the PhaseBuster beyond the all-pass filter sections. The left and right channel inputs are buffered by IC7A and IC7B, which provide high input impedance for minimal loading of whatever feeds the unit.

SSR1 is a solid-state relay that sums the inputs to monaural for the mono-to-stereo mode. Although not specifically intended for audio switching, these clever little chips are capable of handling high-level AC signals without introducing significant distortion. Their only drawback is that they are a bit slow to turn on and off (milliseconds), not a concern here.

The PhaseBuster is powered by a regulated 18 V “wall-wart” switch-mode power supply and further regulated by a pair of Zener diodes to create a bipolar 7.5 V supply for the op-amps. The 18 V source has to float for this to work, thus, the PhaseBuster can’t share ground-referenced power from an existing supply.

Construction

PhaseBuster circuitry is contained on a single circuit board. **Photo 2** shows the board before it’s attached to the front panel.

The only hand wiring consists of bare-wire jumpers between the variable pot and the board.

Almost all the components, including the rotary switches, are on one side of the board, the input, output and DC power connectors are on the back side (see **Photo 3**).

You will need to set the rotational stops for the proper number of rotary switch positions in each case. Tiny numbers stamped into the switch body help with this. Also, break off the anti-rotation pegs on the switches and the pot before mounting the front panel.

The PhaseBuster’s front panel shown in **Photo 4** was made from 1/8” Masonite. Stick a paper print of the front panel artwork onto your preferred panel material, trim the panel to the crop marks, and then centerpunch and drill holes for the controls and LEDs.

Check the crop marks with a ruler first to make sure your printer doesn’t need a programmed-in fudge factor to get the dimensions spot-on. The overall panel dimension is 8.5” × 3.47”, which is a 2 Rack Unit (RU) panel height.

The rotary switches require 3/8” holes. The variable pot needs a 5/16” one. LEDs poke through 1/8” openings. Once the panel is cut and drilled, strip off the paper overlay and replace it with a clean copy. Regular bond paper works well, and you can cover it with a clear plastic shelf liner for a durable surface.

The board and panel “sandwich” is held together by the switches and pot (see **Photo 5**). Solder the LEDs and the rotary pot leads after the panel is mounted as lead length is determined by the panel-to-board spacing (see **Photo 6**). The PhaseBuster can go into a box of your own design or it can be rack-mounted if you make the front panel 19” wide.

The Parts List shows Mouser (www.mouser.com) part numbers for everything except the 18 V power supply. Check online for the best deal on one of those, just make sure that it’s an 18 V regulated switch-mode supply, not the heavier transformer-type that has no regulation. Supplies in this voltage range should come with a plug that has a 5.5 mm outer diameter and either a 2.1 mm or 2.5 mm inner diameter. Jacks for both sizes are included in the Parts List, use the one that mates with your supply. 

Author’s Note: The bare circuit board for the PhaseBuster is available from me for \$20, postage-paid within the US. Specifics are included in the Supplementary Material. Happy phasing!



Photo 6: The LEDs and the rotary potentiometer leads are soldered after the panel is mounted because the lead length is determined by the panel-to-board spacing.



Acoustic Enthusiast Builds His Business and Speakers One Step at a Time



Simon Ashton uses a two-pronged business approach to make his high-end creations available to people.

An Interview with Simon Ashton, Founder of AudioSmile

By
Shannon Becker
(United States)

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

SIMON ASHTON: I always wanted to be a rock star and loved music from a young age, but as I grew older I realized that perhaps that wasn't the most realistic career choice! So, I decided to study sound recording, primarily at the University of Westminster (London, England). During this time, as well as learning about recording techniques, my personal interests in both hi-fi, acoustics, and electronics grew. Rather than work for someone else, I started building acoustic room treatments, offering electronic modifications and later my own speaker designs. My parents have both been self-employed so I suppose this path seemed quite natural to me. I've been slowly building a reputation for high-quality designs and good craftsmanship.

SHANNON: How did you become interested in audio?

SIMON: My parents both loved music since university, so there was always lots of vinyl around me as a kid. My dad moved over to CDs and I got most of the vinyl! One day I snagged the stylus of my turntable on the sleeve of my dressing gown. My dad took me shopping for a new one. But I remember that upon first hearing it with the new stylus, I was really upset because my music didn't sound right

anymore! I think this is when I got interested in the equipment itself and the impact it has on the enjoyment of music.

SHANNON: What was the inspiration behind the Little British Monitor?

SIMON: The inspiration actually came when my wife asked for some music in the kitchen. We had a nice system in the living room and she has been involved with music all her life so she knows what good sound is. The average iPod dock was not going to cut it. I built some small full-range speakers into wooden boxes and hooked them up to a good amp. She loved them and so did many of our friends, so this got me thinking about making a product along the same lines. Over a period of a year or so as I worked on getting the best sound from a small box, the design became a true high-end monitor only miniaturized.

SHANNON: How popular are desktop speakers?

SIMON: To be honest, I think a lot of people will use the Little British Monitor (LBM) as a small speaker anywhere in the home not just on a desk. I call them a desktop speaker because of the size, but I can imagine them on either side of a TV, on a bookshelf, or in the kitchen. Many people love music and will



appreciate good sound when they hear it, but very few people will give house space to a traditional audiophile speaker. I also think the LBM will appeal to musicians who have small home studios. It is possible to record great sound in a small studio but you need good monitors to mix it and those tend to be quite bulky. The LBM is ideal in this situation.

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

SIMON: The electronics were the biggest challenge. I originally thought it would be easy enough to implement a digital amp IC, but upon testing, I found they all have distortion in the midrange and treble that would make the high-quality tweeter in the LBM rather pointless. I had to use proven Class-AB technology, but that not only needed a bigger heatsink, it also needed a dual rail power supply. External switch-mode power supply (SMPS) bricks are all single-rail so I needed to design a rail splitting circuit—one that didn't have a high output impedance nor damage power supply rejection ration (PSRR).

The next challenge was getting Bluetooth to provide good sound. The digital chips in Bluetooth systems output a lot of switching noise on the power lines. This tends to feed into the audio stuff. I've heard this blipping type noise on many commercial Bluetooth speakers. It took a lot of trial and testing to get everything onboard working at its best. I'm proud to say the Bluetooth system in the LBM delivers true hi-fi sound, especially from aptX-enabled sources.

SHANNON: You chose to fund the Little British Monitor through Kickstarter. Why go the crowdfunding route?

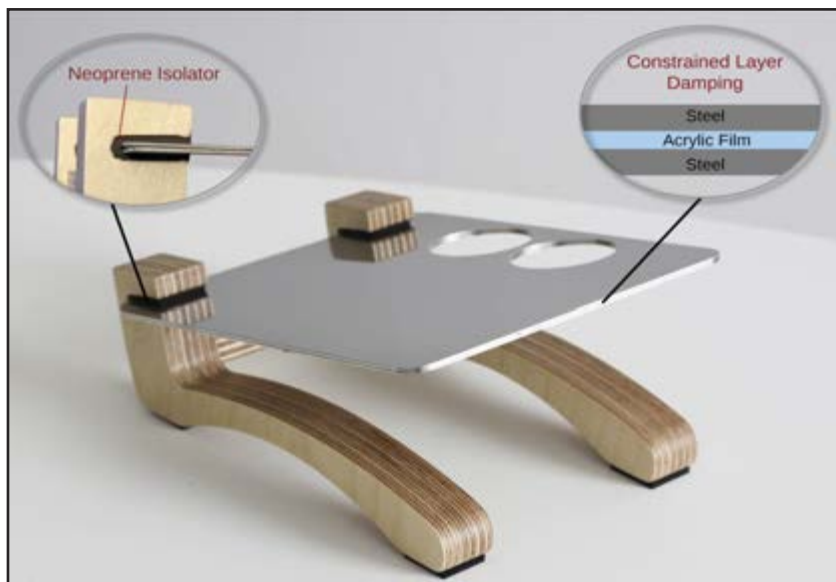
SIMON: I had seen other successful audio projects on Kickstarter. The support they gathered was impressive. My usual process would be to design a product, get reviews, and then build to order. This is great for costly equipment, which sells in small numbers, but the LBM needed to be different.

The LBM was intended to be affordable to any music lover, not only keen audio enthusiasts. That meant building in quantity and funding by pre-order

The Little British Monitor (LBM) is shown with an oak veneer without a grille (a) and with a grille (b).



This pair of Little British Monitors take up very little space when placed on a desktop.



The Little British Monitor's stand has a unique construction. AudioSmile also makes the "Little Feet" available separately as a vibration damping solution. The plate dissipates structural energy and acts as a sink.



The Kensai was AudioSmile's first speaker released as a product. The speaker shown here is the second generation version, the Kensai MKII, which combines a isoplanar ribbon tweeter with a 4" woofer powered by an advanced neodymium motor.

on Kickstarter was the clear way to do this.

I also like the way that Kickstarter is putting consumers in direct contact with the designers. Traditional mass production is impersonal and usually driven only by profit rather than a passion to make good and useful stuff. If a person took the traditional route of going to a bank for a huge loan and jumping into production, there is massive risk involved. What if nobody likes the idea or the price is wrong and it doesn't sell?

Crowdfunding cuts out the bankers and the salesmen. It makes things personal again. A crowdfunding project will let you know if people like your idea enough to buy it before production starts. Also, if things progress, you are funded by the people who actually want the invention. No bank manager would understand and support you in the way the crowdfunding community does. All of this goes hand-in-hand with the rise of small but accurate manufacturing driven by 3-D printing and CNC tech. I predict the return of local production.

SHANNON: You had included an additional funding option to release your design as open source hardware? Why did you want to do this?

SIMON: Open source hardware (OSH) is becoming more popular with the interest in Raspberry Pi and Arduino-type devices. People like to program software and OSH lets you move to the physical world with whatever you want to do. This type of thinking is giving rise to a generation that like to make stuff and want to customize things exactly the way they want.

For example, somebody might like to make the LBM automatically switch on and off when an audio signal is present. Without the full details of the circuit design this would require the reverse engineering of the circuit to integrate your own adjustments. OSH puts all the details in your hands so whatever you think of is pretty much possible. Sadly, the LBM project did not reach the OSH stretch goal, but I'm certain the DIY kit will still appeal to makers and hackers. I may release the LBM as OSH later down the line.

SHANNON: On your website, audiosmile.com, you show two other products, the Kensai MKII and the Supertweeter MKII. When did you design these speakers? Were they also funded via crowdfunding?

SIMON: The Kensai was my first speaker released as a product. I followed that with a supertweeter and now the MKII of each. I funded these myself

to create the final product and keep demo units in stock. I build more when customers place an order. Selling direct means I can produce a better product for the cost and not keeping stock of what are quite expensive designs is only sensible for a sole trader such as myself. Customers like the fact that I send photos of the build process for their specific order as I work on it.

SHANNON: What's next for your company, AudioSmile? Do you have other projects in the works?

SIMON: I hope to use Kickstarter again. It was an experience that restores a little faith in the human nature of sharing and kindness! That said, I believe there is a limit to the monetary outlay people will make on Kickstarter, especially for audio products.

I have a large three-way speaker ready to go on market when I have the time to invest. This is a design for real enthusiasts, competing with larger Harbeth and Spendor models. I hope to put it with select dealers so people can listen to it first.



The AudioSmile Supertweeter supplements narrowing dispersion of existing speakers in the high-frequency range. It features two low-distortion isoplanar ribbon units made from conductively etched lightweight kapton film suspended between a neodymium push-pull magnet system.

I believe there is room for both approaches in my business depending on the type of market to which the design appeals. 



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Passive Offenders

If an amplifier is intended to be classified as high end, each component should be top quality. And although most false sounds occur in the transistors or tubes, the passive components could also be at fault because sometimes they, too, exhibit unwelcome behaviors. This also applies to capacitors. However, by selecting the best components and the appropriate circuit design, top-notch designers manage to avoid these pitfalls.

By
**Helmuth
Lemme**

(Germany)

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

Ideal components can only be found in textbooks. The reality is something far different. Many of them deviate from the physical models and not just the active and amplifying ones but also the passive components, which are actually considered unlikely to produce such problems. When it comes to resistors, they are mostly harmless. At least the metal film types can be used without any problems. But with inductive components (e.g., transformers) it can be very different. Typical problems can involve series resistance, winding capacitance, eddy current losses, leakage inductance, and magnetic saturation and distortions of the core.

Normally, when deviating from the ideal capacitor, you only think of the series resistance and the inductance. However, the impedance drops inversely proportional to the frequency, it reaches a minimum with the resonance frequency and then increases again. This can be a problem in radio frequency (RF) applications. However, the effect is widely known and can be eliminated using the appropriate measures.

Capacitors

There are wound and front-contact capacitors. The wound capacitors consist of two electrically

conductive foils and two insulating foils. Inevitably, one conductive layer will be the last one on the outside of the winding. The layers are connected by wires. The winding as well as the wire connections create inductances that may have an influence in the high-frequency range. When looking at the front-contact capacitors, you should imagine the layers as two combs slid inside each other. The teeth are the layers featuring insulation. The ridge of the comb forms the contacts. In this way, the inductive part of the capacitor is drastically reduced. Electrolytic capacitors made of aluminum are generally manufactured as wraps. In this case, the anode is connected several times to reduce the inductance and to increase the current load rating.

If you use capacitors in an amplifier, they can collect environmental interferences. Therefore, in tube technology, the capacitor's outer layer was originally marked in the schematics. If modern-day capacitors are wrapped, they will also have their outside layer marked, usually in the form of a thick line. If the capacitor is installed so that the outside layer points to the more noncritical side of a circuit, it is to be considered a detail improvement in device engineering. Therefore, the outside layer belongs to the preceding tube's anode since this is the point

with the lower impedance and not vice-versa. If the capacitors have one grounded connection, this should be the outside layer. When using polarized electrolytic capacitors, the polarity needs to be observed. But with bipolar electrolytic capacitors, the mounting position does not matter.

During the heyday of tubes, there were still wound capacitors with three connectors. The ready capacitor was additionally equipped with an insulating shielding foil on the outside, which was connected to ground using the third connector. This shielded the capacitor and reduced the amplifier's interference voltage. Today, wound capacitors with three connectors are not available.

Bad Dielectrics Cause Harmonic Distortions

Some capacitors can also generate nonlinear distortions. Designers who are unaware of that could easily opt for the wrong capacitors (see **Photo 1**). And, the sonic result will not be optimal. However, by selecting the appropriate type and the matching circuit dimensions, the distortions can be kept low.

Although this has been the case for years, it is a timely topic as we attempt to make our devices as compact as possible. It also means that every component must be as small and light as possible. So, the capacitors produced today need to show high capacitances within small overall dimensions.

To achieve this, two things must be done in parallel. First, the insulating layer needs to be as thin as possible. Second, the dielectric constant must be increased to its limit. However, the associated extremely high electric field strengths in the material often lead to complications. Hi-fi amplifiers with different capacitors, but otherwise identical components, may show different sound characteristics. Specific types are deemed to have "good" sound, whereas others sound "bad."

Electrolytic Capacitors

The dielectric of electrolytic capacitors is a few microns of thin oxide film on the positive electrode, which is called the anode. Its surface is roughened by etching, which increases it by the factor of about 100. To maintain the oxide film, these capacitors must only be operated with the correct polarity. Minor punctures heal through the chemical process. In case of a wrong polarity, the film is electrolytically dismantled and the electrolytic capacitor is destroyed. Most manufacturers play it safe, often saying that a wrong polarity of up to 0.8 V is not dangerous. Values of up to 1.5 V may occur for a duration of up to one second and should not be repeated.



Photo 1: Bad capacitors or nonlinear distortions generated by capacitors are a common source of problems.

Practical tests show that it doesn't seem to be that critical. However, with the wrong polarity, a DC current flows through the electrolytic capacitor. When electrolytic capacitors are connect with the wrong polarity, it is dangerous with higher voltages reaching up to 30 V and higher capacitances. In filter chains or when used as electrolytic coupling capacitors, there are current-limiting resistors. Depending on the capacitance and the voltage, a higher current results, which can make the resistor blow up. In case of weak or no current limiting, the electrolytic capacitor produces gas, swells, becomes hot, and can explode. With high capacitances and applications featuring higher voltages, especially for large cup-shaped electrolytic capacitors, overpressure disconnectors are used inside. Once the electrolytic capacitor generates gas and swells, a wire on the inside with a defined breaking point disconnects the power. This prevents the device from further damage or fire.

The bipolar electrolytic capacitors are secure against wrong polarity. Initially, these were opposing series connections made of two polarized ones placed inside a single housing. In more modern versions, both electrodes are equipped with an oxide

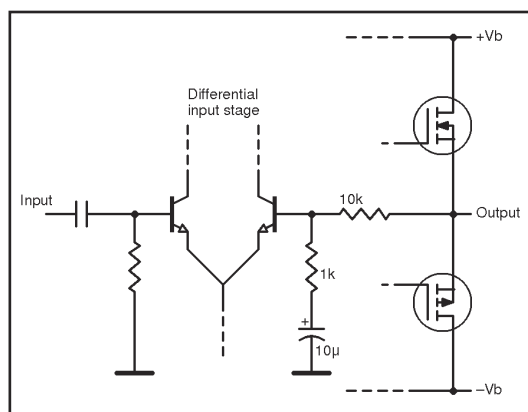


Figure 1: This circuit principle of a modern hi-fi power amplifier features a voltage divider in the negative feedback path. The electrolytic capacitor can generate distortions in the low-frequency range.

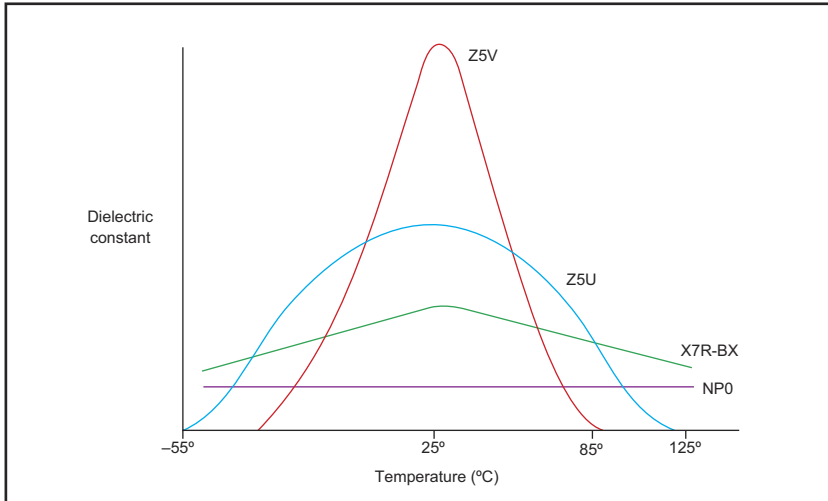


Figure 2: Temperature dependency on ϵ_r for different ceramic dielectrics. The maximum is at the Curie point. (Image courtesy of AVX Corp.)

film. The capacitance is half as high with the same overall dimensions. There are so-called “non-etched” electrolytic capacitors, which have electrodes that are not roughened. Having the same capacitance, they are significantly larger, but they have a lower series resistance, which is why they are used in high-quality crossover units and hi-fi loudspeakers.

Steve Bench, a developer with Applied Digital Access (San Diego, CA) has found that the distortions are highest when the electrolytic capacitor is not biased with a DC voltage. Now, if a signal flows, it will be alternately polarized correctly and incorrectly. If the incorrectly polarized voltage is only connected for a short time, the capacitor should remain undamaged, but it falsifies the signal for the negative half-waves.

The effect is similar to bridging it using a bad diode—even harmonics appear. Many developers found the distortions to be lower if the electrolytic capacitor is continuously connected to the DC voltage of correct polarity, (i.e., the signal amplitudes are always lower than the DC voltage) so no wrong polarity on the curve will occur.

Paul J. Stamler of BEXT (San Diego, CA) recommends measuring the DC voltage at each electrolytic capacitor within the signal path and replacing those without DC voltage with bipolar

ones. This makes it slightly better, but in Bench’s opinion, this solution is still not good enough because some distortion remains. That is because even bipolar electrolytic capacitors work without bias voltage.

One typical application is the coupling of two subsequent amplifier stages. This layout represents a first-order high-pass filter. The lower cut-off frequency is calculated from the capacitance and the input resistance of the following stage. In this case, the lower the signal frequency (or the more the charge of electrolytic capacitor is reversed), the greater the distortion. Far above the cut-off frequency, where charge reversal becomes negligible, they almost disappear. If you cannot avoid electrolytic capacitors, Douglas Self, author of the *Audio Amplifier Design Handbook*, recommends opting for a significantly higher capacitance than required by the signal bandwidth (e.g., by a factor of 10), to reduce the charge reversal.

However, this may not work in all situations because the electrolytic capacitors are used as elements to determine the frequency response (e.g., in a low-cut filter for record play back) and therefore, as a high-pass filter. In this example, you should only use high-quality foil capacitors. Those also have a much narrower tolerance than electrolytic capacitors so the frequency response is no longer left to chance.

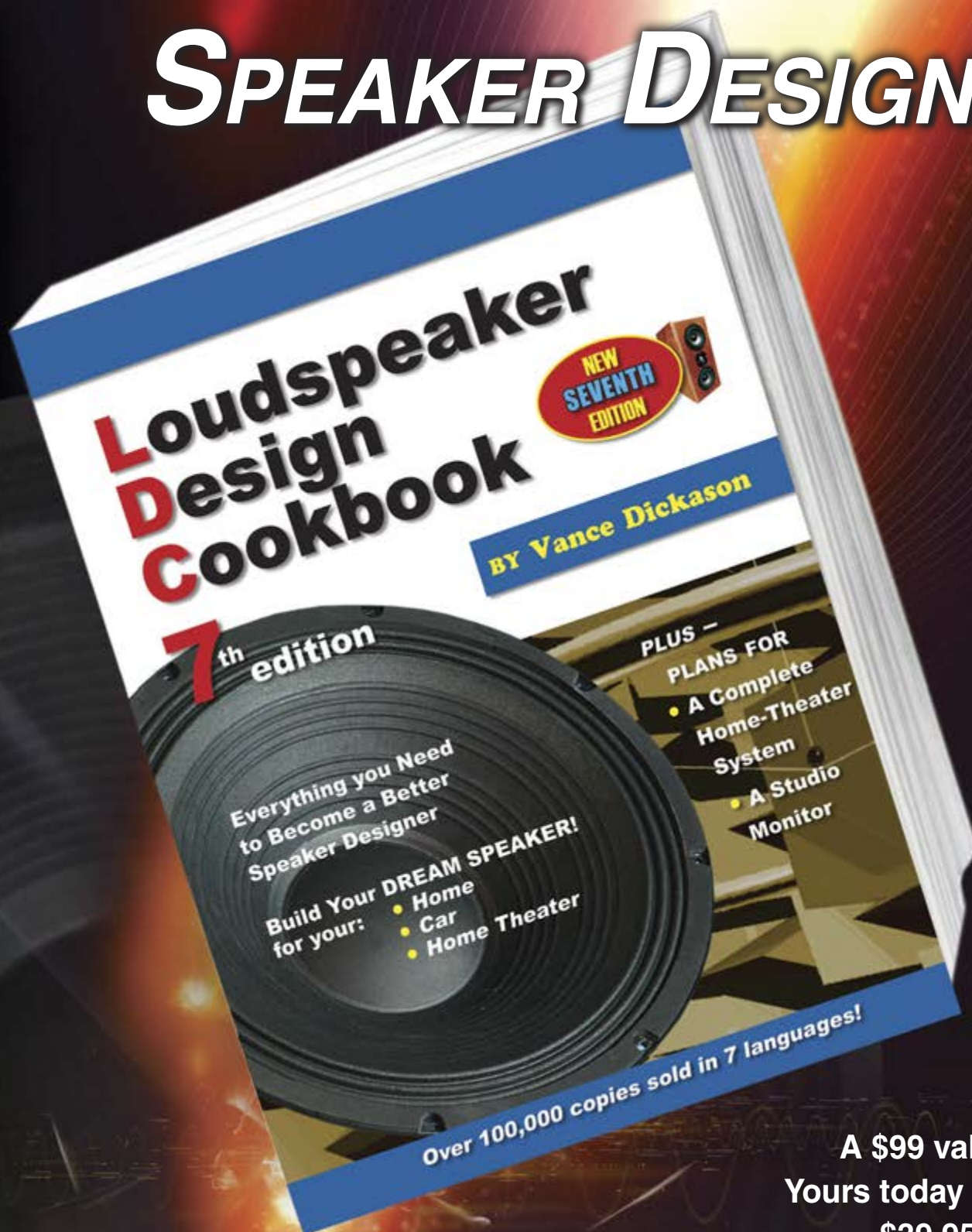
Tantalum and Niobium Are Forbidden

Those much younger capacitor types feature a much higher ratio between the capacitance and the overall dimensions than their aluminum siblings. That is because of the high dielectric constant ϵ_r of the isolator, for tantalum pentoxide it is 27, for niobium oxide it is 41. Their series resistance is lower, too. However, the distortion is much higher, easily reaching the percent range. Also, semiconductor effects in the dielectric play a role. Therefore, they should not be used in the signal path of high-quality audio circuits. Hi-fi experts say they ruin the sound. However, they perform well in blocking the supply voltage.

Table 1: Several ceramic dielectrics are compared.

Class	Class 1	Class 2
Dielectric	Metal oxides and titanates	Titanates and zirconates (lead, calcium, and strontium)
ϵ_r	up to 200	approximately 200 to 10,000
C/Volume ratio	low	high
Temperature departure	linear	nonlinear
Linearity	moderate	bad
Losses	low	high
Insulation resolution	high	moderate
Aging	low	high

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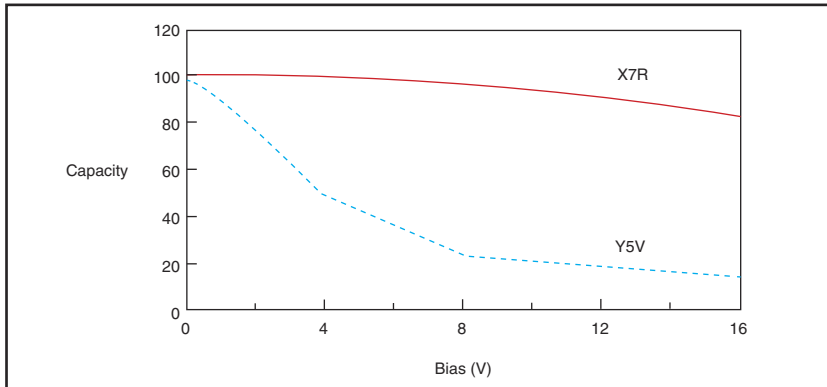


Figure 3: With Class 2 ceramic dielectrics, the capacitance can significantly drop under bias voltage. The "X7R" is moderately strong, but not suitable for high-end applications. The "Y5V" is very strong, but it is absolutely unusable. (Image courtesy of AVX Corp.)

The Mystery of Dielectric Absorption

Even capacitor experts don't know the precise reason for distortion. Observations show that some dielectrics have some sort of "memory" for the electrical fields to which they have been exposed. When charging a capacitor, then short circuiting it for discharge and eliminating the short with a connected voltmeter, the voltage slowly increases again. This effect is called dielectric absorption (DA). It becomes stronger the higher the dielectric constant is—not only with electrolytic capacitors but also with non-polarized capacitors, only in a much weaker way. The voltage regenerated after the discharge is called the "recovery voltage." It is specified as a percentage of the preceding charging voltage. Some capacitors, especially electrolytic capacitors, can achieve values

of more than 10%. For high-quality audio circuits, 1% is already unacceptably high.

When applying AC voltage, the DA avoids complete charging and discharging. Once the signal polarity is reversed, it generates a delayed current with the previous polarity. As a result, a hysteresis effect appears and increases with the frequency. With a hi-fi amplifier, the effect is a loss of accuracy, the playback seems imprecise, compressed and the dynamic range decreases, and the background noise increases.

According to John Curl, a high-end amplifier developer with Parasound, the impact of these effects widely depends on the shape of the curve of the signal: "Music is not a normal test signal, the curve is not necessarily symmetrical. The DA acts disturbing in particular with asymmetrical signals. The dielectric stores a part of the signal and releases it with delay. The capacitor weakens the peaks, so to say," Curl explained.

The standard MIL-C-19978D determines a test procedure for dielectric absorption, to which the capacitor manufacturers largely adhere. Connect the capacitor to the charging voltage for 5 minutes, discharge it for 5 seconds, let it rest with open connections for 1 minute, then measure the recovery voltage and specify as a percentage of the charging voltage.

Assuming an equal capacitance and different dielectric thickness, the capacitor with the greater dielectric thickness has the lower DA. That means that when an electrolytic capacitor is used, it is best to choose the one with the highest possible nominal voltage. Bipolar electrolytic capacitors have a lower DA than polarized ones. However, for most applications the DA is still much too high. Electrolytic capacitors made from tantalum have a higher DA than the ones made of aluminum.

Hi-fi enthusiasts frequently praise the sound qualities of tube amplifiers, which are supposed to be so much better than those of transistor amplifiers. As can be seen upon closer examination, an essential part of this difference is not caused by the amplifying components at all, but by the coupling capacitors. Tube circuits have high supply and signal voltages as well as high impedances. Therefore, relatively small capacitors with a high dielectric thickness are used (e.g., types with plastic film). In transistor amplifiers, it is exactly the opposite. When the voltages and impedances are low, bigger capacitors are needed and electrolytic capacitors are used.

According to Curl, "A big part of the distortions is generated by bad capacitors. Once they are replaced by high-quality types with low DA, the sound is often

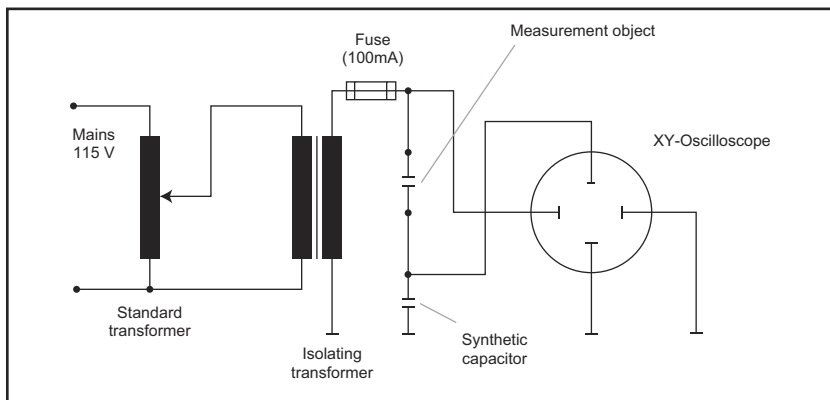


Figure 4: This is a simple measuring circuit for the hysteresis loop of ceramic capacitors. The integration capacitor must be much larger than the test object. The fuse is important in case of a breakdown.

About the Author

Helmuth Lemme studied physics (with an emphasis on solid-state physics and semiconductors). He then worked in the electronics industry for several years, and spent eight years as editor of the German magazine *Elektronik*. Helmuth is now a freelance journalist specializing in industrial electronics. His audio interests include electric guitars and their amplifiers. Helmuth is also the author of *Electric Guitar Sound Secrets and Technology* and he has written many articles on tube technology in *Elektor* special issues.

audibly improved. Then, the difference between tubes and transistors are much smaller. Some residual distortion is still left for which there are other reasons.”

The sound degradation caused by electrolytic capacitors is strongest wherever the signal levels are highest. In terms of audio, this is true in the power amplifier and the speaker crossover. The oldest ironless transistor power amplifiers worked with a single supply voltage. Between the output and the speaker, there was always a big electrolytic capacitor for DC voltage separation, typically about 2,200 μF . On one hand, its negative impact consists of the DA, which generates distortions in the low frequency range, and on the other, its negative impact consists of the series resistance, which is significantly high because of the high signal currents, which reduces the damping coefficient. Bridging them with a high-quality plastic capacitor of several tens of microfarads can prove substantial improvement.

However, today, this kind of circuit is largely extinct. With modern power amplifiers having symmetrical supply voltages, the output capacitor is no longer needed, which is a significant improvement. There is still another weak point though—the negative feedback path. The input stage is designed as a differential amplifier.

The entire circuit shows similarities to an operational amplifier. The input signal is routed to the non-inverting input. The output signal being divided by a voltage divider is routed to the inverting input (see **Figure 1**). This one often features an electrolytic capacitor at its base point. In this way, the power amplifier has an amplification factor of $(R1 + R2)/R2$ for AC voltage, but only the factor 1 with DC voltage. Then, the output only features the offset voltage of the differential amplifier stage present at the input. And, the electrolytic capacitor is a normal, polarized one. Here, it is alternately polarized correctly and incorrectly, typically with several 100 mV, which it is able to withstand. However, this can make it generate harmonic distortions of up to 1 % which is unacceptably high for hi-fi applications.

With the typical dimensioning specified in **Figure 1**, the lower cut-off frequency (a

3 dB drop in gain) is 16 Hz.

Assuming the power amplifier’s input voltage is 1 V_{EFF} at 16 Hz, 0.7 V_{EFF} is measured at the electrolytic capacitor and correspondingly less in case of higher frequencies. In case of dull drum beats, distortions may occur that can be perceived by fine ears. Here, a bipolar electrolytic capacitor is slightly better, but still not ideal.

It is more efficient to make the electrolytic capacitor much larger (e.g., 100 μF , as in this case only 1/10 of the AC voltage is present). However, it is best not to use any capacitor at all and eliminate the DC output voltage taking other measures. Using a suitable control circuit, that task can be accomplished except for a few millivolts, which are harmless to the speaker.

The speaker crossover units are another topic. In this case, electrolytic capacitors mounted prior to tweeters and mid-range drivers are bad, so are bipolar and unetched ones. Usually only the too high series resistance is the reason.

In reality, however, the dielectric absorption should also be considered. Here, plastic capacitors are miles better, therefore, a replacement can significantly upgrade the cabinet. It seems to be less critical with capacitors that are connected in parallel to the woofer as in crossovers with a 12 dB/octave filter slope. Apart from that, the usual large tolerances of the electrolytic capacitors are negative. Sonic variations in speaker cabinets are really unwanted. Plastic capacitors can be manufactured in a more precise way.

Ceramics: Evil of a Different Kind

To make the ratio between capacitance and overall dimensions of non-polarized capacitors as high as possible, one searches for insulating material with the highest possible dielectric constant. “Perovskite”-type ceramics feature very high values. The “forefather” of this group is barium titanate (BaTiO_3). However, this material has a disadvantage: it is ferroelectric. In this case, the electric field strength (E) and the dielectric displacement (D) are not linear as with the ideal capacitor, but they form a hysteresis loop similar to ferromagnetic materials.



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After removing the outer field strength, some dielectric displacement is left, the direction of which can be reversed by an opposing electric field. Over the past several years, this effect is being used for manufacturing data memories. It is extremely temperature dependent. If the temperature exceeds the Curie temperature, it vanishes. Every ferromagnetic material loses its magnetic characteristics once the Curie temperature is exceeded. (Curie temperature is named after the chemist and physicist Marie Curie, 1867-1934).

The nonlinear correlation between E and D still remains. With growing field strength, the dielectric displacement cannot keep up, but it increases more slowly, similar to coils where the iron core, due to its nonlinear characteristic curve, reaches magnetic saturation with increasing current. Then, the dielectric constant $\epsilon_r = D/E$ no longer has a fixed value, instead it depends on the voltage. This effect is common with capacitance diodes, there it is specifically utilized. However, if you take a standard capacitor, this may not immediately realize it. As a result, signal distortions occur in the circuit. The higher the dielectric constant, the stronger the temperature dependency. ϵ_r is highest near the Curie temperature and lower the further it is away from it (see **Figure 2**).

Ceramic capacitors are available in a variety of designs. Originally, small tubes only had one layer. At that time, capacitances of up to about 0.1 μF could also be found. Modern versions are mostly built with

multiple layers, allowing small overall dimensions. They are widely spread in surface-mount design. Ceramic dielectrics are roughly divided into two classes (see **Table 1**). The most interesting feature is the nonlinearity. The decrease of capacitance under bias voltage is closely linked. Class 2 materials can cause extreme results (see **Figure 3**). With Class 1 material, the effect is weaker and it doesn't occur with electrolytic capacitors.

Therefore, the countless ceramic mixtures are tailored in a way that hysteresis and saturation effect remain as low as possible. However, a certain amount remains. **Figure 4** shows a simple measuring circuit with which the D/E curve can be represented using an oscilloscope.

Depending on the material, it takes very diverse forms. Some capacitors show a smooth "S" curve (see **Figure 5a**). Others feature entanglements (see **Figure 5b**, **Figure 5c**, and **Figure 5d**). There are even asymmetrical ones (see **Figure 5e**), when the same material is used. The form depends on the temperature and the measuring frequency. An ideal capacitor shows a perfectly straight line.

Class 1 ceramic capacitors should be treated with extreme caution. Class 2 capacitors should not be used at all in high-quality circuits because of their unpredictable behavior. The tone control part is the most vulnerable point as they are recharged to a much stronger level between amplifier stages. They are, at best, harmless for blocking supply voltages or in points where they are supposed to suppress

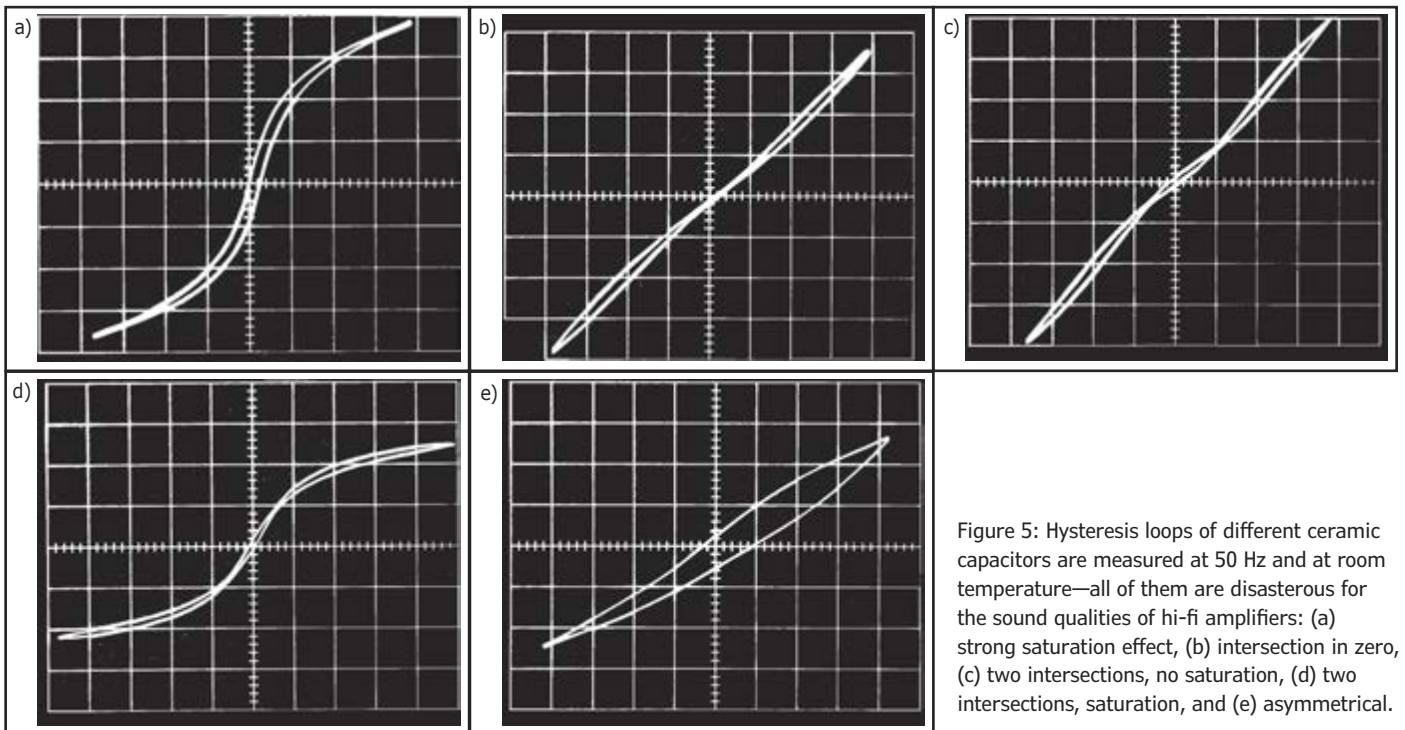


Figure 5: Hysteresis loops of different ceramic capacitors are measured at 50 Hz and at room temperature—all of them are disastrous for the sound qualities of hi-fi amplifiers: (a) strong saturation effect, (b) intersection in zero, (c) two intersections, no saturation, (d) two intersections, saturation, and (e) asymmetrical.

parasitic oscillations or high-frequency irradiations.

There is another unpleasant characteristic that can occur with ceramic capacitors. They are often microphonic because many of these dielectrics are piezoelectric! Because of that, vibrations from the surroundings (e.g., coming from the speaker or the mains transformer) can enter the signal path. This definitely puts an end to hi-fi quality.

Good Capacitors

According to Curl, it is best to keep the signal path completely free from capacitors and coils. However, that is hardly achievable. The ideal capacitor features a vacuum between the plates, which is not feasible.

The second best has air, which is excellent from an electrical point of view, but with a ratio between volume and capacitance that is just as unfavorable as with the vacuum. Plastics with a low dielectric constant show relatively good characteristics. Even in this case, the

dielectric absorption is not completely zero, but it is so small (a magnitude of 0.01 %) that it is no longer a problem.

Teflon has the best reputation, closely followed by polypropylene. Also polystyrene (Styroflex) and polycarbonate are still deemed to be quite good. Polyester (Mylar) gets the worst scores because of its DA of up to 1%.

However, it is still significantly better than electrolytic capacitors and ceramics. Individual developers have different opinions on other, more rarely used dielectrics (e.g., glass, mica, and oil- or wax-soaked paper). Overall, they are considered inferior.

Final Analysis

Ceramic capacitors as well as tantalum- or niobium-based electrolytic capacitors should not be used in the signal path. Building high-quality amplifiers and metrologically precise circuits doesn't necessarily have to be prohibitively expensive. You only need the right know-how in the right places. 

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Some Background Information on Noise in Amplifiers



The perfect amplifier has been described as a “straight wire with gain.” This comparison implies complete freedom from nonlinear distortion, a frequency response from DC to light (well, more-or-less), and complete absence of electrical noise. Having discussed distortion and frequency response in previous articles, let’s consider noise.

By
Richard Honeycutt
(United States)

Noise may be defined as any undesired signal. Any amplifier is susceptible to different types of noise. All of them may be roughly classed as either external noise or internal noise. External noise is noise induced either electrostatically (by capacitor action) or electromagnetically (by transformer action) from external sources. These external sources include lightning, motor-vehicle ignition systems, power-line related radiation (hum), electrical equipment radiation (brush noise), and sunspot, cosmic, and space noise.

We can effectively reduce external noise by placing the low-signal-level conductors and circuitry within a grounded conductive enclosure. This is the purpose of shielding in cables, such as those used for oscilloscope probes.

Random Noise

Internal noise is generated within the circuit itself. Internally generated noise could include harmonics and other spurious frequencies generated within a

circuit. These can result from nonlinear distortion, or from issues with grounding or power supply performance. To begin, we will discuss the random noise created in the circuit without respect to any input signal. This includes noise from several causes.

The first is the discrete nature of electron flow. Since electrons are discrete charge carriers, each time an electron moves it creates a tiny noise pulse. More to the point, each time an electron bumps into another particle it releases some energy, part of which is electrical energy of random frequency content. Since bumping into other particles is partially a result of resistance and partially a result of the thermal agitation of the electrons, both temperature and resistance contribute to this type of noise. This noise, called Johnson noise in honor of J. B. Johnson, who described it in 1928, contains equal amounts of energy at all frequencies. Therefore, by comparison with white light, which contains equal amounts of light at all light frequencies, it is said to be white noise.

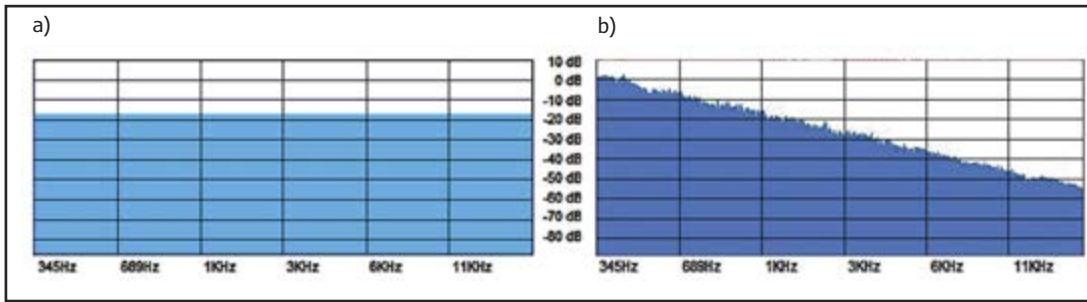


Figure 1: White noise (a) is independent of frequency, while pink noise (b) has a level inversely proportional to frequency.

Figure 1 shows a comparison of white noise (a), such as that generated by random electron motion, and pink noise (b), which is often deliberately generated for use as a test signal. Also, since Johnson noise is generated by a random process, we can only predict the probable voltage or current level of the noise at any given time; the exact level is unpredictable. The statistical prediction of noise level indicates that the likelihood that a certain noise level will exist under certain conditions decreases as the level is increased or decreased from the most likely level.

This rather confusing statement is clarified by **Figure 2**. The graph shows what is called a Gaussian distribution of noise levels. So we can say that Johnson noise is white, which tells us something about its frequency content.

We can also say that it is Gaussian, which tells us something about its probable amplitude. Again, Johnson noise is a property of all conductors. The equation for the most probable amplitude of the Johnson noise in a conductor is:

$$v_n^2 = 4kTR\Delta f$$

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

This equation clearly shows the effect of temperature and resistance upon noise level. It can be simplified by lumping all the constants and most-common variables together:

$$v_n = \left(\frac{4nV}{\sqrt{Hz}} \right) \left(\sqrt{\frac{R}{1k\Omega}} \right) (\sqrt{\Delta f})$$

The equation shown above is valid for a temperature of 298 K (25°C). To use it, simply plug in the value of R (in kilohms) to be used and multiply by the square root of the bandwidth (in hertz) to obtain the total voltage.

Shot Noise

Another type of internal noise results from an electric current crossing a potential barrier, which occurs in any electronic device. This noise was dubbed "shot noise" long ago, because it sounds like lead shot hitting a tin roof. Shot noise acts as a current source rather than a voltage source. Its most probable value is:

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

where q is the charge in coulombs of a single electron, and I_{DC} is the direct current (DC) in the component. Shot noise is a white noise, and it also has a Gaussian distribution of levels.

A vacuum diode produces a shot noise current given by:

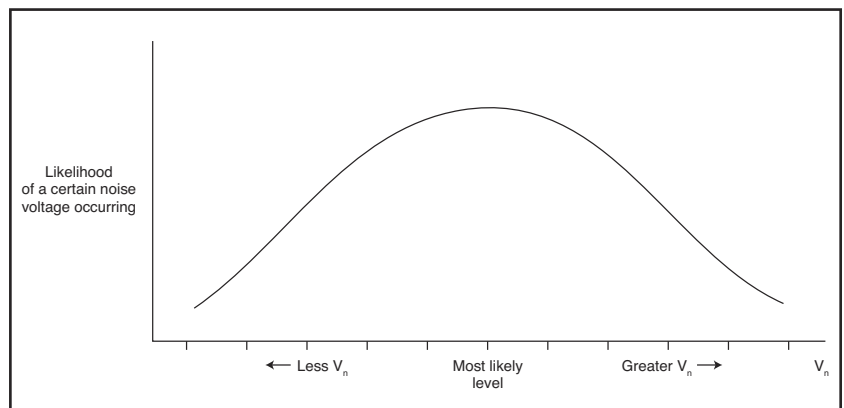
$$i_s = F\sqrt{2qI_p\Delta f}$$

where the noise factor F is less than 1 for an unsaturated diode, equal to 1 for a saturated diode, and is determined by the space charge. When grids are added, the noise increases more and more as additional grids are added. Thus, pentodes are noisier than triodes.

1/f Noise

Many devices generate frequency-dependent noise, which is variously called 1/f noise, contact

Figure 2: The levels of thermal, shot, and 1/f noise levels have Gaussian distribution.



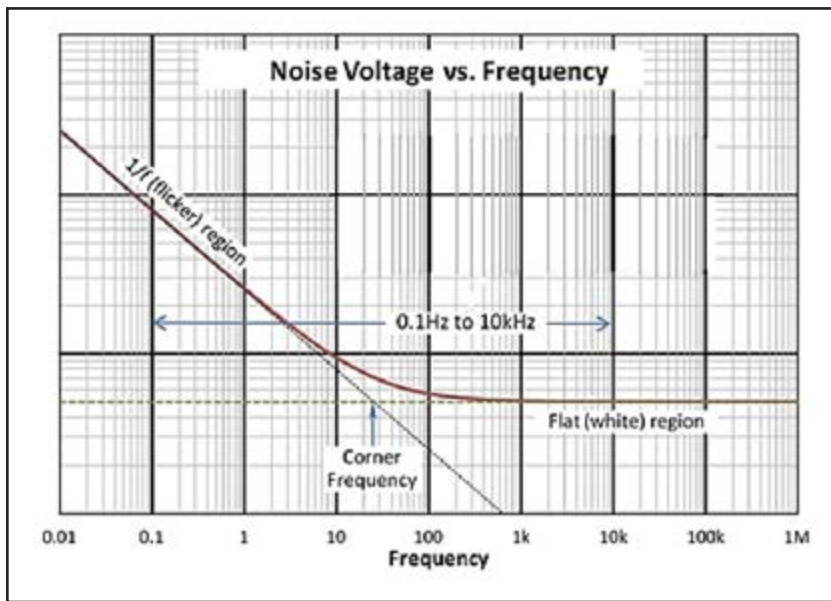


Figure 3: The “corner frequency” is where the white noise becomes dominant over the 1/f noise. (The red curve is the sum of white and 1/f noise.)

noise, flicker noise, or excess noise. This type of noise also has a Gaussian probability distribution, and its most probable value at any single frequency is given by:

$$i_f^2 = \frac{K_f I_{DC}}{K_2 f^\alpha}$$

where K_f and K_2 depend upon the properties of the device.

The exact frequency dependence is given by $1/f^\alpha$, where α is an experimentally determined constant having a value between 0.5 and 1.5, but most commonly very near unity. (You may have figured out by now that 1/f noise is the same as pink noise, having equal energy per octave, rather than equal energy per hertz, like white noise. However,

the term “pink noise” is generally applied only to intentionally generated test signals.)

Because of the decreasing noise voltage as frequency rises, in any circuit with 1/f noise, there will be a “corner frequency” where the value of thermal noise begins to dominate the noise signal (see **Figure 3**).

In hollow-state circuits, 1/f noise is generated most prolifically by carbon, metal oxide, or metal film resistors, but not wirewound resistors (see **Photo 1**). However, for some applications, the parasitic capacitance and/or inductance of wirewound resistors prohibit their use.

We do not yet have an explanation for the mechanism by which 1/f noise is created, but there are other natural phenomena exhibiting the 1/f frequency distribution, including sea levels, earthquake magnitude vs. number/year, and loudness of music over a period of time.

The important things to notice about all this information on noise are the following:

- The total power associated with Johnson and shot noise depends upon the bandwidth of the circuit.
- The value of the Johnson-noise-voltage source also depends upon temperature.
- Greater resistance also results in greater Johnson noise.
- The value of 1/f noise depends upon the DC current and the circuit’s low-cutoff frequency.

It is probably not worth your time to try to memorize the equations for the various types of noise, but understanding the brief introduction that has just been given will be a help in the study we will later make of how to minimize noise in an amplifier circuit.

The internal noise sources for a hollow-state amplifier can be summarized as follows:

- v_{ns} —the Johnson noise produced by the grid resistor in parallel with the Thevenin resistance of the signal source
- v_{nf} —the 1/f noise produced in resistors
- i_s —the shot noise resulting from current flow in the tube

Since we usually characterize amplifiers in terms of voltage gain, the noise currents must be turned into noise voltages before they are amplified. This occurs when they flow through the resistance from the input or output to ground. The total equivalent input noise results from the addition of all these



Photo 1: Resistors of different technologies have different noise performance: (from top) 1 W carbon-comp, 0.5 W carbon comp, 0.5 W carbon film, 0.25 W carbon-comp, 0.5 W carbon film (different manufacturer), 0.125 W metal film, 1 W wirewound, and 2 W wirewound.

noise sources' values in a root-sum-square fashion:

$$v_{n,in} = \sqrt{v_{ns}^2 + \left[i_n \left(\frac{r_p r_L}{r_p + r_L} \right) \right]^2 + v_{nf}^2}$$

In general, the schemes for minimizing noise work to reduce or eliminate the effects of one or more of these equivalent noise generators. To simplify noise calculations, the effects of thermal noise and shot noise can be combined and specified as an equivalent noise resistance R_{eq} .

For a triode, $R_{eq} = 2.5/g_m$. The value is typically between 150 Ω and 2 k Ω .

For a pentode,

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

The value is commonly between 1 k Ω and 50 k Ω .

Once we know the value of R_{eq} , we can determine the equivalent RMS input noise voltage from a tube and its grid resistor from:

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

In hollow-state amplifiers, the 1/f noise is pretty much limited to that produced by resistors, and cannot be readily calculated; instead, we just use low-noise resistors. Metal-film resistors are quieter than carbon-film ones, and carbon-film ones are quieter than carbon-composition resistors. Since 1/f noise depends on the current density (amperes per unit area), a higher-wattage carbon-comp resistor will produce less noise for the same current in the device than—say—a 1/8 W one.

Although thermal noise, shot noise, and 1/f noise in amplifiers can be bothersome, hum and buzz are often far more irritating. It may seem that hum problems could be eliminated by using enough filtering in the power supply, but in fact, grounding, layout, and shielding are at least equally important.

Good Grounding

The key to good grounding is to avoid having any impedance shared by multiple ground leads. For example, **Figure 4** shows a three-stage amplifier whose cathode resistors are all grounded. If each is grounded directly to the circuit's common ground point, all is well. But if the bottom of each cathode resistor is connected to a separate terminal, then subsequently that terminal is grounded through a wire. The wire acts as a common grounding impedance, and its connection point to the cathode

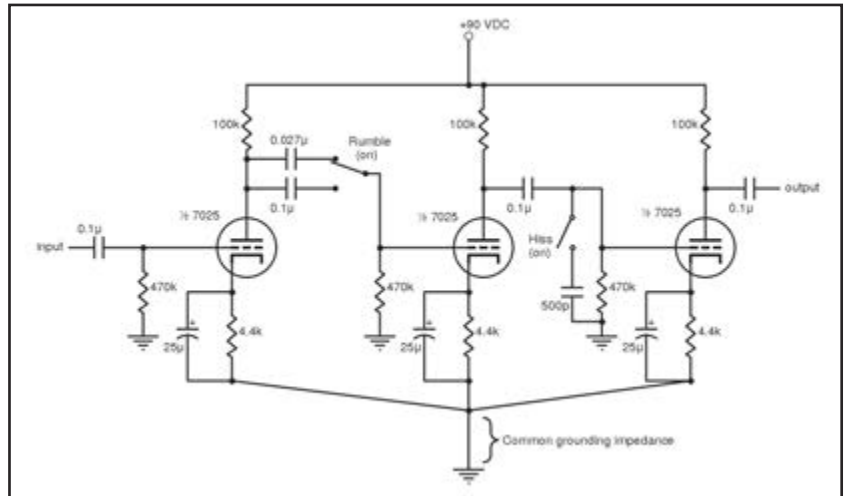


Figure 4: Different stages should not be grounded through the same wire.

resistors will not be a true ground. It 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 on where they are located in a circuit. This problem is greatest in the low-impedance portions of a circuit (power supplies and output stages).

So what is the best common ground point? Usually, it is the ground terminal at the input connection. Occasionally, you can get by with grounding directly to a metal chassis at virtually any point close to the component lead to be grounded, but it's far safer not to take chances.

Aside from grounding problems, the other likely cause of hum and buzz is signals induced capacitively from AC electric fields, or inductively from AC magnetic fields.

To avoid induced hum, distance is your friend: the farther low-signal stages are from hum generators (e.g., power transformers), the better. Laying out the circuit so that the input stage is at one end of the chassis, followed by subsequent gain stages, and ending with power amplifier stages and then the power supply is a good solution.

Induced hum and noise can also be reduced by shielding. Power transformers in steel cases produce less AC magnetic leakage than open ones. In very low-voltage amplifier stages (e.g., phono preamps), the signal lead from the input connector to the grid of the first stage should use shielded wire. The tubes of the first stage or two should have shields. In the days of octal tubes, we had the option of using metal tubes, which provided superior shielding.

In a future article, we will discuss more specific steps you can use in creating a low-noise hollow-state amplifier design. 



Here are a few places where you might find a copy of *audioXpress* or possibly meet one of our authors and staff members:

November 2–6, 2015

Hyatt Regency Jacksonville Riverfront Hotel, Jacksonville, FL
acousticalsociety.org

Acoustical Society of America 170th Meeting



With an amazing Technical Program and Itinerary already planned for the 170th meeting of the Acoustical Society of America, this important event features valuable technical sessions and paper presentations. The technical sessions for the 170th meeting include 90 sessions with 742 abstracts scheduled for presentation and include topics in architectural acoustics, engineering acoustics, musical acoustics, noise, physical acoustics, psychological and physiological acoustics, structural acoustics and vibration, speech communication, and signal processing in acoustics among others. An instrument and equipment exhibit will run from Monday to Wednesday. The exhibition will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems and other exhibits on acoustics.

November 6–8, 2015

Hilton Westchester, Rye Brook, NY
www.chestergroup.org

New York Audio Show



Another valuable opportunity for companies in the consumer and high-end audio market is the New York Audio Show, promoted by the Chester Group. With more than 40 demo rooms and top headphone brands, this weekend show features a wide selection of loudspeakers, electronics, turntables, headphones, streaming music, computer systems, and accessories with opportunities to meet the designers, creators, and guest speakers from the industry. Live music, competitions, guest speakers, and special events also contribute make The New York Audio Show an entertaining event and valuable networking opportunity.

November 14–15, 2015

Shenzhen Virtual University Park, Shenzhen, China
www.iseat.org

ISEAT2015



Held every two years, the International Symposium on ElectroAcoustic Technologies (ISEAT) will be chaired this year by Professor Yong Shen, director of the Shenzhen Acoustics R&D Center of Nanjing University. ISEAT is a professional organization for researchers, engineers, teachers, and students in electro-acoustics. Exhibitors include most of the audio test gear companies, and several large speaker companies exhibit (and recruit staff) at the event. The venue is the Shenzhen Virtual University Park, which is located in the Shenzhen High and New Technology Industry Development Zone on the bank of Shenzhen Bay. About 600 representatives from companies, universities and research institutes in China and overseas will attend this Symposium. ISEAT always invites prestigious scholars to give keynote speeches, and renowned experts, senior engineers and senior managers will share their latest discoveries, developments, and applications for new technologies, materials, equipment and market trends in electro-acoustics.

November 18–20, 2015

2-1, Nakase, Mihama-ku, Chiba City, Chiba Prefecture, Japan
www.inter-bee.com

Inter BEE 2015



Held for half a century since 1965, the 51st International Broadcast Equipment Exhibition (Inter BEE) 2015/ Makuu Messe, is one of the most important shows in Japan and certainly for all major Japanese electronic companies active in the audio, video, and broadcast markets. The show combines four exhibit categories: Video and Broadcast Equipment, Professional Audio Equipment, Professional Lighting Equipment and ICT/Cross Media. The exhibition is held under the support of the Ministry of Internal Affairs and Communications, the Ministry of Economy, Trade and Industry, the Japan Broadcasting Cooperation (NHK), the Japan Commercial Broadcasters Association and the Association of Radio Industries and Businesses, and with the cooperation of many other relevant organizations.

January 3–4, 2016

The Tuscany Suites & Casino, Las Vegas, NV
www.almainternational.org/events/alma_symposium__expo_2016

ALMA International Symposium & Expo 2016



The Association of Loudspeaker Manufacturing & Acoustics (ALMA) International is the only not-for-profit association of loudspeaker and electro-acoustic professionals dedicated to advancing the industry. For 2016, the ALMA International Symposium & Expo (AISE) theme will be: "Hi Res Audio: A Bridge To Build or a Bridge Too Far?" The topic is to discuss the pros and cons, as well as the opportunities and the risks attached to high-resolution audio (HRA) and what will it take to "build a bridge" to mainstream consumer as well as pro high resolution audio adoption. Dr. Mark Waldrep will deliver the Keynote Address, "The Myth of High-Resolution Audio/Music," at the ALMA International Symposium & Expo 2016 on Sunday evening, January 3, 2016. The exhibition will be improved for 2016 with additional space and expanded exhibit hours. *audioXpress* and *Voice Coil* magazines will participate in the exhibition and hope to scheduling meetings with ALMA members to discuss the publications' 2016 editorial calendars.

January 6–9, 2016

Las Vegas, NV
www.cesweb.org

CES Show 2016



With the motto, "Technology Changing the World," the Consumer Electronics Show (CES) 2016 will once again be the world's gathering place for all who thrive on the business of consumer technologies. Innovative technologies such as 3D printing, unmanned systems, health, smart homes and safe driving will be addressed at CES across more than 20 Marketplaces. At Tech East, there will be innovations in audio, HDMI, automotive electronics, gaming, video, wireless devices, wireless services, digital imaging/photography and all "i" devices. The Tech West exhibits will feature emerging technologies, encompassing revolutions in fitness and health, the Internet of things, smart home and other high-growth technologies changing the way we live, work and play. And, Tech South, the newest official show location for 2016, will host a variety of C Space activities, including conference sessions, networking events, exhibits and meeting rooms—all geared towards the advertising and marketing community at CES.

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