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March 2017 ISSN 1548-0628

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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 KCK Media Corp., at 111 Founders Plaza, Suite 904, East Hartford, CT 06108, US Periodical Postage paid at East Hartford, CT, and additional offices.

Head Office:
KCK Media Corp.
111 Founders Plaza, Suite 904
East Hartford, CT 06108, US
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Subscription Management:
audioxpress
P.O. Box 462256
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Postmaster: send address changes to:
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Serious Audio Challenges

Every time I get the chance to meet with audio test and measurement companies, usually during trade shows and conventions, I am fascinated by the things I learn and the challenges that these companies work daily to solve. Increasingly, I feel inspired by the ideas we exchange and the new challenges I learn they are helping to solve.

Most test and measurement companies focus on meeting existing market requirements for existing processes, from pure research projects to product development and, most frequently, creating reliable, simple, and repeatable processes for quality control, high-speed production tests, and troubleshooting all kinds of situations in the field or at the test bench.

I also see those same manufacturers pushing the boundaries of what's possible in order to provide the tools that audio research demands, in terms of measuring extended frequencies and analyzing signals in completely new ways to provide useful windows for the advancement of audio science.

There's nothing more challenging. The people who daily provide the research on which the audio industry evolves, depend on test and measurement tools to generate the data to measure and demonstrate their areas of work. Meanwhile, the test and measurement companies must follow those cutting-edge research challenges very closely to develop and provide the tools upon which that research depends. The diversity of the field work needed to understand continuously improving areas of research and convert that knowledge into practical solutions that can verify and certify all the generated data, is precisely what makes test and measurement instrumentation so interesting.

Apart from the already challenging electronic designs, every day we find examples of completely new requirements in areas of research in the acoustics and electro-acoustics, mechanical and electrical, analog and digital signals, wired and wireless domains. There is a pressing need for better understanding of materials, complex sound interactions in the acoustic domain, extended frequency ranges, the adoption of higher bit-depth, higher sampling, and for matching and simulating our own hearing and perception mechanisms. All this is propelled by a surprising number of audio applications expanding into new highly sophisticated domains, from hearing enhancement to far-field voice recognition, acoustic compensation to three-dimensional soundfield processing, head-related transfer functions (HRTF) with motion tracking, new materials and transducer design, and increasingly pushing microspeakers and electromechanical systems to the limits of physics and DSP optimization, to just mention a few trending examples.

No wonder we are seeing completely new instrumentation companies interested in audio research and introducing new methods and technologies to provide better quality and larger quantities of measurement data, and increasingly bridging computer simulation solutions with real-world measurements. And yet, the audio industry increasingly needs to interface those new instruments with existing measurement and analysis tools in order to use them in a practical way, while matching the required levels of certification.

The audio industry has been founded on the belief that subjective impressions matter the most and that "we can hear what we cannot measure," as someone once said. Fortunately, the demands in the aforementioned applications require that we do measure what we hear and use objective measurements to back it up.

This is also why I feel so grateful knowing that each year our Focus on Test and Measurement special edition is one of our most popular issues. The DIY tradition in this field apparently also remains strong. I would like to deeply thank all the authors for their contributions. You are contributing in a positive way to some seriousness in the audio domain.

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.

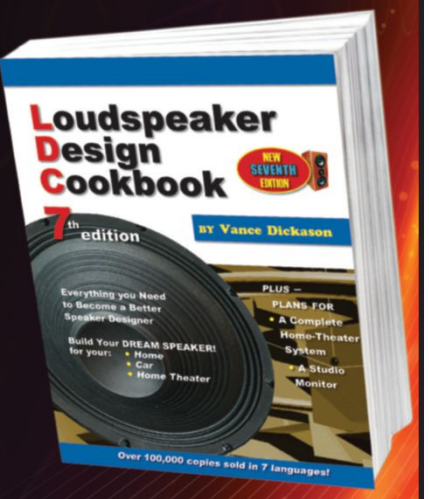
Dr. Richard Honeycutt fell in love with acoustics when his father brought home a copy of Leo Beranek's landmark text on the subject while Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical 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.

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

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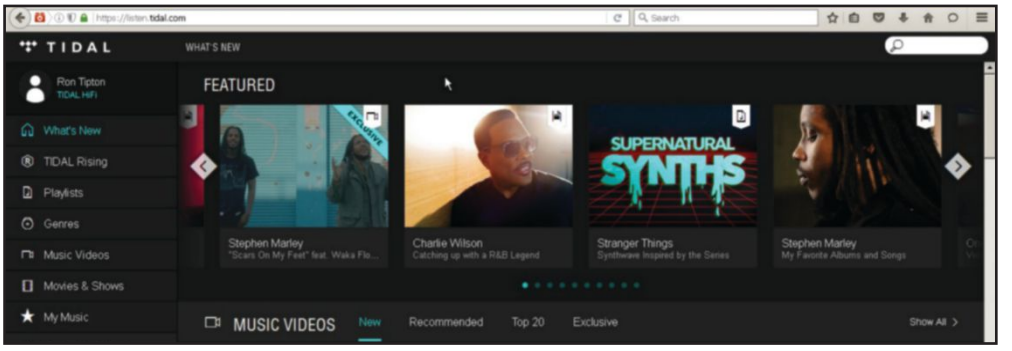
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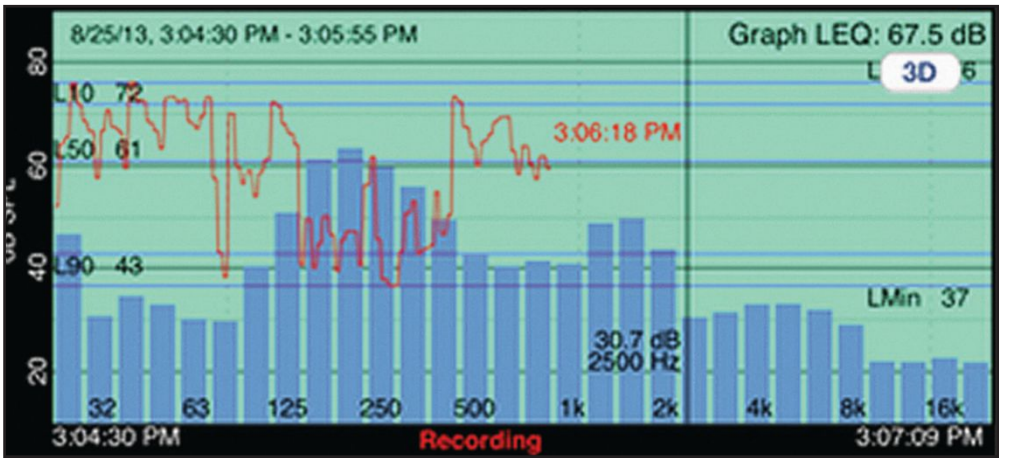
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Streaming Music Experiences (Part 3)

Streaming from TIDAL HiFi

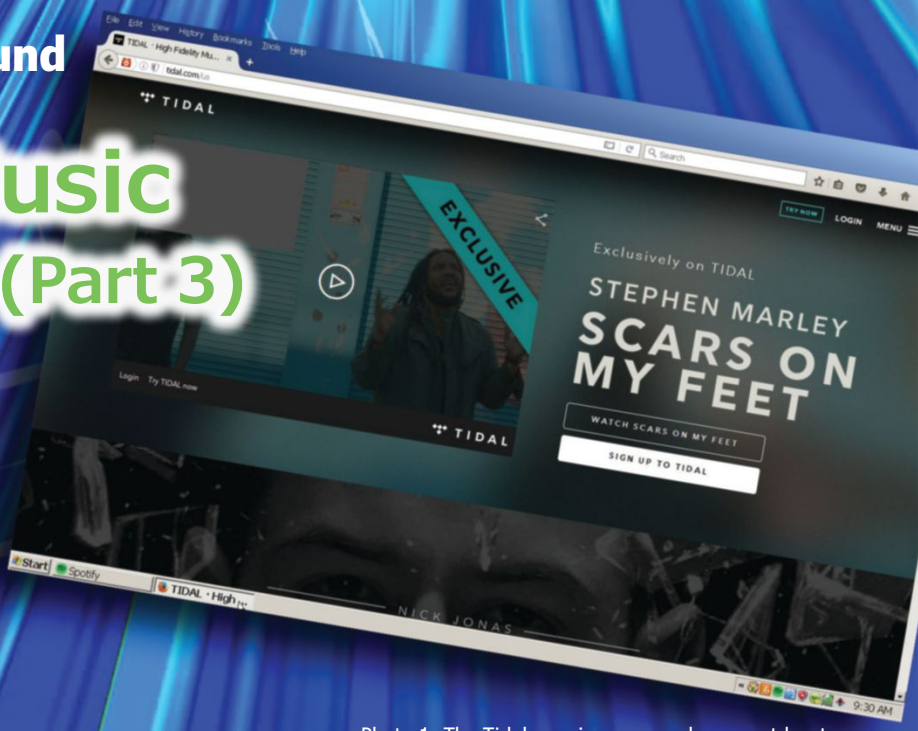


Photo 1: The Tidal opening screen changes at least daily as new material is advertised. Get started by clicking Try Now, Sign Up to Tidal, or Logon if you already have an account.

By
Ron Tipton
(United States)

How good are streaming music services? Are we able to enjoy the best sound reproduction using music that's streamed over the Internet given the bandwidth restrictions and variable data rates? In this article series, we continue our real-world experiences with streaming services, as we search for the best possible quality.

Both TIDAL and Qobuz stream full CD quality music: 16-bit depth at 44.1 kbps. The monthly subscription cost for each is about $26, each with a 15-day trial period before the subscription fee is applied. You can cancel at any time but there is no refund for unused days. Qobuz also streams part of its 30 million song collection in "Hi-Res with Sublime"—24 bits at up to 192 kHz, but this service is only available for an annual cost of about $292 (€219.99) with no free-trial period. You can browse the website to discover which tracks are available in high-resolution and visit the blog at http://blogsv2.qobuz.com/qobuz-blog-en/ for the latest information. When Qobuz recently became available in the US, I already had a CD-quality TIDAL account so let's look at its features.

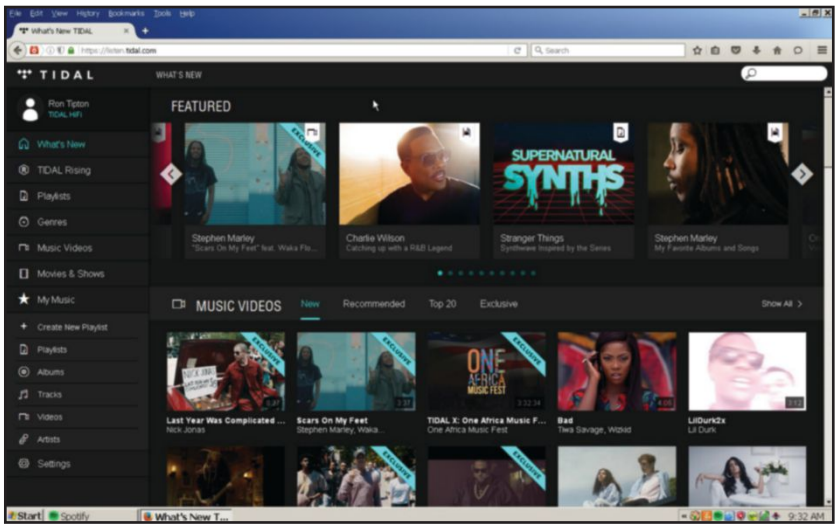


Photo 2: After logging on, this screen displays featured selections and is usually different from one logon to the next. The Options list is in the left-side column.

TIDAL

By accessing TIDAL via its website, tidal.com, you can load the opening screen (see **Photo 1**). Here you can TryNow, Logon if you have an account, or Setup a new account. If you don't need or want the CD-quality service, you can sign up for MP3 streaming (320 kbps) for about $13 (£9.99) per month.

I entered my user name and password, which loaded the screen shown in **Photo 2**. Notice that my name and service plan, TIDAL HiFi (Lossless

FLAC), is displayed in the upper left corner. The list of where you can go next is displayed in the left-side column. I usually start by clicking Genres, which loads the screen shown in **Photo 3** with all the music categories. Strangely, they don't seem to be in any particular order—alphabetical would have been more convenient. The music sounds very good, but I wanted to know how good it actually is. I devised some tests to quantitatively answer this question, which I will discuss in detail.

Streaming Distortion

The first question I wanted to answer is: Does the streaming channel introduce distortion? It seemed the best way to do this was to record the same track streamed at different times and compare them. To be sure of a truly valid answer, this would have to be continuously done. But I decided that samples on several consecutive days would provide a practical answer.

Recording the stream was easily done with a program from NCH Software named SoundTap. It isn't free, but it is inexpensive at about $30. As best I can determine, recording the stream is legal in the US provided it is solely for your personal use. SoundTap is easy to use. You "arm" it by clicking on the Start Recording bar (see **Photo 4**).

Recording does not start until it detects an audio stream and this is displayed in the text box. Stop recording by clicking the Stop Recording bar. The saved file folder and name are displayed in the text box. The horizontal bar graph indicates the recording level, which should be two-thirds to three-quarters full-scale. There is also a setup screen where you can choose MP3 or .wav file recording and the folder for the saved files.

Comparing the recorded samples is easily done with Audio DiffMaker, which is a gem of a free program from Liberty Instruments. **Photo 5** shows a screenshot of a completed comparison. You start by using the setup screen to select your sound card and other parameters. Then you enter the two file names for comparison using the LOAD buttons. Then click EXTRACT. The program time and amplitude aligns to two files and then subtracts one from the other, leaving the residual noise plus any file differences. The sample rate error is also calculated and displayed in the text box at the screen bottom.

Table 1 summarizes my comparison results of the four music tracks from different albums. The large numbers in the Null Depth column indicates no perceivable difference between the compared samples. The sample rate errors are also negligible. I have graphed the difference files in **Figure 1**. Notice that I had to amplify each one by 48 dB. The noise level is less than -100 dBu. This is about the noise level of my sound card, and it includes any noise in the music files and any noise added by the Internet channel. Non-audible indeed. **Figure 1** shows the "raw" data as graphed by PSI-Plot—no filtering or smoothing was applied, just the 48 dB amplification.

Loudness

Streaming services have supposedly become more aware of maintaining equal loudness between

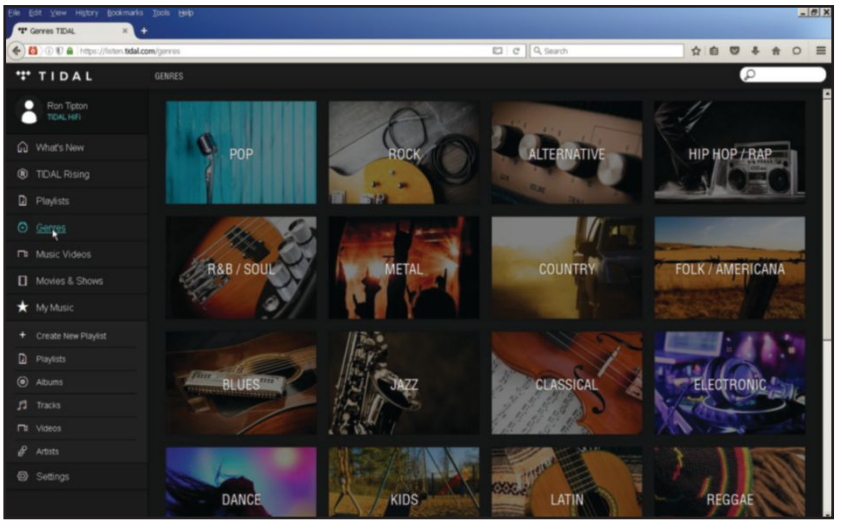


Photo 3: Clicking Genres in the left column loads this screen, which lists all the music categories. They seem to be in random order but there aren't too many to scroll through.

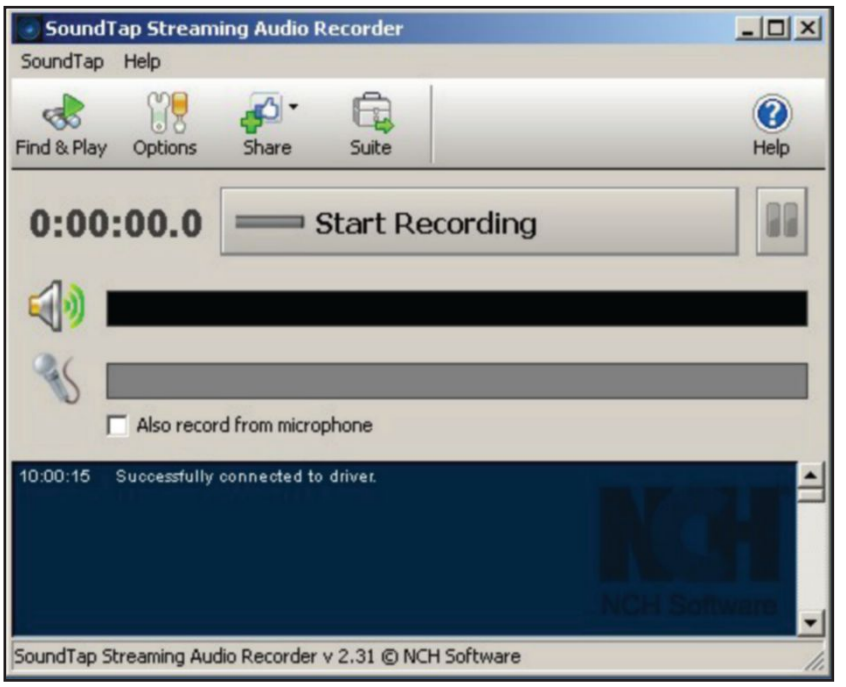


Photo 4: Click Options to choose MP3 or .wav file recording and to select the "save file" folder. Click Start Recording to arm the capture, recording won't start until an audio stream is detected and this will be displayed in the text box.

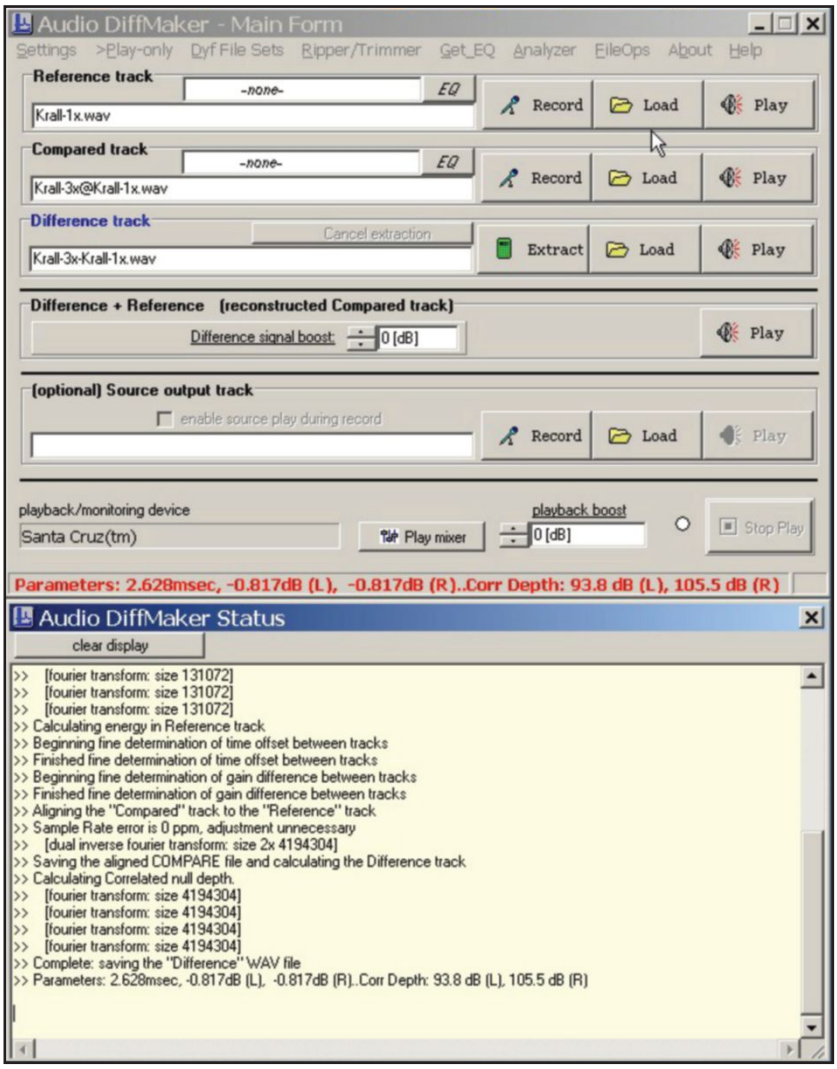


Photo 5: As DiffMaker runs, it constantly updates the text box with the current operation. Upon completion, several results are displayed: the sample rate error, the time added to align the files, and the left and right correlation depths. If there is no completion, an error message is displayed. Possible errors include the files being too long or a large mismatch between them.

Media	Sample Rate Error	Left Null Depth	Right Null Depth
Paul Desmond, "Desmond Blue," *Desmond Blue*, Marathon Media International	-0.0003 ppm	97.3 dB	88.1 dB
Diane Krall, "S Wonderful," *Look of Love*, Verve Records	0 ppm	93.8 dB	105.5 dB
"Memory," (from *Cats*) *The Great Broadway Collection, Vol 2*, London Theater Orchestra & Singers	0 ppm	97.4 dB	101.5 dB
"Old Man River," *The Great Broadway Collection, Vol 1*, London Theater Orchestra & Singers	0 ppm	120.4 dB	116.9 dB

Table 1: This table displays the sample rate error and the left and right null or correlation depths for four TIDAL tracks from different albums. Media information is also included. The column values were produced by the DiffMaker software, using the same tracks recorded on different days.

tracks. Usually this is not a problem when the tracks are from the same album. But when a playlist includes tracks from different albums, sometimes many albums, it can become a concern.

I captured a TIDAL playlist named *Classical Relaxation*, which was billed for "meditation, yoga, and sleep." If any group of music tracks should have normalized loudness, this one certainly should.

I used two software loudness analyzers: MusicScope and the Waves Audio PAZ Psychoacoustic Analyzer, to look at the first six playlist tracks—all from different albums. My results are summarized in **Table 2**, and it is apparent normalization wasn't used. The table just puts numbers to what I had already observed. I had to adjust the playback volume several times to achieve comfortable listening.

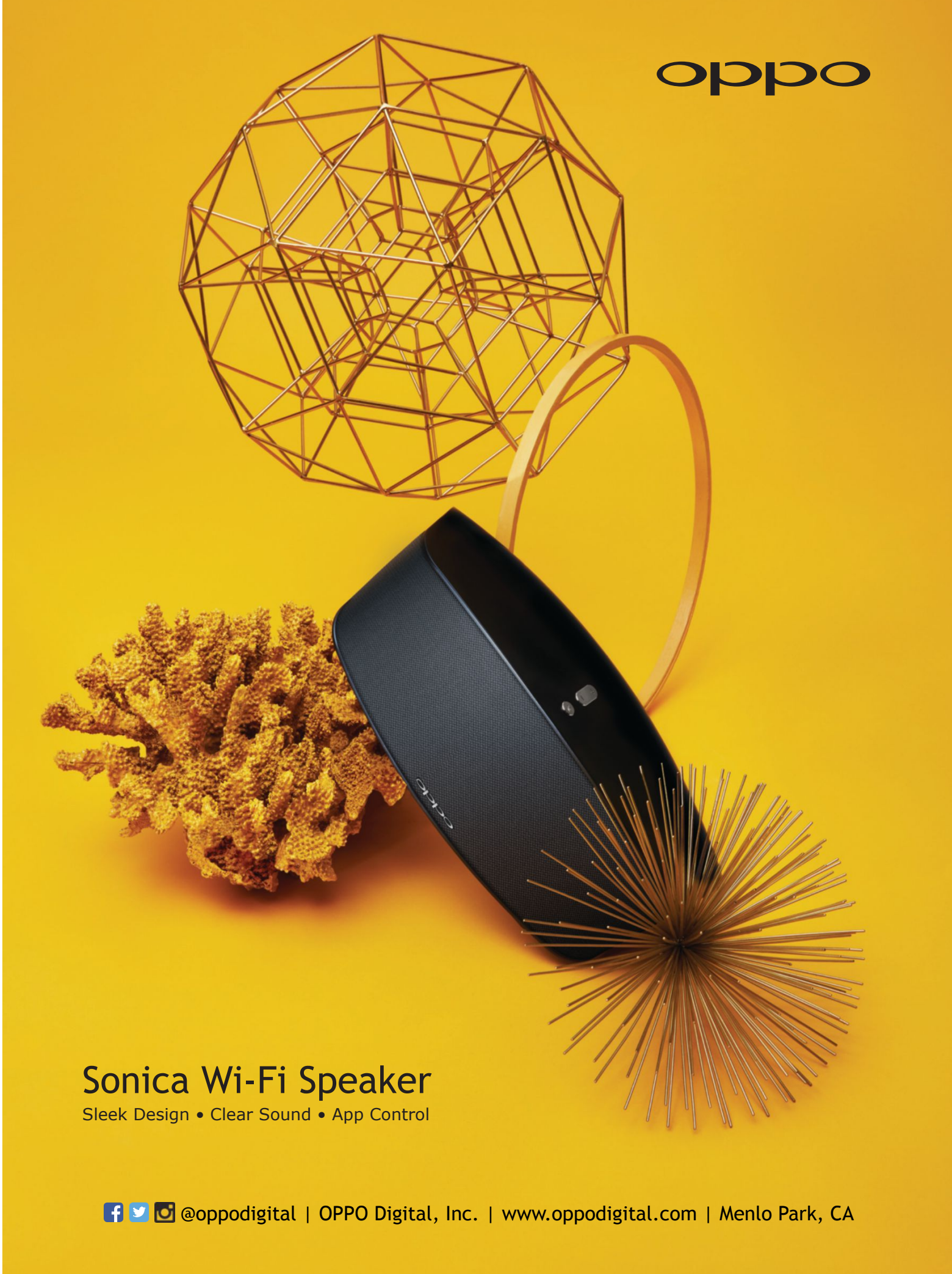
The MusicScope analyzer implements European Broadcasting Union (EBU) recommendation EBU-R128 and the ITU-R BS-1770 standard loudness measurement algorithm. Although PAZ uses these standards, it is also designed to compensate for the way we actually hear sound—the psychoacoustics. Thus, the two sets of values do not necessarily correlate. The importance is the difference between the values in each column.

The MusicScope screenshot shown in **Photo 6** details the short-term (3 seconds) loudness as a function of time in the vertical graph at screen center, labeled S-Mode. This is a useful and fairly inexpensive program, about $28. I've included a copy of the User Manual in the Supplementary Material available on the audioXpress website.

There is at least one DAW plug-in that can be used for real-time loudness equalization: the LC1 from www.magix-audio.com/us/lc1. Its price of

WAV File	MusicScope (Integrated Loudness)	PAZ Analyzer (Integrated Psychoacoustic Loudness)
Track-01	-30.2 dB	-16.4 dB
Track-02	-23 dB	-9.9 dB
Track-03	-39.4 dB	-28.1 dB
Track-04	-24.6 dB	-9.3 dB
Track-05	-24.9 dB	-11 dB
Track-06	-27.8	-7.8 dB

Table 2: Here we have the integrated loudness values for the first six tracks of a TIDAL *Classical Relaxation* playlist. As the numbers indicate, listening is not very relaxing because the playback volume is not uniform. Because of the difference between analyzers, the two columns do not necessarily correlate but both indicate a large loudness variation.


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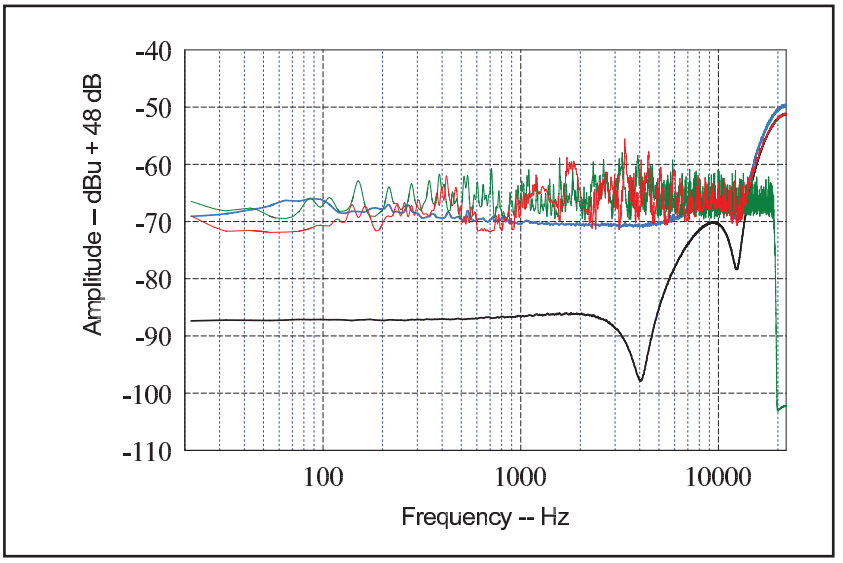


Figure 1: This graph of the four different files from Table 1 show inaudible levels, in all cases below -100 dBu. Each file was amplified by 48 dB but no filtering or smoothing was applied.

$1,359 probably makes it impractical for "home use," but it would be practical for streaming services although many instances of it would be needed to handle all the simultaneous streams.

I have read that, in earlier days, Spotify used optional loudness normalization. However, it was done incorrectly with dynamic compression, which sometimes added distortion. Spotify removed it, updated the loudness processing algorithms, and added it back as an option that you can disable in Preferences/Advanced Settings: "Set the same

Project Files

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

Resource

R. Tipton, "Signal Dynamic and Loudness," *audioXpress*, May 2015.

Sources

TIDAL
Aspiro AB | tidal.com

L-Normalizer
BeDSP | bedsp.net/l-normalizer

Audio DiffMaker
Liberty Instruments, Inc. | www.libinst.com

SoundTap
NCH Software | www.nch.com.au/soundtap

PSI-Plot
Poly Software International, Inc. | www.polysoftware.com/plot.htm

LCn
TC Electronic | www.tcelectronic.com/support/software

PAZ Psychoscoustic Analyzer
Waves Audio, Ltd. | www.waves.com/plugins/paz-analyzer

Qobuz
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MusicScope
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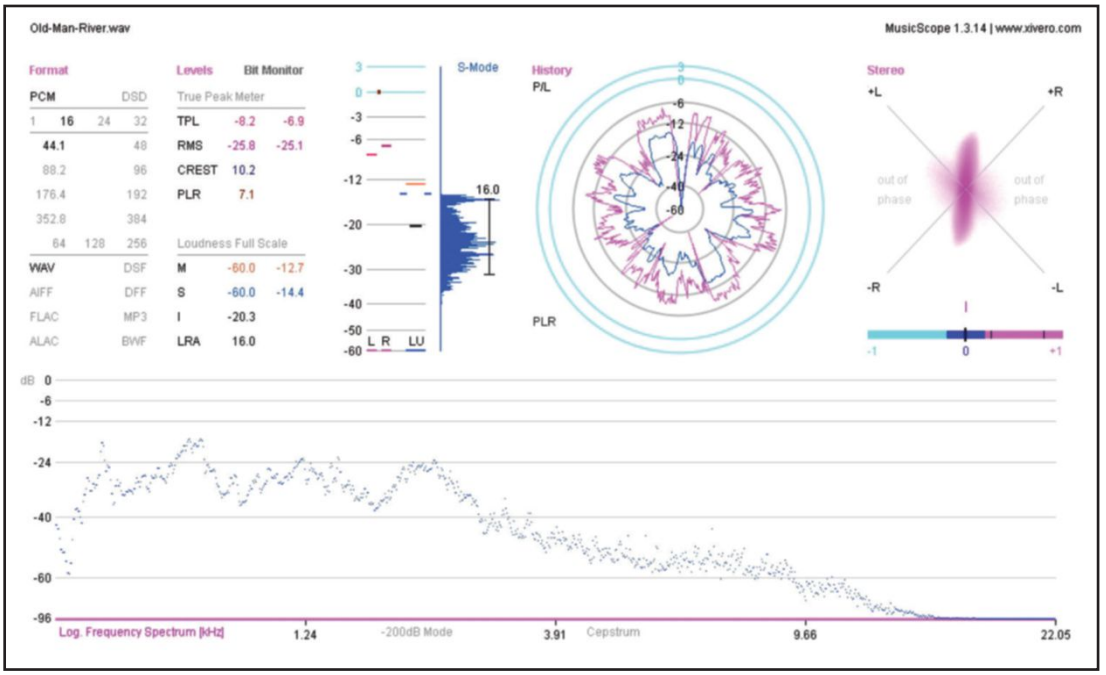


Photo 6: This MusicScope screenshot of "Old Man River" displays a lot of information. Of chief interest here are the loudness measurements, which include Momentary, Short-term (3 seconds), and Integrated over the entire file. The top center graph includes the vertical S histogram, blue in this color reversed photo.

volume level for all songs." I will evaluate the Spotify streaming experience in an upcoming article in this series. Currently, several music streaming services, including Google's YouTube videos, apply loudness to discourage producers from continuing to engage in the so-called "loudness wars."

I also found two loudness normalization programs that could be suitable for home use: L-Normalizer (a DAW plug-in) and LCn (a stand-alone). They don't do real-time normalization, but you could use either to more-or-less automatically produce your own normalized playlist.

A review is beyond the scope for this article series, but I'm considering a "Loudness Revisited" article. In the meantime, I have included user guides for both programs in the Supplementary Material.

Dynamic Range

Figure 2 shows a plot of the positive-peak dynamic range of the four tracks listed in **Table 1**. The figure was created using the dynamics program I wrote and described in my article, "Signal Dynamics and Loudness" (*audioXpress*, May 2015).

Two tracks show slight "squiggles," but overall the dynamics look good. The tracks also sound good so my conclusion is: TIDAL appears to stream its source material without any processing.

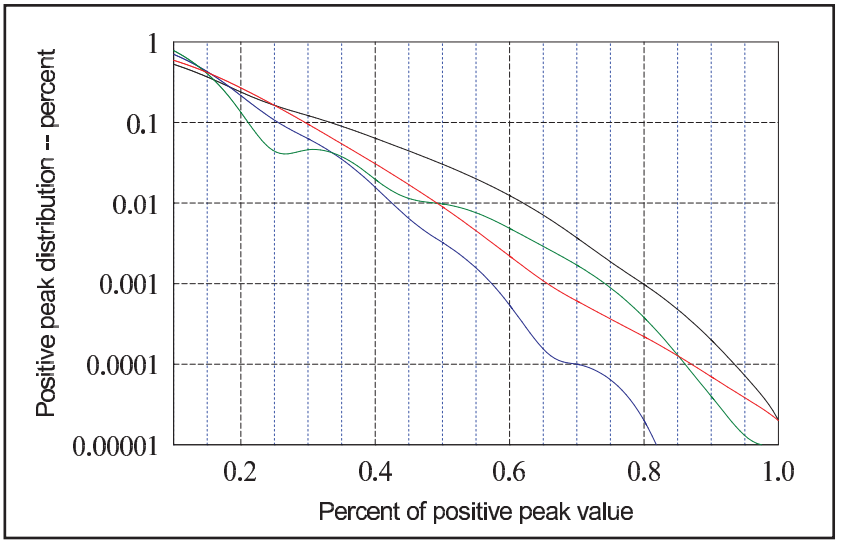


Figure 2: This graph shows generally good dynamics of the four tracks listed in Table 1. "Desmond Blue" is the black line, "Memory" is the blue line, "Old Man River" is the green line, and "S Wonderful" is the red line.

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— Brad Divens, *FOH Enrique Iglesias*

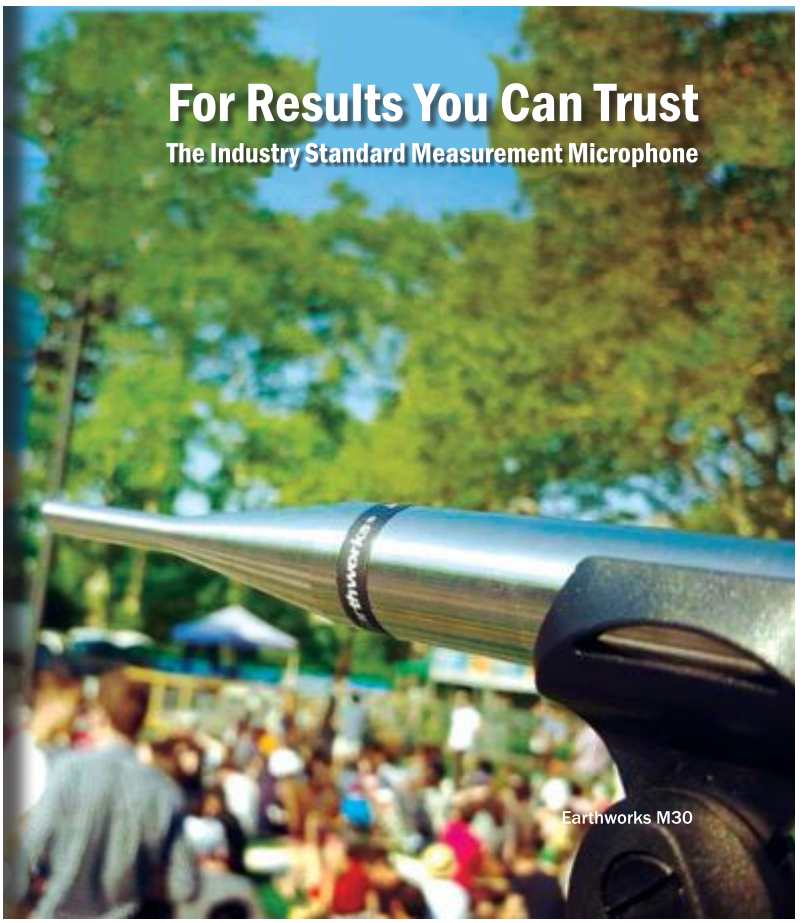
"The Earthworks M30 measurement mic is the first piece of gear I grab for when I'm ready to tune my P.A. system. I know its frequency response is very accurate and will tell me exactly what's happening in the room."
— Rick Camp, *FOH Jennifer Lopez*

"Every show I've mixed for Umphrey's McGee is available for download. That requires accurate tuning for my mix to translate to the recording. The M30 is the most accurate way I can measure that."
— Chris Mitchell, *FOH Umphrey's McGee*

"I've been using the M30 all around the world this year and love it. It's incredibly accurate and dependable."
— Benjamin Adamson, *FOH Kamasi Washington*

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Community Noise Effects and Measurements



Learn more about community noise, which is often ignored during the design phase of public spaces, and its impact on acoustics.

By
Richard Honeycutt
(United States)

When most people hear the word "acoustics," they usually think of sound in buildings. Unfortunately, in my experience, the majority relate acoustics only to sound-amplifying systems! I say "unfortunately" because the general public's ignorance of acoustics, outside of electro-acoustics—a word absent from most people's vocabulary—leads to the abysmal situation we face in many restaurants, libraries, school cafeterias, airports, bus and train stations, and other public spaces, not to mention venues supposedly intended for listening! Sound gurus Carolyn and Don Davis have authored a book whose title captures this sorry state of affairs: *If Bad Sound Were Fatal, Audio Would Be the Leading Cause of Death* (AuthorHouse, 2004).

This month's Sound Control article addresses only one aspect of acoustics that is too often ignored: community noise. The consideration of community noise should be a high priority for all city planners and owners of residential and commercial properties. In the US, NASA sponsored a study on transportation noise pollution completed by students and faculty at North Carolina State University in the early 1970s. Now—some 40 years later—few of its recommendations have been implemented. However, the Environmental Protection Agency (EPA) and the Federal Aviation Administration (FAA) have conducted studies specific to certain types of transportation noise in the years since then (see https://catalog.gpo.gov).

In 1999, the World Health Organization (WHO) published *Guidelines for Community Noise*, edited by Birgitta Berglund, Thomas Lindvall, and Dietrich H Schwela. In this publication, community noise is defined as noise generated outside the workplace. I would add, "and/or outside the domicile." *Night Noise Guidelines for Europe* was published by WHO/Europe in 2009.

Health Effects

Aside from legal requirements controlling community noise, residents should be concerned about the health effects of noise. Aside from possible hearing damage, sound has been found to lead to two other classes of response in humans: stress and annoyance. The stress response is characterized by a possible startle reaction (especially if the sound is loud, unexpected, or unpredictable). Physiologically, the stress response results in release of cortisol into the bloodstream, causing vasoconstriction and consequent increase in blood pressure.

A meta-analysis of 43 studies of the relationship of noise to ischemic heart disease performed by Elise van Kempen, et al, in 2002, showed a significant increase in systolic blood pressure in noise-exposed subjects. [1] The WHO reported a correlation between long-term sound exposure above 67 to 70 dBA and hypertension. More recent studies suggest sound levels above 50 dBA at night can increase risk of myocardial infarction (heart attack). Other studies have emphasized the

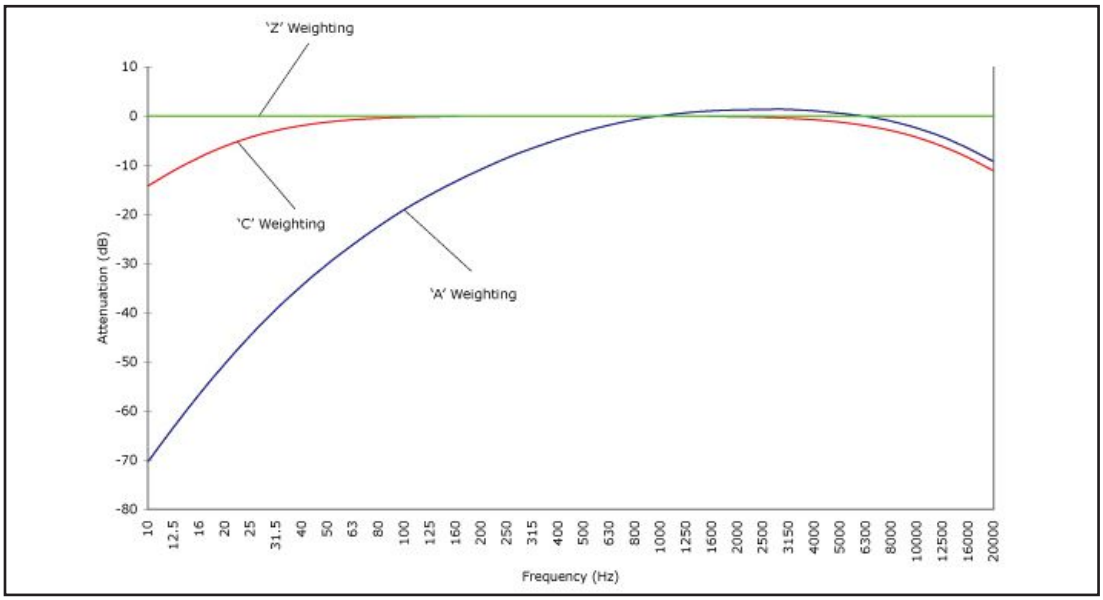


Figure 1: Sound level meters usually allow a choice among three frequency weightings.

negative effects of sleep disturbance caused by sound.

An extreme stress response is observed in sensitive persons such as children or those with post-traumatic stress disorder (PTSD), in which the activation of the "fight-or-flight response" can be severe. Such effects have also been documented in wild and domestic animals.

In both the physiologically and the psychologically mediated responses, documented consequences include decreased cognitive performance, inability to concentrate, and decreased creativity. Persons with latent mental/emotional instability may become destabilized as a result of sound-induced stress.

People can become accustomed to some sounds, with a consequent reduction in stress effects, but stress reaction to other sounds, notably gunshot sounds, cannot be ameliorated by repeated exposure. Ragner Rylander has determined that involuntary muscle contractions indicative of the "fight-or-flight response" have been observed even among policemen who regularly train with guns and are accustomed to the sound.[2] This same research indicates that a person's attitude toward the type of sound has little if any effect upon the sound-induced stress response.

Most studies of the non-auditory effects of sound have concentrated on annoyance more than on physiological responses. The assessment instruments used in sound-annoyance research typically ask for yes/no answers to the question, "Do you find this sound very annoying?" The responses are highly variable from person to person, with some people finding sounds as low as 45 dBA very annoying, and others having annoyance thresholds as high as 75 to 80 dBA.

Reasons for this variability include the use of hearing aids by some listeners and the personal attitude toward the type of sound. Hearing-aid users often experience increased difficulty with speech comprehension in the presence of noise, as compared to those who do not use hearing aids.

Although attitude toward the type of sound does not affect the physiological response, it has a strong effect on the annoyance experienced by a given listener. For example, a classical music lover may be more annoyed by the presence of a loud rock music club next door than would a person who enjoys rock music. Or a person who is accustomed to a quiet rural environment may be more annoyed by gunshot sounds than would an inner-city gang member.

Sound Level Guidelines

In order to provide usable guidelines on sound levels for land-use compatibility decisions, researchers and governmental agencies have focused on sound-level limits that represent an average response in terms of annoyance, knowing there will be listeners who will be annoyed even at the permissible sound level.

Community noise is generated by several important sources, some of which are under human control. Weather can generate significant noise, as can automobile, train, or airplane traffic. Construction and industrial equipment, and even large HVAC chillers can create unacceptable levels of noise. Outdoor shooting ranges and amphitheaters are common causes of noise complaints, and in rural areas, events such as tractor pulls and "mudding" with trucks or ATVs must be considered.

Because of the great variability among individuals as to annoyance stemming from a certain level of sound, much research has been done in an attempt to quantify the relation between sound level and annoyance. For most assessments, the



Photo 1: The decibel screen can show the values and the waveforms corresponding to the minimum, average, and maximum values of the noise.

US federal government uses guidelines developed by the EPA in 1974 and reaffirmed by the Federal Interagency Committee on Urban Noise (FICUN) in 1980 for land-use compatibility considerations. These guidelines specify a sound level of 65 dBA (average day-night level) as the maximum that is compatible with residential areas. The National Rifle Association's (NRA) guidelines for firing ranges specify that sound levels on adjacent properties are acceptable if the sound levels at the receiver (person hearing the sound) do not exceed 65 dBA for more than 8 hours out of a 24-hour period. The NRA guidelines also stipulate that "shooting activities should not continue into the nighttime hours, between 10:00 PM and 7:00 AM, unless needed for mandatory low light training by law enforcement personnel."

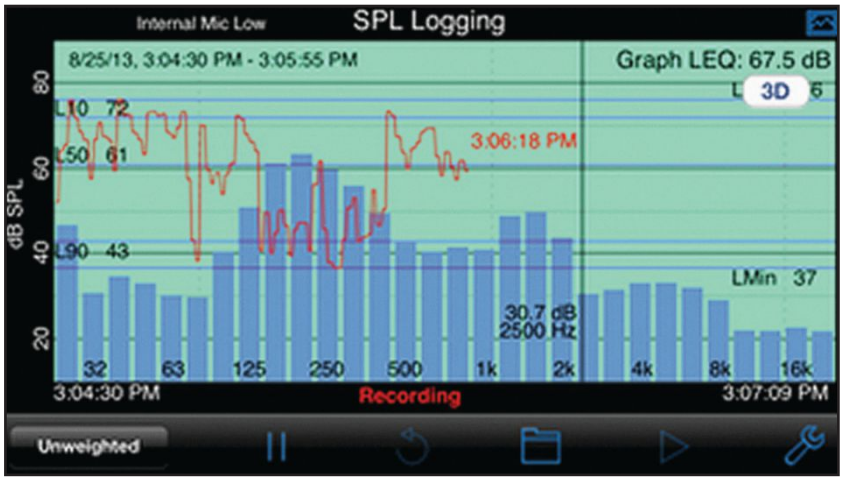


Photo 2: The Studio Six apps provide more flexibility and features for the professional user.

The 2007 Army regulations specify a maximum peak sound level of 87 dB for residential areas. Note that unlike many urban noise regulations that specify maximum sound level at the property boundary, the EPA, the US Army, and the NRA guidelines specify sound on the nearby owner's property, not simply at the boundary of the property on which the sound is being generated. Researchers generally recognize that certain individuals will find specific types of noise annoying at peak levels lower than 87 dB, and therefore, realize that the guidelines are, at best, a convenient approximation of good practice in land-use compatibility decisions.

Controlling community noise is best commenced at the design stage for the possible generators of the noise, and also for residential and commercial buildings in which people must be protected from noise annoyance. Unfortunately, all too often community noise problems are addressed only after they have begun to cause annoyance. In either case, preventing or correcting a community noise problem often necessitates careful measurements. These measurements must be made using appropriate equipment, and in some cases, may require a schedule of measurements spanning hours, days, weeks, or even months, to account for diurnal, seasonal, or other time-related effects of noise generation and propagation.

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[1] E. van Kempen, et al, "The Association Between Noise and Blood Pressure and Ischemic Heart Disease: A Meta-analysis," *Environmental Health Perspectives* 110 (3), March 2002, pp. 307-317.

[2] R. Rylander, "Noise, Stress, and Annoyance," *Noise and Vibration Worldwide*, June 2006, p.p. 9-13

Resources

C. and D. Davis, *If Bad Sound Were Fatal, Audio Would Be the Leading Cause of Death,* AuthorHouse, 2004.

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R. Honeycutt, "Smartphone Apps for Sound Measurement," *audioXpress*, November 2015.

Night Noise Guidelines for Europe was published by WHO/Europe, 2009.



Photo 3: These sound level meters are among those that are suitable for professional use. (Photos courtesy of Studio Six Digital; PCE instruments; Casella CEL; NTi Audio; and Brüel & Kjær)

Most community noise measurements only address sound level. There are several metrics for sound level, including:

- Instantaneous sound pressure level: L_p
- Equivalent continuous sound pressure level: L_{eq}
- Sound exposure level: L_E
- Maximum sound pressure level: L_{max}
- Minimum sound pressure level: L_{min}
- Percentile sound levels: L_N
- Time- and frequency-weighted noise: for example, L_{AF} is an A-weighted, FAST noise level

All sound level measurements are averaged over a certain length of time. For years, the standard averaging times have been designated as SLOW, FAST, or IMPULSE. The latter is not a true instantaneous peak measurement, but is averaged over a shorter time than FAST. When L_P is displayed, it does not indicate a different averaging time. It is the maximum of a series of repeated measurements.

Most sound level meters (SLMs) offer A, C, and Z frequency weightings. As you can see in **Figure 1**, the Z weighting is actually flat—no weighting at all. The A and C weightings cut low- and high-frequency contributions to roughly correspond to the frequency response of human hearing for sound fields of 40 and 60 phons, respectively.

In addition, SLMs can be type-approved for accuracy using International Electrotechnical Commission (IEC) [in French: Commission électrotechnique internationale (CEI)] class or American National Standards Institute (ANSI) type:

- IEC Class 1 SLMs have tighter accuracy requirements for midrange accuracy and frequency-response flatness than Class 2 units.
- Similarly, ANSI type 0 SLMs have the tightest tolerances, and are for laboratory use. Type 1 units meet somewhat less stringent requirements and are for precision measurements in the field, and Type 2 SLMs are for general-purpose use.

SLMs having no Class or Type designation are not suitable for professional use. Often, verifying conformance to a legal or contractual requirement will necessitate the use of a specific Class or Type of SLM.

The most versatile SLMs include features useful to professionals involved in community noise:

- Data memories
- Provision for recording .wav files of the noise
- Provision for annotating measurements by either speech or text entry
- Ability to automatically repeat and store measurements at specified intervals.
- SD card storage to permit user choice of the maximum number of measurements whose results can be stored
- Triggered or delayed start
- Correction for diffuse fields and for windscreens
- External software to allow FFT and/or octave-band analysis
- Ability to be connected to an external GPS to record the latitude and longitude of the measurement site

The least expensive way to measure sound level is by using an app on a smartphone. Sample screens from one such app are shown in **Photo 1**. However, as discussed in the Sound Control article that appeared in the November 2015 issue of *audioXpress*, many of these apps are suitable for hobbyists only.

Photo 2 shows a representative screen from a Studio Six Digital smartphone app that provides professional-level functionality. When used with a professional measurement microphone (e.g., the Studio Six usbPrecisionMic), this app can make measurements to ANSI Type 1 standards.

Photo 3 shows five SLMs that are suitable for professional use. The unit at left is shown with the manufacturer's optional ANSI Type 1 microphone. These instruments span quite a broad range of prices and feature sets.

Often community measurements are made outdoors. In such cases, microphone accessories such as bird spikes, windscreens, or nose cones may be needed (see **Photo 4**).



Photo 4: This outdoor microphone includes several forms of environmental protection. (Photo courtesy of NTi Audio)

Fresh From the Bench

Audio Precision's APx1701 Transducer Interface

Learn more about Audio Precision's APx1701, which when combined with an APx500-series analyzer, creates a test center suitable for measuring speakers, headphones, and microphones, as well as electronics.

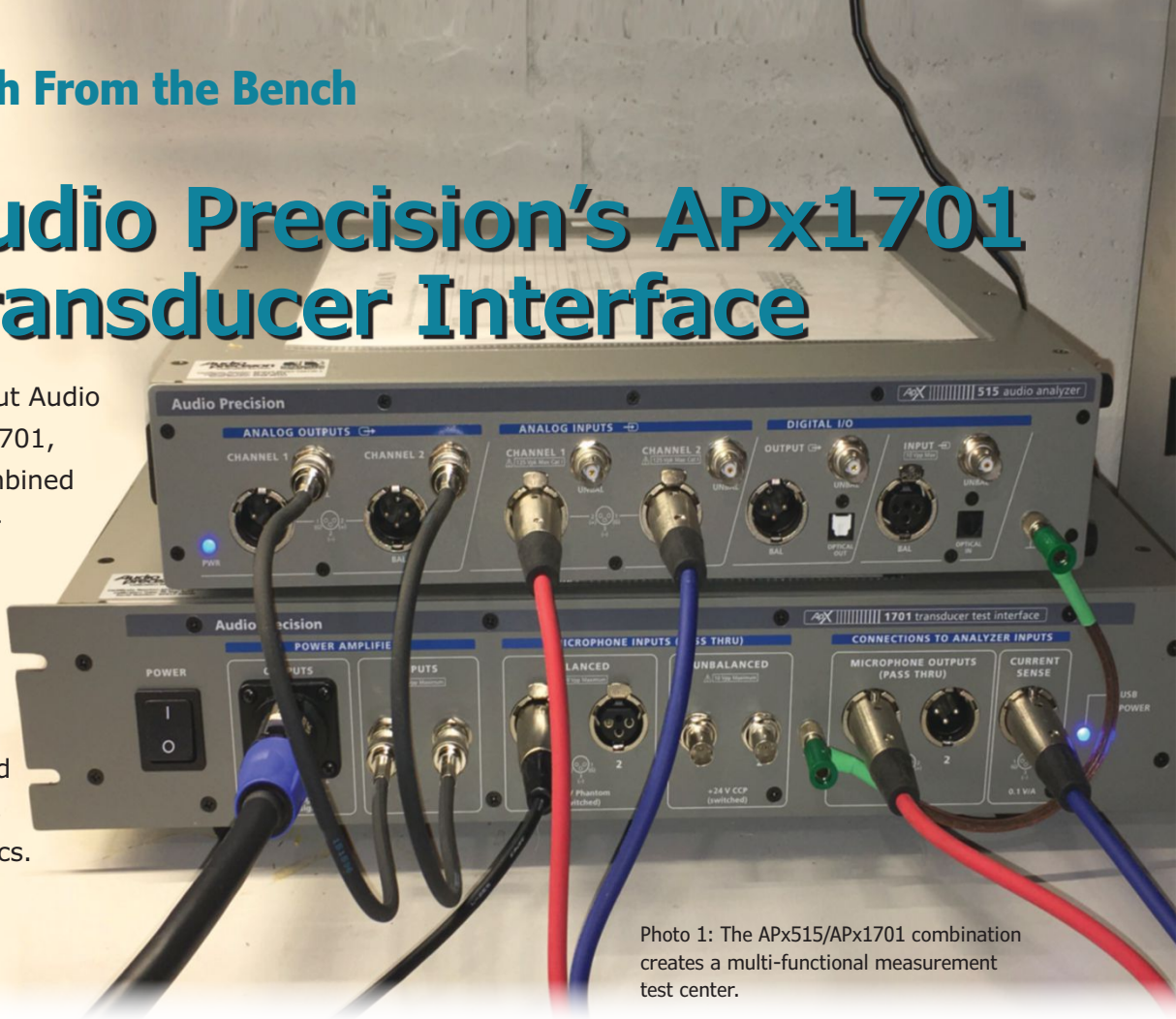


Photo 1: The APx515/APx1701 combination creates a multi-functional measurement test center.

By
Stuart Yaniger
(United States)

Without question, the traditional gold standard in audio test gear has come from Audio Precision, based in Beaverton, OR. Its test gear has been the go-to choice for the past 30 years, and any serious engineer has made extensive use of the equipment for characterizing the electronics end of the audio production and reproduction chains. Audio Precision's founder, Bruce Hofer, is one of the best creative minds in audio test and measurement.

The trade-off for performance has, as expected, been complexity and difficulty. Although I've never personally owned a System One or a System Two, my friends who have them universally praise the performance and universally condemn the difficulty of setup and use.

Audio Precision's next generation of gear, the 2722, has improved in both areas, but in return it is an expensive and complex piece of equipment, suitable for setup by engineers comfortable with GPIB interfacing or LabVIEW. Audio Precision introduced the APx500 series as a lower cost alternative with improved user-friendliness, and my experience has borne this out—in less than an hour I had unpacked the APx515 and was getting good measurements.

Although acoustic measurement is built into the APx500 software, in addition to an APx515 (or other analyzers in the '500 product line), the user needs to supply a power amplifier to drive the speakers or headphones under test and microphone preamplifiers. To perform impedance measurements, a series resistor and differential amplifier are also needed. Each of these must be accounted for in the calibration process, and any adjustments of volume using their controls will require that the calibration be rerun. And of course, low-noise and low-distortion electronics for the microphone and loudspeaker end of the measurement are not inexpensive.

As a result, most engineers I know keep two separate measurement systems on hand, one for electronics and one for transducers. While effective, this is redundant, clumsy, and unnecessarily expensive. From my point of view, a major stumbling block is having to learn the peculiarities of two separate software systems.

Meet the APx1701

To fill this gap, Audio Precision has introduced the APx1701 Transducer Interface. In combination

with an APx500-series analyzer, the APx1701 creates a test center suitable for measuring speakers, headphones, microphones, and electronics (see **Photo 1**).

The APx1701 comprises two XLR balanced microphone inputs with 48 V phantom power (note that these are pass-through inputs, i.e., they will provide phantom power, but send the signal directly to the APx500-series analyzer's analog inputs for amplification) and two unbalanced BNC microphone inputs set up as 4 mA constant current for powering microphones such as the PCB Piezotronics microphones I recently reviewed (see *audioXpress,* December 2016) or Audio Precision's new line of test microphones.

The microphone inputs are TEDS-compliant and will read identification and calibration data from IEEE1451.4-compatible microphones and pass it to the APx500 software. The microphone preamp pass-through output is a balanced signal sent to two XLR jacks, irrespective of whether a balanced microphone or a constant current microphone is used.

The APx1701 also comprises two medium power (60 to 100 W, depending on the load) low noise and low distortion amplifiers. The amplifiers have unbalanced BNC input jacks, and output is taken through a SpeakOn connector. Their specified gain is fixed at 20 dB, and current is limited to 6 A. Rated signal-to-noise is 134 dB, referred to the maximum 30 V output. Total harmonic distortion plus noise (THD+N) is specified at -106 dB from 10 Hz to 20 kHz, but the specification does not mention the power level for that rating.

An additional feature of the APx1701 is a small (100 mΩ) resistor in series with the (-) power

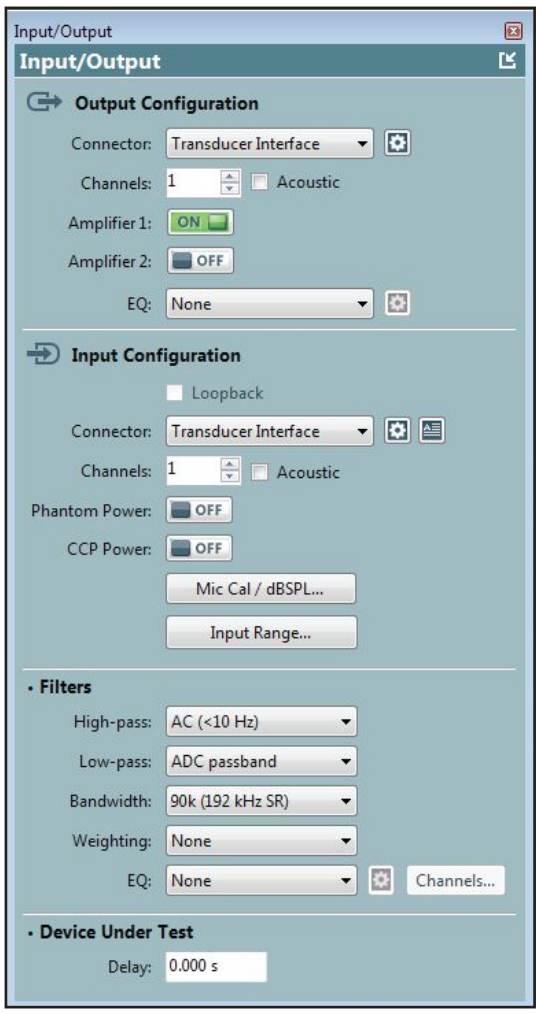


Figure 1: The inputs and outputs are controlled by a selection screen using drop-down menus.

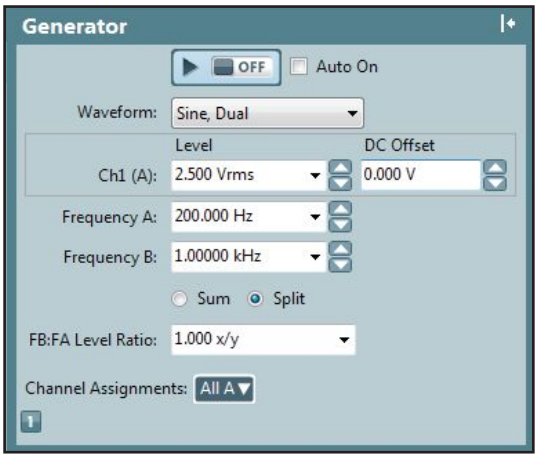


Figure 2: The Generator screen enables the user to select waveforms, levels, and channel assignments for the APx1701 integrated with an APx500 series analyzer.

Parameter	Value	Lower Limit	Upper Limit
F_S	47.02 Hz	----	----
Q_{MS}	4.74	----	----
Q_{ES}	0.69	----	----
Q_{TS}	0.60	----	----
S_D	226.98 cm²	----	----
R_E	8.45 Ω	----	----
L_E	0.66 mH	----	----
R_2	---- Ω	----	----
L_2	---- mH	----	----
K_{rm}	----	----	----
E_{rm}	----	----	----
K_{xm}	----	----	----
E_{xm}	----	----	----
R_{MS}	1.58 Ns/m	----	----
C_{MS}	0.45 mm/N	----	----
M_{MS}	25.41 g	----	----
V_{AS}	32.96 l	----	----
Bl	9.60 Tm	----	----
η_o	0.47 %	----	----

Figure 3: The APx1701 allows rapid acquisition of Thiele-Small (T-S) parameters. The data is for a 170 mm driver under test, using the added mass method.

Fresh From the Bench



Photo 2: I used the APx515/APx1701 combination to take a nearfield measurement of a 170 mm woofer.

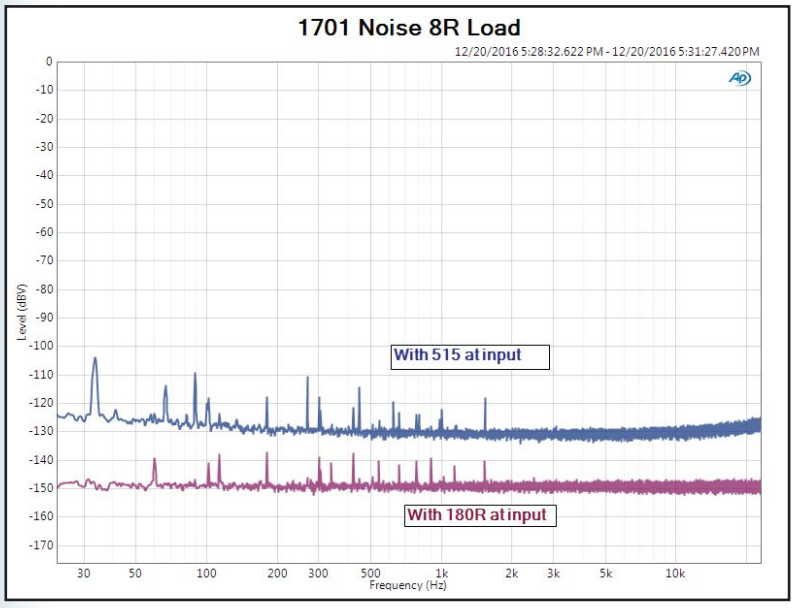


Figure 4: The upper curve of this graph shows the noise floor of the APx515/APx1701 combination and the lower curve shows the APx1701 alone, with the APx1701 terminated in an 8R (8 Ω) load.

amplifier output—this resistor develops a voltage across it proportional to the amplifier's output current. A built-in differential amplifier measures the voltage across this resistor and sends it to an XLR jack on the front panel marked "Current Sense." The Current Sense is most commonly used for impedance measurements, but can also be used for any other sort of measurement where current to the transducer needs to be analyzed. The retail price of the APx1701 is approximately $3,600.

Integration

Figure 1 shows the input and the output selection screen. When "Transducer Interface" is chosen, the software automatically accounts for the APx1701's power amplifier gain of 20 dB when driving its input and looking at the output. It can also turn each amplifier channel on and off. Likewise, when the output voltage is set from the Generator screen (see **Figure 2**), the software adjusts the analyzer output to get the correct voltage at the output of the APx1701.

At the input side of things, the microphone inputs can be selected (unbalanced constant current or balanced) and the phantom power turned on and off via the selection screen. Microphone calibration and sensitivity can be input through a menu choice on the Input Configuration screen so that sound pressure levels can be shown in decibels as well as volts.

The software will enable selection of either acoustic or impedance measurements. There are two basic software modes—Bench and Sequence. The former is, unsurprisingly, aimed at engineering and development work, the latter targets production testing. For Sequence, you can program in Pass/Fail criteria, enabling the instrument to be used by non-engineering personnel on a production floor. The two environments provided by these modes seems intuitive, but there's a few examples of where placement is (to me) non-intuitive. The most glaring is impedance measurements, which seem like something useful for bench testing, but are actually part of the Sequence environment. This caused me some head-scratching until I asked Audio Precision's tech service to take pity on me. They patiently walked me through this and were nice enough not to mention that I could have figured this out for myself if I had just consulted the manual.

Once that was explained, I was getting accurate impedance measurements in minutes. The APx515/APx1701 combination made obtaining Thiele-Small (T-S) parameters a snap—in Sequence mode, of course! You can use either the added mass or

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known-volume box method, and the entire process is two button pushes—one free air and one in the box (or one without and one with added mass). **Figure 3** shows an example output screen for a high-quality 170 mm driver that I have been testing for use in my next speaker design—this was obtained using the added mass method. The test setup is shown in **Photo 2**.

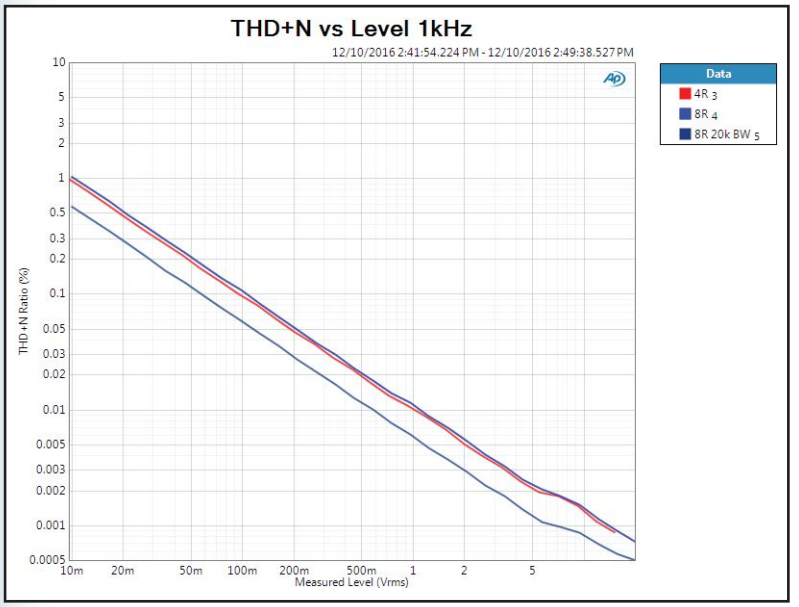


Figure 5: The THD+Noise of the APx515/APx1701 combination is extremely low and noise dominated. The upper curves are the THD+Noise vs. level for 4 Ω and 8 Ω loads, 80 kHz bandwidth, the lower curve is the same measurement for an 8 Ω load with a 20 kHz bandwidth.

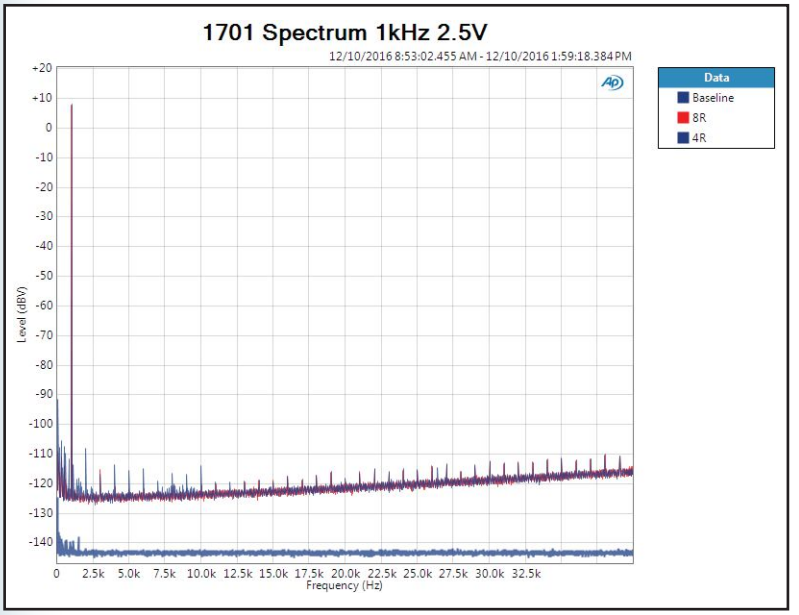


Figure 6: The distortion spectra of the APx515/APx1701 combination at 2.5 V into 4 Ω and 8 Ω loads are impeccable. The lower curve is the noise floor for the measurement.

Performance

The test setup to measure the APx1701's performance is fairly easy. I connected the APx1701 to my lab computer (a Hewlett-Packard desktop running Windows 7) via the USB port. Likewise, I connected the APx515 to the same computer, and used the supplied BNC-terminated coaxial cables to connect the APx515's outputs to the APx1701's inputs. Then, I connected the APx1701's output to a dummy load made from a 4R (4 Ω) or 8R (8 Ω) 100 W non-inductive power resistor, depending on the load measurement. I connected the APx515's input across the load using a 1× oscilloscope probe.

First, I measured the noise floor with the Y axis units being decibel level referenced to 1 V (dBV) and the output of the APx1701 connected to the 8 Ω dummy load (see **Figure 4**). The upper trace shows the noise across the dummy load with the APx515 connected, but with the signal generator turned off. The lower trace shows the noise across the dummy load, but with the input of the APx1701 shorted with a 180 Ω resistor.

These spectra show that the APx1701's noise will not be the limiting factor in a measurement, and even with the increased noise floor from connecting the APx515 analyzer, the noise is orders of magnitude below anything that is likely to be acoustically measured. Note that the 20 dB difference in the noise floors (-130 dBV vs. -150 dBV) corresponds to the 20 dB voltage gain of the APx1701.

There's a few small spikes in the spectra, probably a mix of internally generated noise from the digital circuitry and pickup from my not-at-all-quiet lab environment. That said, the noise floors are quite low, and even the worst of the spikes doesn't exceed -105 dBV (with the APx515 connected).

Figure 5 shows the THD+Noise vs. level for both the 4 Ω and the 8 Ω loads. These measurements were taken at 1 kHz, but 10 kHz measurements looked nearly identical. The top two traces were taken with an 80 kHz measurement bandwidth, the lower trace was taken with a 20 kHz bandwidth. The difference in level between these traces confirms that the measurement is dominated by noise. You can also see that the built-in protection circuit limits the voltage swing at the 4 Ω load impedance well before any distortion appears—measurements right up to the power limits of the APx1701 will be reliable.

Figure 6 shows the distortion spectrum at 2.5 V output and a 1 kHz fundamental into 4 Ω and 8 Ω loads. The 4 Ω load results show some even-order distortion not present for the 8 Ω load, but even so, at -116 dB down from the fundamental (0.00016%), the distortion is ridiculously low.

Figure 7 shows a log frequency plot of the distortion spectrum for a 2.5 V output and a 10 kHz fundamental into an 8 Ω load. The distortion is similarly low and dominated by noise (all spikes in the spectrum that do not fall at multiples of the 10 kHz fundamental are noise). In most amplifiers, distortion rises with increasing frequency because of falling open loop gain and hence lower feedback with increasing frequency. The clever engineers at Audio Precision seem to have done some circuit tricks to overcome this.

The intermodulation distortion (IMD) results are similarly impressive. **Figure 8** shows the spectrum for a 10 kHz and 11 kHz tone pair at 10 V output into 4 Ω and 8 Ω loads. Intermodulation products appear at 1 kHz multiples, with the largest being at 13 kHz with a level of -125 dB (referenced to the signal for the 4 Ω load. Again, this is outstanding performance, with the distortion absolutely negligible compared to that of any transducer.

Example Applications

When the legendary speaker company NHT was going through one of its many corporate restructurings, I inherited a couple dozen Peerless 170 mm woofers in varying condition. I had tried using them in a few prototype speakers, but never got quite the results I wanted. The bass always sounded rather murky and uncertain. Of course, I had run measurements on their T-S parameters during development, using my sound card setup, and used these measurements to design the boxes. My measurements were repeatable, so I never understood why my results weren't better. Hating a mystery the way I do, the first thing I did with the APx1701 was re-run those measurements.

One of the cool features of the integration is that the amplifier output voltage can be selected from a drop-down menu—no knob-twiddling or external voltmeter measurements needed. So in less than two minutes, I was able to see how the impedance curve varied as a function of drive level, varying the drive from 250 mV to 2.5 V. The test setup was straightforward. I connected the output of the APx1701 to the woofer, and connected the Current Sense output of the APx1701 to one of the balanced inputs of the APx515 analyzer. The APx500 software then uses the current and applied voltage to calculate the impedance, which is then displayed.

The results are shown in **Figure 9**. As much as we always hear the disclaimer that T-S parameters are small signal, the level at which they are taken is not standardized, and this graph shows that indeed something as basic as resonant frequency is a strong function of drive level. With increasing

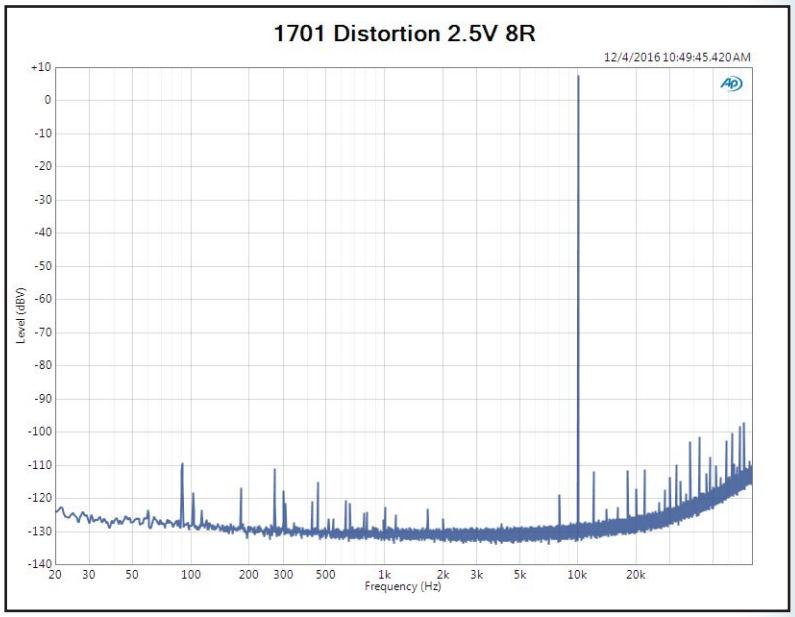


Figure 7: The distortion spectrum of the APx515/APx1701 combination at 10 kHz is dominated by low levels of noise—as with the 1 kHz results, the distortion is remarkably low.

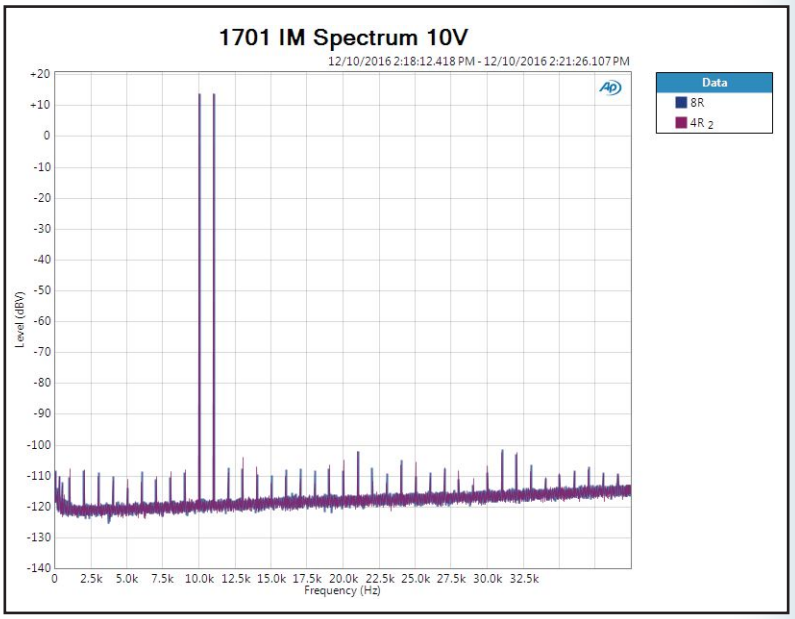


Figure 8: The10 kHz/11 kHz intermodulation distortion at 10 V output into 4 Ω and 8 Ω loads is astonishingly low, with the largest component at -125 dB (<0.0001%) relative to the signal.

Resource

S. Yaniger, "Innovative Measurement Microphones from PCB Piezotronics," *audioXpress*, December 2016.

Source

APx515/APx1701 combination
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drive, the resonant frequency drops significantly. Note that all of these voltage levels are relatively low, so the woofer is certainly not being stressed.

The shift in resonant frequency with drive is an indication of nonlinearity of the suspension (and possibly the motor), and it's evident that the frequency response of the driver in a box would shift around considerably when playing music. No wonder I was dissatisfied with the sound!

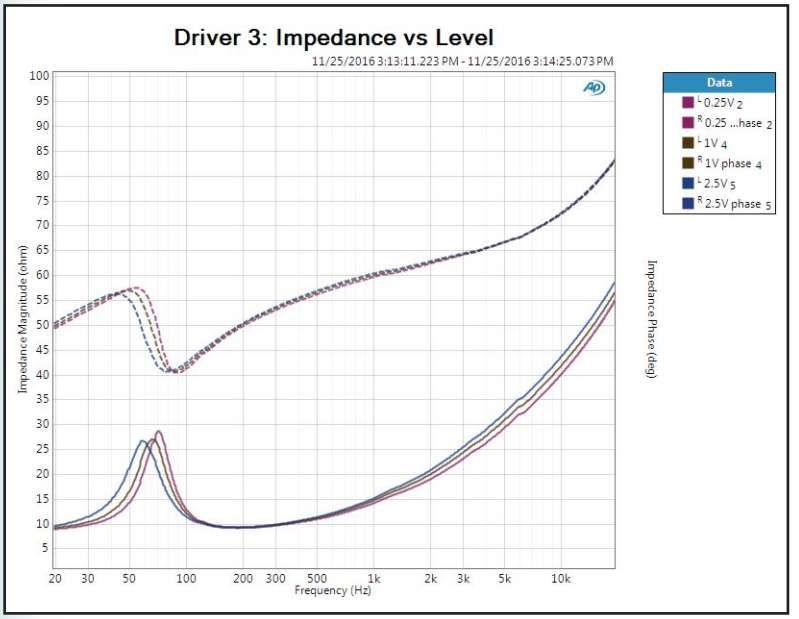


Figure 9: The impedance of a 170 mm Peerless woofer is shown for drive levels of 0.25 V, 1 V, and 2.5 V. Note the significant shift in resonant frequency with drive.

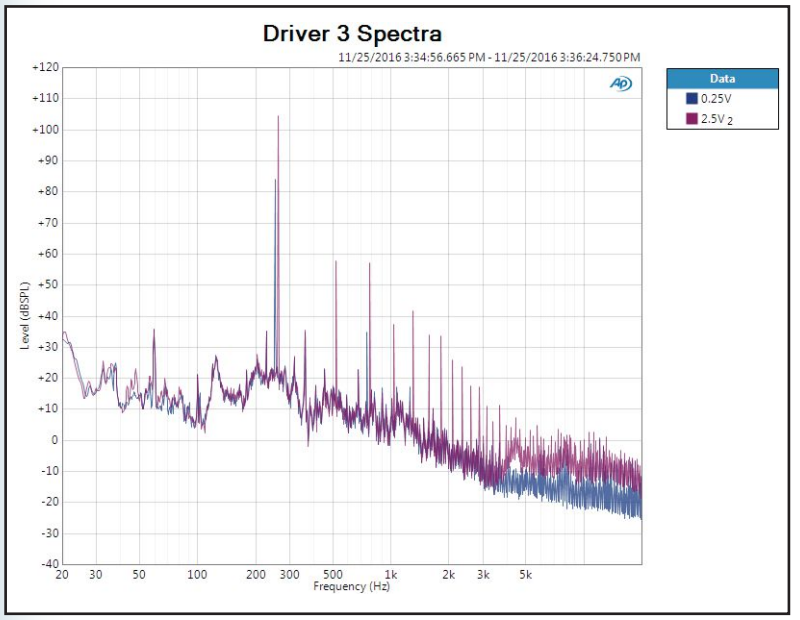


Figure 10: The distortion spectrum of the same 170 mm woofer from Figure 9 at two drive levels shows the distortion changes predicted from the resonant frequency shift.

An indication of this effect is evident in the distortion spectra, which are shown in **Figure 10** for two drive levels (250 mV and 2.5 V) at approximately 250 Hz. I altered the frequency slightly between the two different levels so that the spectra could each be seen and compared. The setup for this measurement was also straightforward. I connected the woofer to the output of the APx1701. Next, I placed my test microphone (an iSEMcon EMX-7501) about 3 cm away from the woofer cone and connected it to the APx1701's balanced microphone input, which provided the phantom power and passed the microphone signal to the APx515 analyzer.

Note that at 250 mV drive, the third harmonic is dominant at about -50 dB relative to the 250 Hz fundamental. Any symmetrical nonlinearity will have the odd-order harmonics predominate, and we can see that this is the case here. When the drive is increased to 2.5 V, the second harmonic at 500 Hz appears at a level comparable to that of the third harmonic, indicating an asymmetric nonlinearity (i.e., the cone's forward and backward motions are not equal). We can also see the appearance of other even harmonics (e.g., the fourth at 1 kHz) characteristic of an issue with the suspension or motor.

One might guess that this nonlinearity correlates with the previously noted shift in resonant frequency with drive, proving a purely electrical way to predict driver distortion. Using the Audio Precision system, I was quickly able to evaluate several different driver types to see if this is the case—I will elaborate on this in a future article.

I tested six of the Peerless drivers and found similar results. However, despite having frequency response, impedance, and distortion similar to the other woofers, Driver 3 subjectively sounded different and worse than the others—the difference wasn't huge, but it was noticeable. What could be the cause? My first step was to look at the impulse response of the drivers. I obtained this using the log frequency sweep as a stimulus and the same setup as described for the distortion measurements

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.

above. **Figure 11** shows the impulse responses of Drivers 1 and 3, with 2.5 V of drive signal. On casual inspection, they look pretty similar. To see if there are any issues with delayed energy, the impulse response was transformed to give the Energy Time Curve (see **Figure 12**). The drivers don't show much difference there, either. Quite a mystery!

Audio Precision also has a proprietary signal processing function to detect voice coil rub and buzz. Could this shed light on the mystery?

Figure 13 shows the rub and buzz measurements. We can see that Driver 3 has significant peaks at a series of discrete frequencies compared to Driver 1 (and indeed, all the other woofers tested), suggesting that the sonic difference I heard is related to mechanical issues in the woofer's motor. This function could be a very useful quality control tool when looking at a population of drivers.

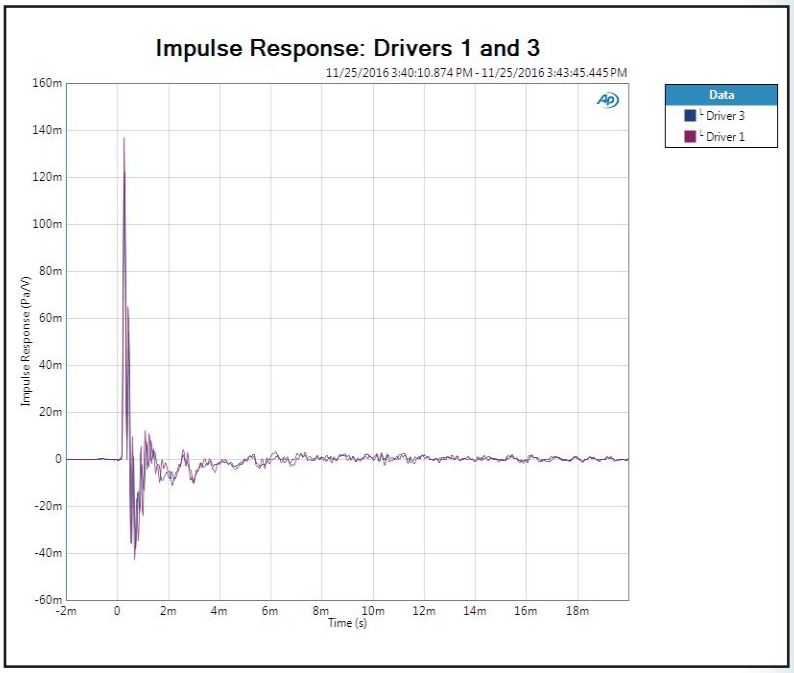


Figure 11: The impulse responses of two 170 mm woofers derived from a log frequency sweep do not show much difference between them, despite some differences in sound.

Negatives

My wife likes to say, "Perfection is a good starting point." In that spirit, there's still a few things I'd want to see improved. First, there's noise.

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Fresh From the Bench

Not electrical, which was quite low, but mechanical. The fan was loud enough to interfere with some low-level speaker measurements I was trying to do. The workaround is to use a long microphone cable and position the interface and the analyzer in a separate room, but this was rather inconvenient since my electronics test bench and speaker measurement area are in the same room.

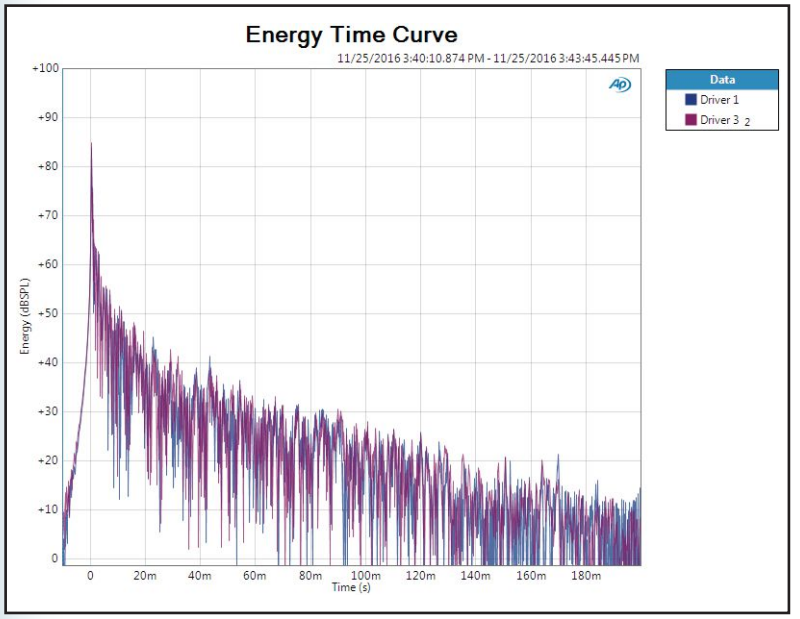


Figure 12: The APx515/APx1701 combination enables rapid acquisition of energy-time curves. This graph is derived from the impulse responses shown in Figure 11.

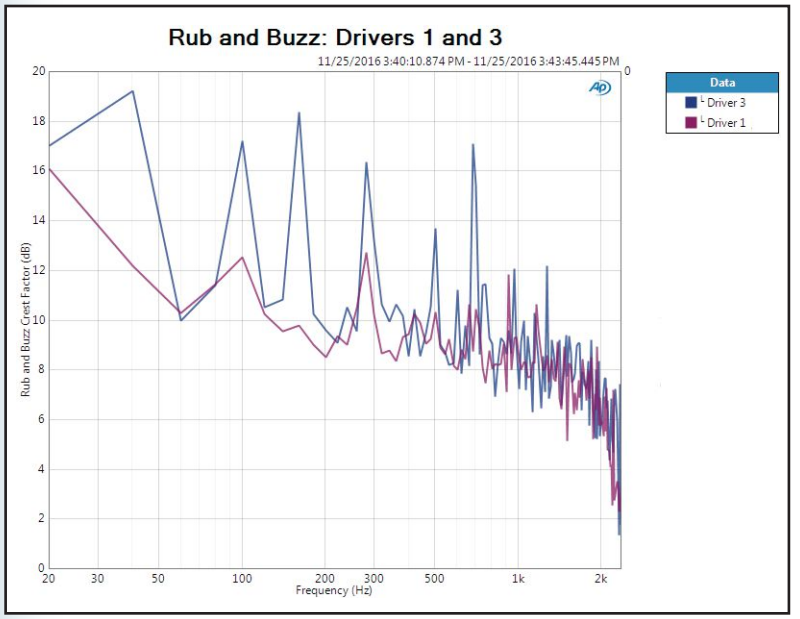


Figure 13: Audio Precision includes a proprietary rub and buzz measurement. These curves are taken from the same drivers as the previous two figures and graphically shows why Driver 3 sounded significantly worse than the others.

Second, it would be nice to have a balanced input option. The existing unbalanced connection through a BNC is fine if the unit is always getting its input from an APx500-series analyzer, but I can think of several potential measurement scenarios where a balanced input would be preferred. You could construct a simple balanced-to-single-ended converter, but I would hate to compromise this system's performance with more quotidian electronics in the signal chain.

Finally, documentation. The APx500 software's documentation is extraordinarily comprehensive (700 pages in the manual!) and clearly written. By contrast, the APx1701 comes with a rather thin pamphlet that gives some general suggestions, but doesn't share the magisterial splendor of the software's manual. It would be nice if there were more detailed explanations of how to set up the basic measurements needed for transducers beyond some spare block diagrams. It's a credit to the user-friendliness and intuitive quality of the software that I was nonetheless easily able to puzzle things out.

Reviewer Confession

Although I was supposed to be reviewing the APx1701 as part of a test and measurement system and keep my official scientist lab coat on at all times, my monkey brain got irresistibly curious. The APx515 contains a state-of-the-art DAC and pre-amplification, and the APx1701 is clearly a fantastic performer (see **Photo 3**). So I hooked it up to a pair of speakers, took some snippets of music files from my home recordings and used them as my "test" signals, with my ears as the measurement tools. Not unexpectedly, the sound was exceptional. If I were a marketing person at Audio Precision, I would be calling an engineer into my office and saying, "If we can add a pass-through function into the software that would enable users to easily play music through the APx515/1701 combination, we could have a high-end audio cult hit on our hands. After all, the price is lower than the cost of top-ranked commercial DACs or power amplifiers alone, and it's a completely integrated DAC and amplifier…"

Conclusions

The APx1701, along with the acoustic measurement software options in the AP500 system, converts a remarkably capable system for electrical measurement into an equally capable acoustic measurement system, with the added ability to acquire quantitative high-resolution impedance data. The integration is outstanding. Setup and measurement times are remarkably quick, and data that require amplification gain changes can

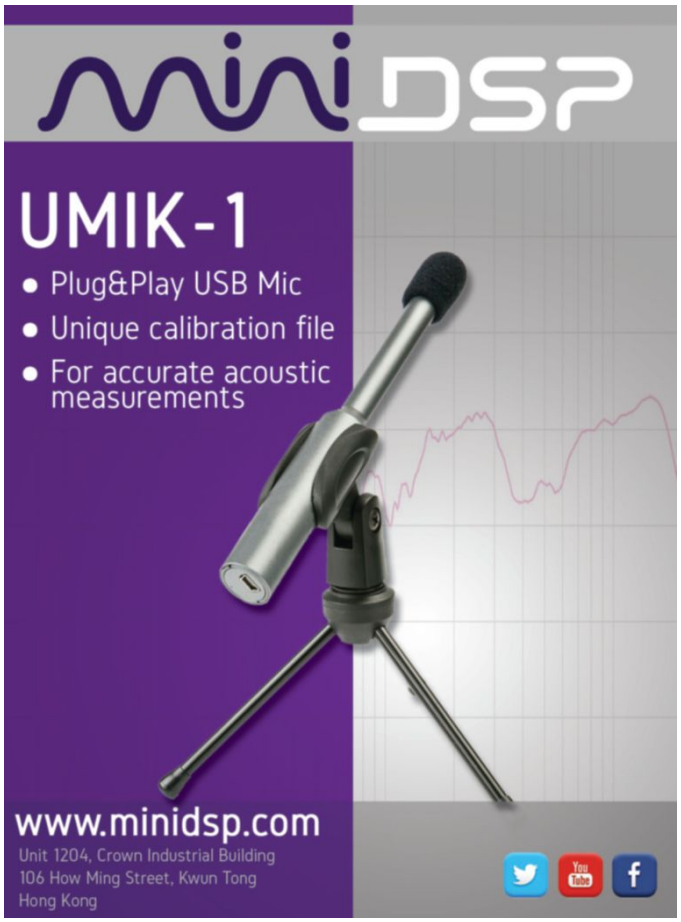


Photo 3: The APx1701 Transducer Test Interface is an APx accessory device that integrates instrument-grade amplifiers and microphone power supplies for designers and production test engineers seeking clear insight into the behavior of their electro-acoustic devices. And it sounds great too.

be obtained with simple menu selections. The need for additional boxes (e.g., microphone preamps and power amps) with all of their variability and calibration complexity is eliminated. The performance of the amplification is state-of-the-art and orders of magnitude better than any imaginable transducer. Given the performance, the price is remarkably modest.

My post-doctoral advisor, the late Nobelist Alan MacDiarmid, used to say, "Time is the most valuable chemical." This homily has stuck with me over the years, and perhaps explains why I have enjoyed my time with the APx515 and APx1701 combination. Sure, the performance is very fine, but the great thing was how the integration saved me so much time. Measured by the amount of high-quality data I was able to generate per hour, I'd estimate my productivity was an order of magnitude better. Moving from setup to setup or measurement type to measurement type went from an hour minimum to a minute or less. Changing measurement conditions or setting up the frequency sweeps took seconds.

Bottom line: I have been contemplating selling a kidney to pay for my own APx515/APx1701 combination, and am still ducking phone calls from Audio Precision asking for my loaner units back.





You Can DIY!

DIM-100—A Device for Interfacing and Measuring with Sound Cards (Part 1)

Begin with the Design



For the serious audio hobbyist, a quality PC sound card with an audio measurement and analysis software running on a PC or a laptop are the best tools to test your designs. With this design, you can build your own device used for interfacing and measuring with sound cards, which I called the DIM-100.

By
George Ntanavaras
(Greece)

During the last few years, I have spent a lot of time measuring audio devices, mostly loudspeakers. I used a laptop with an external sound card and several jigs that I built to interface the unit under test with the sound card, which proved useful especially for taking a loudspeaker's impedance measurements. I have used several PC software programs, however these days my favorites are ARTA and the Room EQ Wizard (REW).

When I decided to design my own device, the DIM-100, I wanted to include all the functionality and the convenience that I had wanted while taking those past measurements. I added a True RMS voltmeter to measure the exact level of the signals that enter and exit from the sound card. Although I extensively tested and used this device with ARTA and REW, I believe it could also be used with other software programs.

The Functionality

The device includes a lot of functionality and consists of three PCBs, as the block diagram shows (see **Figure 1**). Half of the first PCB (PCB1) includes the power supply for the digital and the analog circuits and its other half includes the circuits for the measurement of the DC resistance of the loudspeaker's voice coil, the circuit that measures the AC RMS current that passes through the loudspeaker, a reference resistor of 10 Ω to verify the accuracy of the impedance measurement and the necessary relays to select the device's functionality.

The second PCB (PCB2) controls the device's main functionality and includes: two switches, one for each channel, that select the amplification or the attenuation of the input signals; the buffer/amplifier circuit for the input signals; the volume control potentiometers and the line drivers for the output signals; a 12-position rotary switch, which selects the signal that will be measured by the voltmeter; a rotary switch that selects the device's mode of operation and a rotary switch that selects the type of impedance measurement that will be measured by the device.

The third PCB (PCB3) is a highly accurate True RMS AC voltmeter that measures all the signals that enter and exit from the device in millivolts, volts, or decibels.

The DIM-100's functionality can be easily seen and understood by examining its front panel

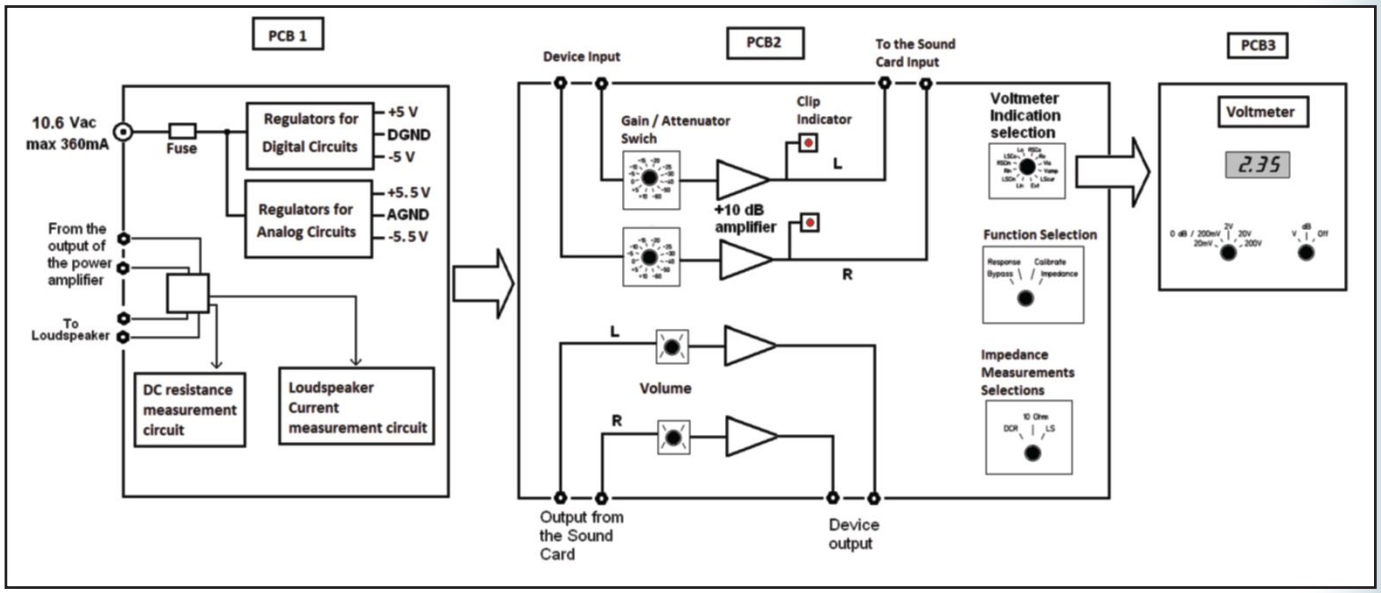


Figure 1: The device includes a lot of functionality and consists of three PCBs, which can be seen from the block diagram.

(see **Photo 1**). The front panel contains several connectors and switches. Two RCA connectors are used as inputs for the Left and the Right channel measurements. Below them, two switches can be used to select the amplification or the attenuation of the input signal from +10 dB to -60 dB, enabling the device to accept signals from a few millivolts RMS to several volts RMS.

There are also two RCA connectors used as outputs for the Left and the Right channels. Next to them, two potentiometers adjust the output level. Other controls on the front panel include: two 4 mm sockets, where the loudspeaker under measurement is connected; two 4 mm sockets, where the output of the external power amplifier is connected; a RCA connector for the external input of the voltmeter; a power connector, where an external AC/AC power supply is connected; and a 3.5 mm connector that provides a DC voltage of 0.1 V/Ω when the device is set to measure the DC resistance of a loudspeaker driver. There is also a 12-position switch that selects which signal will be measured by the voltmeter and a switch that sets the device in one of the four modes of operation—Bypass, Response, Calibrate, and Impedance measurement. (Later, I will explain the functionality of each mode.)

The panel also contains a switch that selects the device's operation when it is set to Impedance mode between the following: measurement of the DC resistance of the voice coil, measurement of a 10 Ω reference resistance that exist internally in the device to check the accuracy of the measurement, and the AC impedance of the loudspeaker. Finally, the panel also has an LCD of the True RMS voltmeter with a rotary switch that selects the sensitivity from 20 mV up to several volts and a second rotary switch that selects the indication of the measured voltage in volts or decibels.

The First PCB

For the DIM-100's supply, my first thought was to use a battery to eliminate any possible noise or



Photo 1: The DIM-100 can be used to interface and measure impedance in conjunction with sound cards.

ground loops, but the device's consumption was so high that it would not permit the use of a battery. So I used a Class B external plug-in AC/AC power supply with a 240 VAC/15 VA mains input protected with a 130°C thermal fuse and a 9 VAC/10 VA secondary output.

When all the relays are activated, the DIM-100's maximum consumption is 360 mA. And, it drops to 170 mA when they are deactivated. With 230 VAC input to the power supply, its output voltage, for the above two consumptions, varies from 10.4 to 10.8 VAC. With a ±10% variation of the 230 VAC, it gives a secondary output voltage from 9.4 up to 12 VAC, which is the maximum input voltage that should be applied because the filter capacitors C1–C8 have a working voltage of 16 VDC.

Figure 2 shows the first PCB's completed electronic diagram. The power supply's output is connected through the connector on the front panel to pins AC1 and AC2 of the circuit. L1 and L2 are ferrite beads inductors that have small impedance at low frequency, which increases to a high impedance at high frequencies. They offer electromagnetic induction (EMI) protection for the device from several megahertz up to gigahertz by suppressing any noise coming from the power supply. A slow-blow fuse, mounted on the PCB, protects the external AC/AC power supply from any short circuits that could happen within the device.

I wanted to eliminate any possible noise interference between the device's digital and analog circuits, so I used separate regulated supplies for each of them. For the analog power supply, the AC voltage is rectified by the diodes D1 and D2 and filtered by the capacitors C1 and C3. Transistors T1 and T2 with resistors R11, R12 and capacitors C2, C4 form a capacitor multiplier to suppress the ripple of the rectified voltage level before it enters to the regulators. Additional smoothing is offered by the capacitors C11 and C12 at the output of the

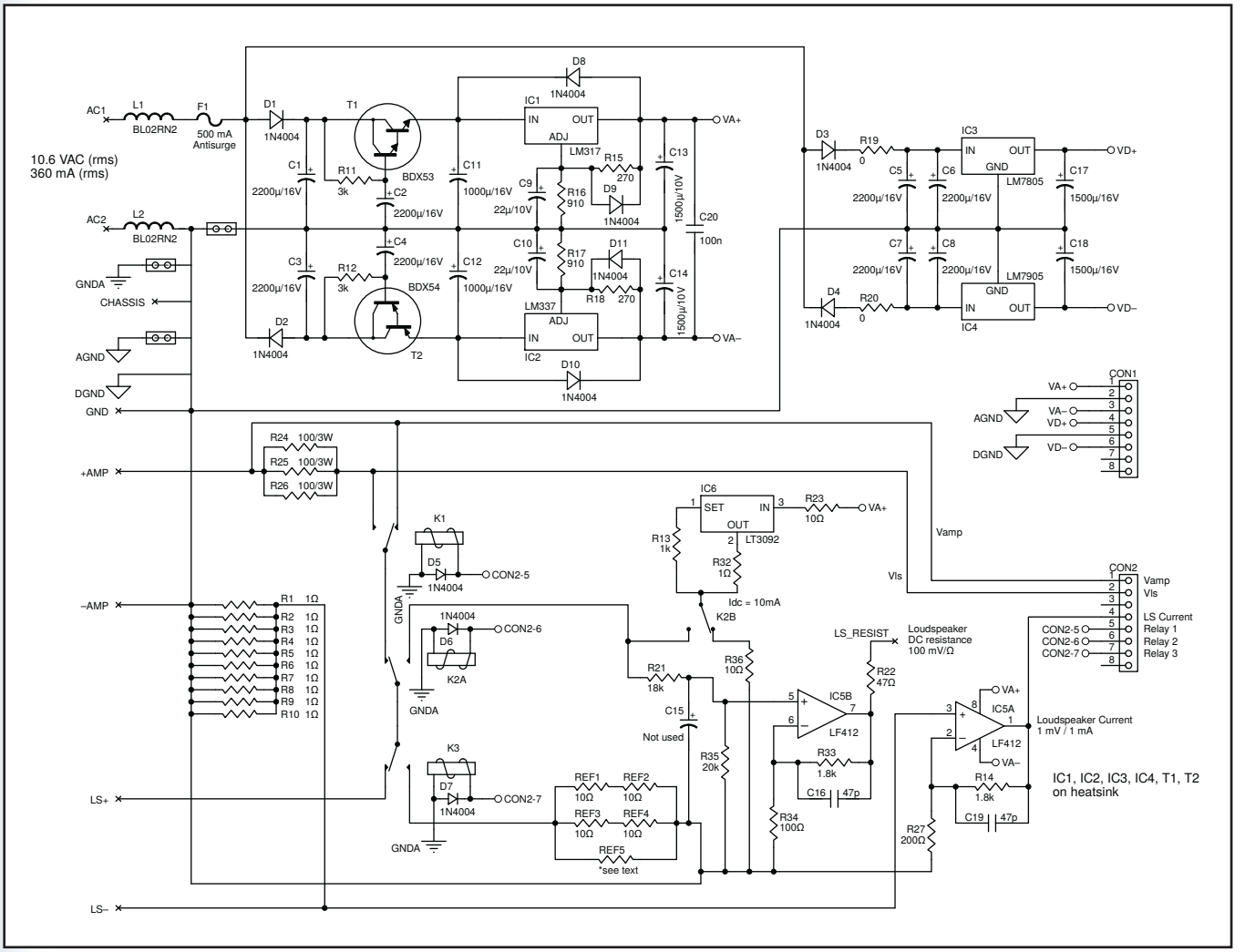


Figure 2: Here is the electronic diagram for the first PCB (PCB1).

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capacitor multiplier. The filtered voltage enters the LM317 and the LM337 regulators, which are adjusted for an output voltage of ±5.5 V. The capacitors C13 and C14 improve the stability and C20 decouples any remaining noise at the outputs of the two regulators.

For the digital power supply, diodes D3 and D4 rectify the AC voltage and capacitors C5, C6, C7, and C8 filter the voltage. Resistor R19 and R20 form low-pass filters with the capacitors to smooth the ripple of the input voltage, but this was unnecessary so I did not use them. The IC3 and IC4 regulators produce the output voltage of ±5 V and capacitors C17 and C18 improve the stability of the regulators.

Transistors T1 and T2 and regulators IC1, IC2, IC3, and IC4 require cooling. I used the bottom of the device's metal box as a heatsink (see **Photo 2**). All of them were electrically insulated from the chassis with mica thermal insulators.

Connector CON1 transfers the power to the device's other two PCBs. Resistors R24, R25, and R26 are connected in parallel to form a resistor with a nominal value of 33.3 Ω/9 W. This resistor is inserted in series to the loudspeaker under measurement. It is used by the analyzer software to measure the loudspeaker's impedance. If greater accuracy is required, the exact value of the above resistance should be measured and then inserted into the PC analyzer software.

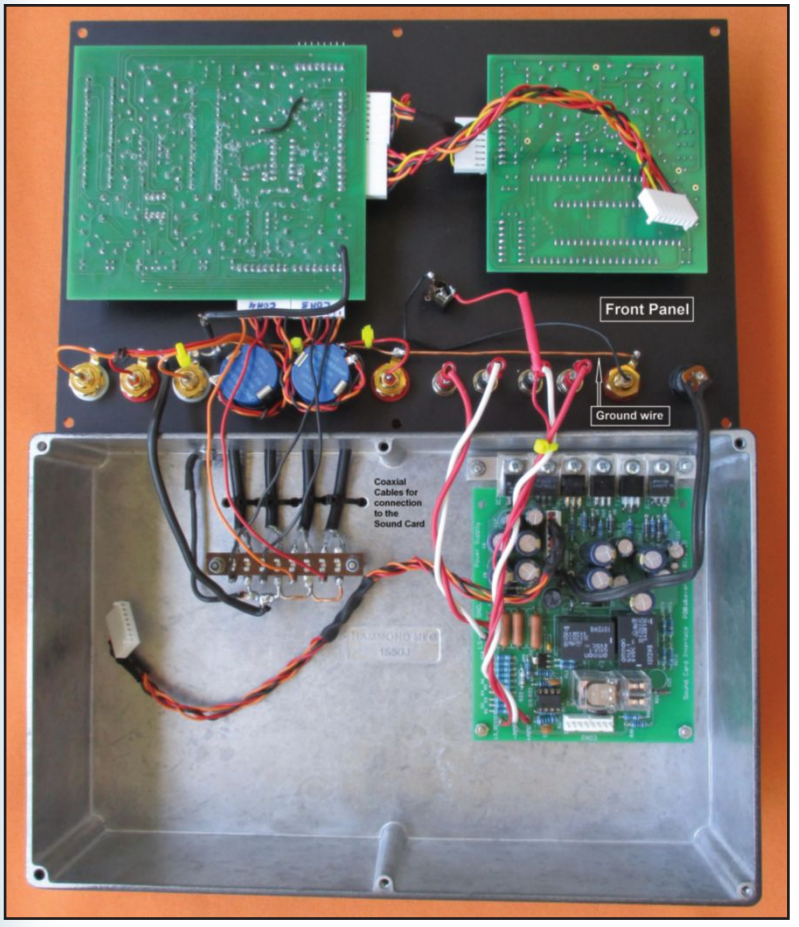


Photo 2: Here is an internal view of the DIM-100.

Relay K1 is used to bypass the above resistors when they are not needed for the measurement. Relay K2 is used to select an accurate current source of 10 mA DC that is used to measure the DC resistance of the loudspeaker's voice coil. The current is produced by IC6, which is a LT3092 programmable two-terminal current source that requires two resistors (R13 and R32) to set the output current. It is stable without any input or output capacitors, offering high DC and AC impedance, 1% initial accuracy, a low-temperature coefficient, and current regulation better than 10 ppm/V with a supply voltage from 1.5 to 40 V.

When this DC current flows through the loudspeaker's voice coil, a voltage is developed that is analogous to the voice coil's DC resistance. Resistor R21 and capacitor C15 form a low-pass filter to additionally smooth this voltage for a more stable output. However, this filter was not needed in the final circuit, so I did not place capacitor C15 on the PCB. Resistor R21 and R35 form a divider, which in combination with the gain of the op-amp IC5B, give a total amplification of 10 to the input voltage and increasing the output voltage 0.1 VDC per each ohm.

Resistors REF1, REF2, REF3, and REF4 are connected in a series/parallel combination to form a 10 Ω reference resistor. I used four resistors of the same value to increase the power dissipation and to compensate for the tolerance of the resistors. The test voltage on the 10 Ω resistors should not exceed 2.5 V, due to the limited power dissipation of these resistors. Resistor REF5 is optional and can be used to trim the value of the resistors exactly at 10 Ω, but only if it is greater.

Relay K3 is used to disconnect the loudspeaker and connect the reference 10 Ω resistance. Resistors R1 to R10 are connected in parallel to form a resistance of 0.1 Ω, which is used to sense the current that passes through the loudspeaker. Here, I also used 10 resistors of the same value to increase the power dissipation and to compensate for the tolerance of the resistors.

The circuit around IC5A measures the voltage developed on these resistors, amplifies it by 10, and gives a voltage of 1 mVRMS per each milliamp RMS at its output. This voltage can be selected and measured by the device's voltmeter. Diodes D5, D6, and D7 are used to suppress any reverse voltage developed by the relay coils during switch off.

The Second PCB (PCB2)

PCB2 is the device's main unit, containing the circuits that process the signals from and to the sound card, and the switches that control the device's functionality.

Figure 3 shows PCB2's electronic diagram. The Left and the Right channels are identical, so I will only describe the circuits of the Left channel. Capacitor C3 blocks any external DC voltage, positive or negative, from entering the device. Because of this, I used a 50 VDC bipolar capacitor since I didn't expect greater voltage in my measurements. If this is not the case, the capacitor's voltage should be increased accordingly.

After the capacitor, the input signal goes to a 12-position, high-accuracy divider formed by resistors R1 to R23. In combination with the next amplifier circuit around IC1A (with a gain of +10 dB), the input signal can be amplified or attenuated from +10 dB to -60 dB. Resistor R51 with diodes D3 and D4 protect the electronic circuits from overload and with capacitor C10 form a low-pass filter. R67 buffers the output of the op-amp, which drives the sound card's input. Using the values shown in **Figure 3**, the circuit's -3 dB frequency bandwidth ranges from 0.15 Hz to 300 kHz.

The circuits around IC3A and IC3B form a comparator, which monitors the voltage that enters the sound card and indicates when a clipping occurs. This is accomplished by adjusting the reference voltage that goes to the comparator with trimmer R59. Diodes D9 and D10 form an OR gate that drives the transistor T1 when a voltage above or below the threshold level appears at the sound card's input. The OVRLD_LEFT (Overload LED) is turned On to indicate that a clipping occurred. C1 and R78 form a delay circuit to enable the LED to remain On for additional time after the clipping occurs.

The output signal from the sound card goes directly to the volume level potentiometer through connector CON4. The output of the potentiometer is connected to the input of the op-amp IC2A, which buffers and amplifies it by 15 dB (set by the resistors

About the Author

George Ntanavaras graduated from the National Technical University, Athens, Greece, in 1986 with a degree in Electronic Engineering. He currently works in the Development Department for a Greek electronics company. He is interested in the design of preamplifiers, active crossovers, power amplifiers, and most loudspeakers. He also enjoys listening to classical music.



R70 and R80). Resistor R48 buffers the op-amp's output to drive long cables without problems.

Switch S3 selects one of the four main modes of operation of the device: Bypass, Response, Calibrate, or Impedance.

Bypass is the device's normal mode of operation. Whatever exists on the device's inputs enters the sound card's inputs after going through the attenuator circuit with its buffer. Also, whatever exists in the sound card's outputs goes through the volume potentiometers and the output buffer to the device's output.

The Response mode performs the same operation, but additionally internally connects the power amplifier's output that is connected to pin (+Amp) of the PCB1 to the Left Input of the sound card. In this mode—and providing that in the ARTA software, the Right channel has been selected as the preferred input of the recorder—the Left channel becomes the reference channel for the response measurement. This internal connection saves one external cable but only when a power amplifier is connected to the loudspeaker through the device.

The Calibrate mode is selected when the ARTA software has to calibrate the levels of the two input channels for the impedance measurement. In this case, the input attenuators for the Left and the Right channels should be placed in the same position.

The Impedance mode is selected to measure the loudspeaker's impedance. In this case, the next switch (switch S4) can select one of the three modes of operation: DCR, 10 Ohm, or LS.

In the "DCR" mode the DC resistance of the loudspeaker's voice coil is measured by using an accurate current source of 10 mA DC as was described earlier in the article. In the "10 Ohm" mode, the relay K3 on the PCB1 is activated and an internal reference resistance of 10 Ω is connected for measurement instead of the loudspeaker. In this case, the audio analysis software should measure an exact value of 10 Ω for the entire frequency range. I use this mode as a quick way to confirm that the entire measurement setup is correct—usually before I start any measurements, but sometimes between the measurements to confirm that everything remains fine. In the "LS" mode, the loudspeaker is normally connected and its impedance will be measured from the software.

S5 is a 12-position rotary switch that selects the signal that will be measured by the device's internal

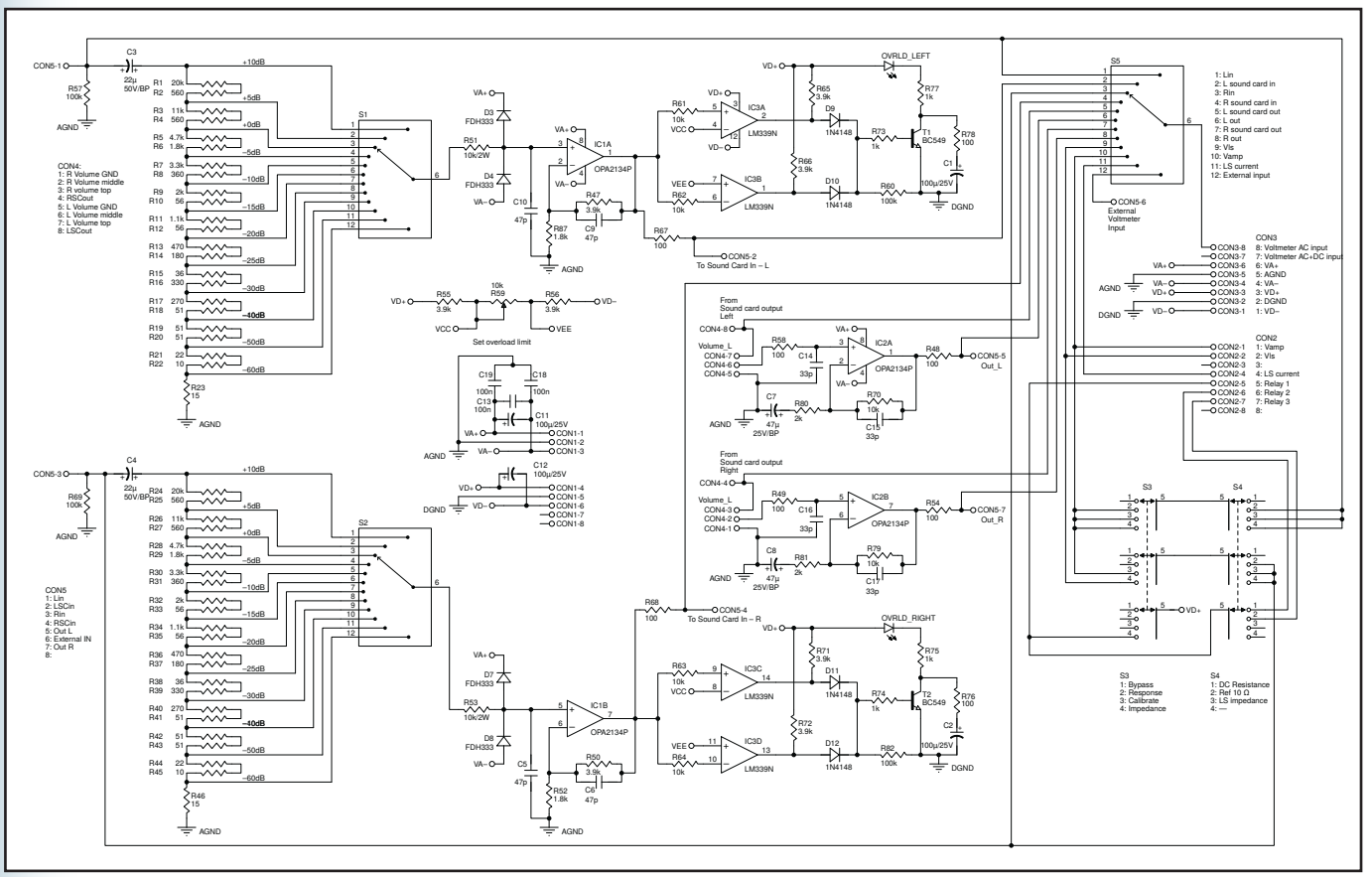


Figure 3: This is the electronic diagram for the second PCB (PCB2).

voltmeter. The 12 positions on the switch include:

- Lin: The signal that is present at the Left input of the device is measured.
- LSCin: The signal that is present at the Left input of the sound card is measured.
- Rin: The signal that is present at the Right input of the device is measured.
- RSCin: The signal that is present at the Right input of the sound card is measured.
- LSCo: The signal that is present at the Left output of the sound card is measured.
- Lo: The signal that is present at the Left output of the device is measured.
- RSCo: The signal that is present at the Right output of the sound card is measured.
- Ro: The signal that is present at the Right output of the device is measured.
- Vls: The signal that is present at the (+ LS) and (- LS) loudspeaker sockets in the front panel of the device is measured.
- Vamp: The signal that is present at the (+ Amp) and (- Amp) power amplifier sockets in the front panel of the device is measured.
- LScur: The AC current that passes through the loudspeaker is measured.
- Ext: The signal that is present at the external input of the device is measured.

With the above functionality, every signal that enters or exits the device can be measured, which is a convenient feature of the DIM-100.

The True RMS Voltmeter (PCB3)

The third PCB (PCB3) is a True RMS voltmeter based on one of my earlier designs that was described in an *audioXpress* article called "A True RMS Audio AC Voltmeter," (July 2009). The voltmeter design can also be used as a stand-alone unit for any other application.

The voltmeter is based on an Analog Device AD636JH chip and on the AD636 circuit (for Portable, High-Z Input, RMS DPM and Decibel Meter Circuit) shown in Figure 17 of the Analog Device's AD636 datasheet (see Resources). The AD636 is a low power monolithic IC that performs true RMS to DC conversion on low-level signals from 0 to 200 mV RMS. It includes an auxiliary decibel output derived

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Potcore Inductors, Toroidal Inductors
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Marchand Electronics Inc.
(585) 423 0462 www.marchandelec.com

from an internal circuit point that represents the logarithm of the RMS output. The 0 dB reference level is set by an externally supplied current and was selected to correspond to an input level of 774.6 mV RMS.

According to the manufacturer, the frequency response ranges from 1.2 MHz at 0 dBm to greater than 10 kHz at -50 dBm. The AD636 computes the true root-mean-square (RMS) of a complex AC (or AC plus DC) input signal and gives an equivalent DC output level. The true RMS value of a waveform is a more useful quantity than the average rectified value because it is a measure of the power in the signal.

An averaging capacitor is the only external component required to perform the measurements to the fully specified accuracy. Its value optimizes the trade-off between low-frequency accuracy, ripple, and settling time. The AD636's 200 mV full-scale range is compatible with many popular display-oriented ADCs. **Figure 4** shows the voltmeter's complete electronic diagram.

Connector CON3 transfers the power supply voltages (analog and digital) and the input signals from the PCB2 to the PCB3 (the voltmeter). Diode D5 is short circuited with a jumper, and it is used only if the voltmeter is powered by a battery to prevent accidental inverse polarity. Capacitor C11 decouples any possible DC component of the input signal presented at pin 8 of CON3. The capacitor can be bypassed if as input is used the pin 7 of CON3. In this case, the voltmeter measures the AC and the DC component of the input signal. The signal is then driven to the 1 MΩ high impedance input attenuator with the five positions: 20 mVRMS, 200 mVRMS, 2 VRMS, 20 VRMS, and 200 VRMS. For safety reasons, due to the construction of the PCB, the maximum input voltage for the multimeter and the device should always be kept below 60 VRMS.

The output of the attenuator switch S1, goes through the protection circuit R23, D1, and D2 to the amplifier around IC5. Here, I used an OP27 op-amp, which is a high-precision, low-noise operational amplifier for amplification of very low signals. The circuit has a gain of 10, so that the voltmeter can measure voltages lower than 20 mV. This amplifier's output drives the AD636 buffer amplifier, which is connected in an AC bootstrapped configuration to minimize the loading. The buffer amplifier's output drives the 6.7 kΩ input impedance of the AD636. The AD636's linear and decibel outputs are driven through the S2 switch to the ICL7106's input.

I used an LT1004-1.2 V reference diode to provide a stable voltage reference for the ADC in the linear RMS mode by using the trimmer R_LIN (500 Ω). In the decibel mode, the trimmer R_DB (100 kΩ) is used to set the 0 dB indication. I inserted the diode D4 in series to provide correction for the temperature coefficient of the decibel scale factor. The trimmer

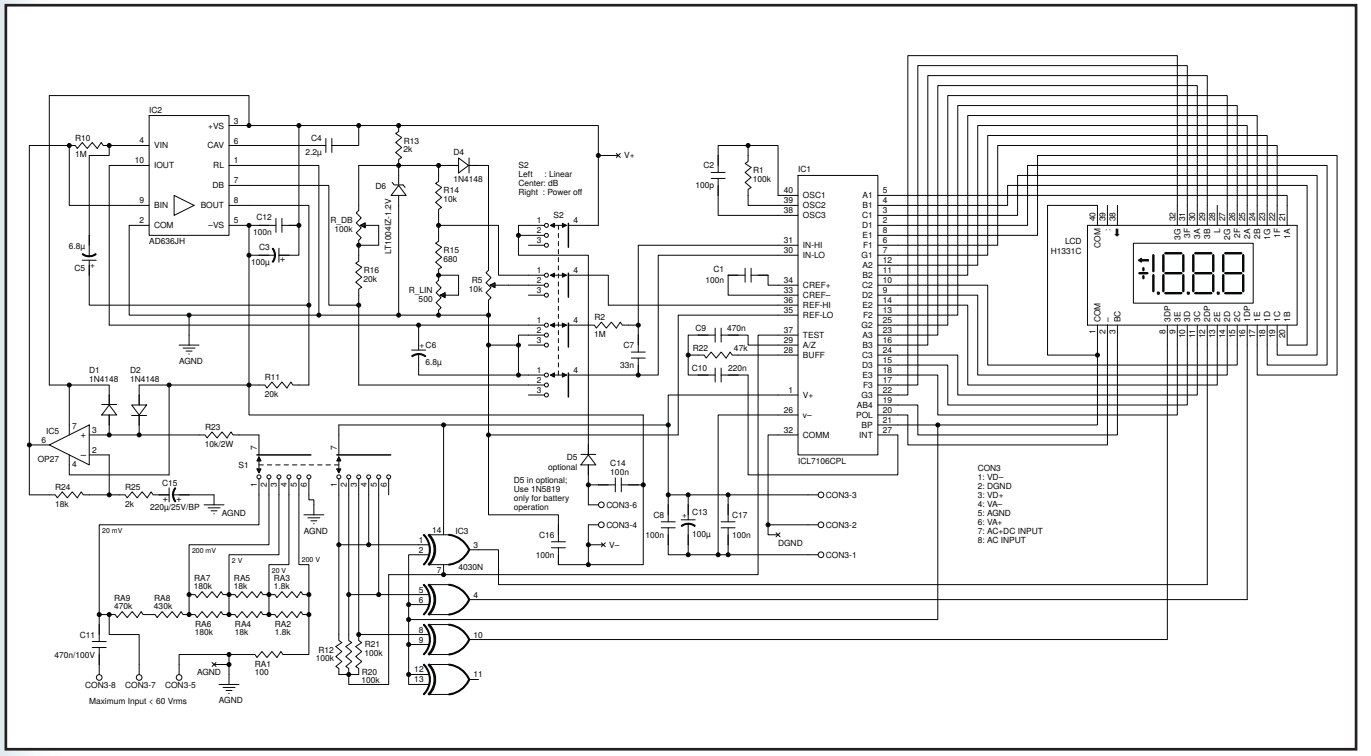


Figure 4: The third PCB (PCB3) contains the voltmeter.

R5 (10 kΩ) is used to adjust the scale factor of the decibel scale.

The IC1 is an Intersil ICL7106 low power, 3.5 digit ADC with seven segment decoders, display drivers, a reference voltage, and a clock. The ICL7106 is designed to interface with a LCD, and for this reason, includes a multiplexed back-plane drive. The ICL7106 is placed below the LCD to facilitate the large number of the connections between them.

It also features auto zero to less than 10 µV, zero drift of less than 1 µV/ºC, an input bias current of 10 pA (maximum), and a rollover error of less than one count.

The circuit around IC3 in combination with the half part of S1 adjust the decimal point of the display to the proper position according to the input scale. This enables users to easily read the displayed value, eliminating the need to multiply or divide the display value according to the input scale. The display that I used is a reflective 3.5 digit LCD with a 0.5" character height that can easily be viewed even in the sunlight.

In the second part of this article, I will discuss the assembly of the DIM-100 and the test measurement results I obtained once it was completed.

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Eagle software program
Eagle PCB Design | https://cadsoft.io

LTspice program
Linear Technology | www.linear.com/designtools/software/#LTspice

REW software
Room EQ Wizard | www.roomeqwizard.com

Front Panel Designer program
Schaeffer AG | www.schaeffer-ag.de/en/downloads/front_panel_designer/?no_cache=1

8¾" Balanced Drive Die Cast
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Build an Audio Dummy Load

Learn how to build an essential piece of test equipment for your audio electronics workshop, using easy-to-find parts.

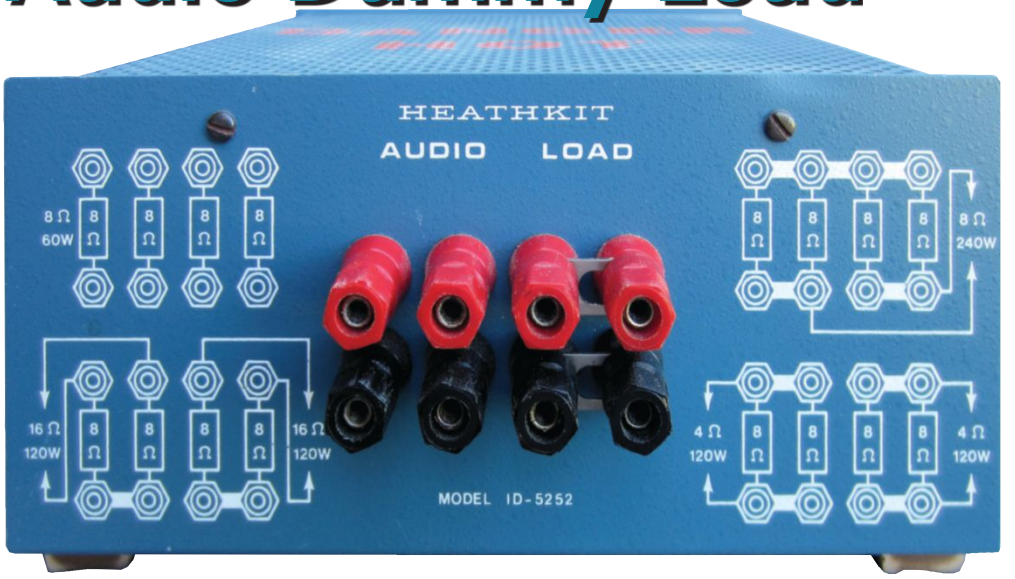


Photo 1: The Heathkit ID-5252 Audio Load was built by the author around 1980. Four 8 Ω, 60 W resistors can be connected in various series and parallel combinations.

By
Gary Galo
(United States)

An audio load is an essential piece of test equipment for any audio electronics workshop. One of the most popular audio loads in DIY circles was the Heathkit ID-5252, which I built more than 35 years ago (see **Photo 1**). The ID-5252 contains four 8 Ω, 60 W power resistors. The biggest limitation of the Heathkit load is power handling. Series and parallel combinations can be used to produce various resistances and power levels—up to 240 W at 8 Ω, using all four resistors. But, simultaneously testing both channels of a high-powered amplifier requires two ID-5252 units.

It's been a quarter century since the Heath Co. got out of the kit business. And, although most types of test equipment once offered by Heathkit can be purchased factory-assembled from countless manufacturers, no one seems to be making an audio load. If you need one, you'll have to roll your own.

(As an aside, Heathkit re-emerged in 2013, and currently offers an AM radio kit, a 2 m amateur radio antenna, and upgrades to a few of its old products, including the IG-18 Audio Generator and the AJ-1510 FM Tuner. I don't know how many electronics enthusiasts will pay $150 for an AM radio with no dial calibration and no speaker, but I wish the company well.)

Audio Load Parts

Fortunately, the raw parts for building an audio load are easy to find. Parts Express carries 4 Ω and 8 Ω load resistors designed specifically for audio testing. They are available in 100 W and 200 W power ratings, and are reasonably priced. The resistors are manufactured by Loyal Machinery and Electronic Co. Ltd., (LMEM) in Taiwan, and are part of the company's MH-series. The specific type carried by Parts Express is the MHNL. The "N" indicates non-inductive and the "L" indicates that the resistor has flying leads rather than terminals.

Figure 1 shows the schematic diagram for the dummy load. To me, it made sense to go with the 200 W resistors—two at 4 Ω and two at 8 Ω, for a total of four resistors. Parts Express specifies 5% tolerance for these resistors. Using my B&K Precision 879B LCR meter, I measured the actual resistance for the four samples at 4.072 Ω, 4.078 Ω, 7.956 Ω, and 8.034 Ω, for tolerances of 1.8%, 1.95%, 0.43%, and 0.55%. B&K Precision specifies an accuracy of 2% for single-digit resistances, so the LMEM resistors are well within

tolerance. Parts Express claims that the resistors are non-inductive, and measurements with the B&K Precision meter show the inductance to be very low on the 200 W versions: 20 µH for the 8 Ω resistors and 14.5 µH for the 4 Ω resistors. The inductance changes no more than 0.5 µH as the test frequency is varied between 100 Hz and 10 kHz. The very low inductances are reflected in the 10 kHz impedance measurements, which are within 0.1 Ω of the resistance measurements.

Heatsinking

Each resistor is encapsulated in its own heavy, aluminum heatsink, and various notes on the Parts Express website indicate that they can be used without any additional heatsinking. However, this isn't true if the resistors are to be used at or close to full power for extended periods, so I added an additional heatsink to each of the four resistors in my dummy load. Parts Express carries an anodized aluminum heatsink that's convenient for this purpose—Sure Electronics' DC-HS11123. The Sure heatsink comes with a lot of extra hardware that's not needed for this project, including eight mounting brackets, two bags of machine screws, and an acrylic LED diffuser panel. All you need for one resistor is the heatsink and four of the metric number 3 mm × 8 mm machine screws, plus four 6/32 × 1/2" machine screws available at your local hardware store.

The Sure heatsink has a proliferation of holes tapped for the metric number 3 machine screws. The only ones needed for this project are four along one edge of the heatsink, which will be used to mount the heatsink/resistor assemblies to the chassis bottom plate. The heatsink also has four untapped holes on the large, flat side, which are exactly the right size for a 6/32 tap. I used these to secure the resistors to the heatsinks, using 6/32 × 1/2" panhead machine screws.

In addition to the right size hole, the key to success in tapping aluminum is to use a high-quality tap. This is no place to look for a bargain. Cheap taps are prone to breaking, and once a tap breaks, the end stuck in the metal you're tapping will be very difficult to remove. At that point, you're likely to utter colorful language inappropriate for polite company (I speak from experience!). Irwin/Hanson taps are high-quality, American-made tools that are available at all the major hardware stores. If you live outside the US, you can use a metric 3.5 mm × 0.6 mm tap and machine screws instead of 6/32.

One edge of the resistor will be flush with one edge of the heatsink—this is the side that will be mounted to the bottom plate of the enclosure. The opposite resistor edge with extend about 1/8" beyond the other edge of the heatsink. You'll need to file four notches in the beveled edges on the wider side of the resistors to provide clearance for the 6/32 machine screws (see **Photo 2**). Use a round file and file just enough for the screws to clear. When the machine screws are tightened, one edge of each screw head will secure the resistor to the heatsink. You can substitute a flange-head or button-head machine screw—the built-in washer on these types will give you a little more bearing on the resistor case. However, I had no problem

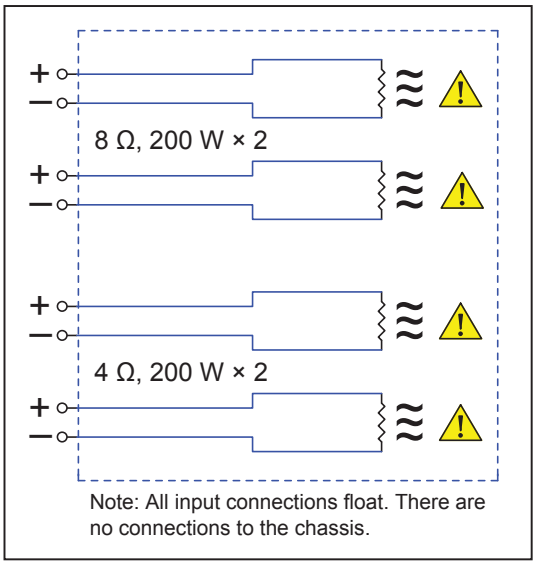


Figure 1: Here is the schematic diagram of the audio dummy load. The author chose a pair of 8 Ω and a pair of 4 Ω resistors at 200 W each, but these can be changed to suit individual needs. All input connections float, and there are no connections to the chassis. High-power dissipation will generate potentially dangerous levels of heat, so caution is required on the part of the builder and the user.

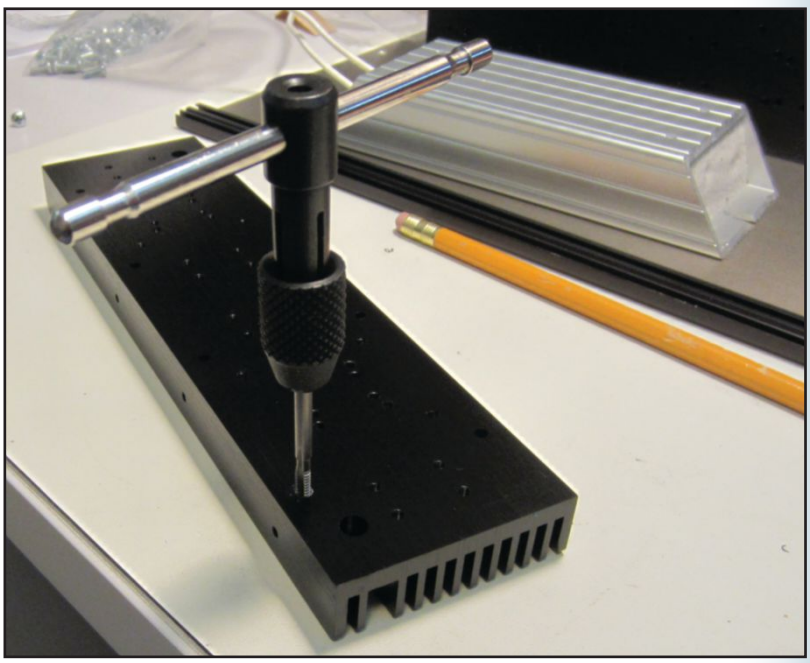


Photo 2: Four holes in each DC-HS11123 heatsink must be tapped for the 6/32 machine screws. Fortunately, the holes are exactly the right size for a 6/32 tap.

securing the resistor to the heatsink with panhead screws. Do a "dry run" to make sure everything fits properly. An alternate method of mounting the heatsinks would be to use the two large notches on the ends of the resistor case. But, this will require drilling holes in the heatsink.

Next, apply a thin layer of thermal compound to each resistor (see **Photo 3**), and mount the resistor on the heatsink. I used white, silicone-based thermal compound impregnated with zinc oxide. Rawn 10004 Silicone Heat Sink Thermal compound is rated for temperatures as high as 260°C (500°F), and is available from MCM Electronics. Similar products are available from a variety of electronics suppliers. The trick during application is to use enough so the compound will ooze into the grooves of the resistor body when the resistor is tightened against the heatsink. This will maximize thermal transfer. If you're in doubt, fill each groove first. If you apply to much compound, you'll need to wipe off the excess after tightening the screws. **Photo 4** shows a completed resistor/heatsink assembly.

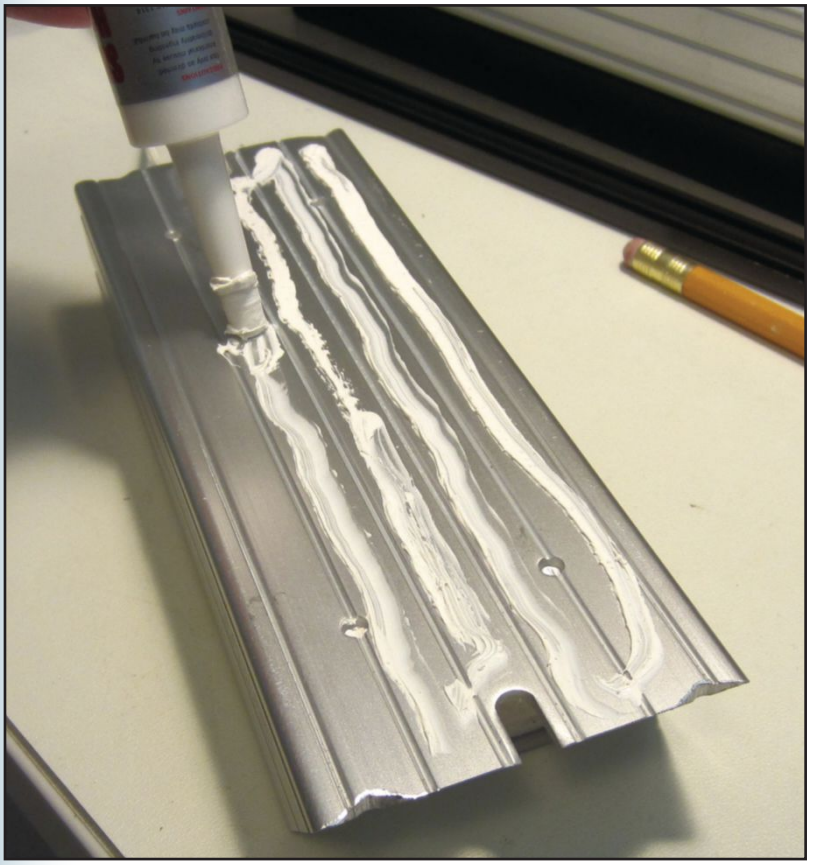


Photo 3: A thin bead of thermal compound should be applied to each resistor to ensure good thermal transfer to the heatsink. The notches filed into the ends of the resistor case are necessary to allow clearance for the 6/32 mounting screws.

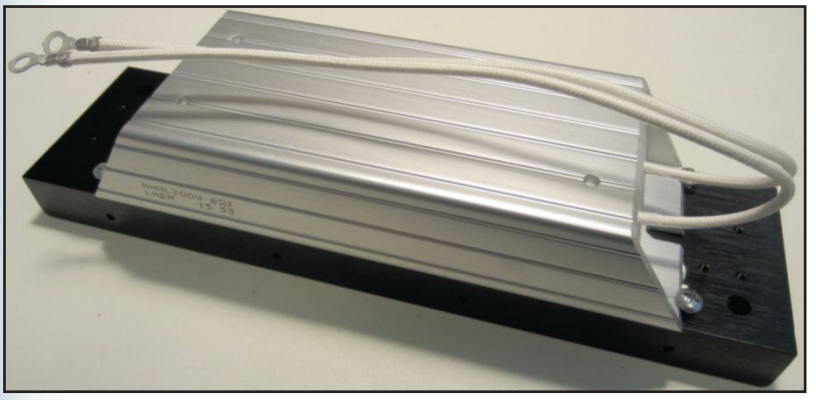


Photo 4: One completed resistor/heatsink assembly. The resistor and heatsink are flush on the bottom; the resistor overhangs the top edge of the heatsink by about 1/8".

Modular Enclosure

My favorite enclosures for audio projects are the modular aluminum types made by Wolgram Engineered Plastics, also sold by Markertek under the Connectronics brand. These are the same enclosures that were made by Sescom back when it was run by the late Franklin Miller. I used the 2RU10 Rack Chassis for this project. If you don't want or need the rack ears, try the MC-20A instead—it has the same internal dimensions.

The resistor/heatsink assemblies should be secured to the chassis bottom plate with four of the metric number 3 mm × 8 mm machine screws supplied with each heatsink. I used five-way binding posts mounted on the front panel for the connections to the resistors.

Photo 5 shows the final assembly of the dummy load, with the four resistor/heatsink assemblies mounted on the bottom plate, and the resistor leads secured to the binding posts. The resistor leads are not polarized, but in the interest of neatness, connect the bottom resistor leads to the bottom (black) binding post terminals. Note that the resistor leads are connected only to the binding posts. All connections float, and there are no connections to the chassis.

The solid top plate of the Wolgram/Connectronics enclosure won't provide any ventilation for the resistors, so I replaced it with a piece of perforated aluminum. The Small Parts 3003 perforated aluminum sheet, available from Amazon, is the same thickness as the 2RU10 and MC-20A top plates—14 AWG. You'll have to cut it to fit, and secure the center brackets on the sides of the case using small fender washers along with the supplied machine screws.

Each resistor flying lead has two solid conductors covered with insulation designed for







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high-temperature applications. Nonetheless, I think it's prudent to avoid having the resistor leads touch the resistors or the heatsinks. Fortunately, the leads are stiff enough that you can easily dress the leads between the resistor/heat sink assemblies when you secure the front panel in position.



Photo 5: Final assembly of the audio load includes two 4 Ω, 200 W resistors on the left and the 8 Ω resistors on the right, mounted in a Wolgram/Connectronics 2RU10 enclosure. Five-way binding posts are mounted on the front panel to facilitate connections to the load resistors.

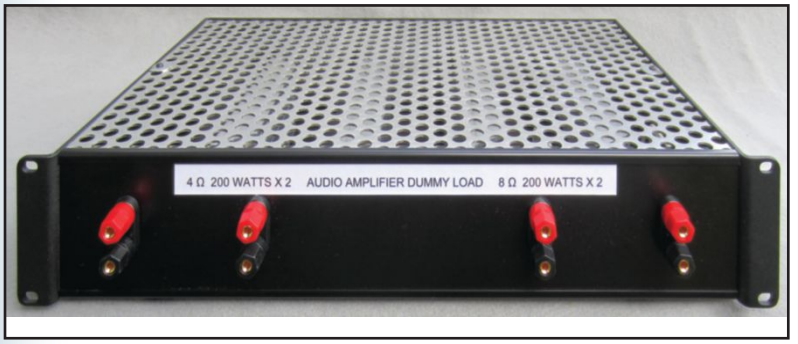


Photo 6: For the completed audio load, I replaced the solid aluminum top that was supplied with the 2RU10 enclosure with a perforated aluminum one for ventilation.

Photo 6 shows the completed audio dummy load. Having one pair each of 4 Ω and 8 Ω resistors is ideal for my own needs. However, you may use any combination that suits your own needs. Following Heathkit's approach, you can create a variety of load values using series and parallel combinations.

Testing

Most readers probably don't need a lesson in Ohm's Law, but it's a bit sobering to consider the demands made on an amplifier and a load resistor at a power level of 200 W. To dissipate 200 W into 8 Ω, an amplifier must deliver 40 VRMS to the load at a sustained current level of 5 A. This is no mean feat. In addition, dissipating that much power into a load resistor will generate considerable—indeed, potentially dangerous—levels of heat. These resistors appear designed for heavy duty use, but user caution is mandatory.

The LMEM datasheet for the MH200 lists a chassis-mounted surface temperature specification of 240ºC (464ºF) at full power, and a chassis-mounted load rating of 1.5 hours ON followed by 0.5 hours OFF, repeated for 1,000 hours. The specified chassis is 300 mm × 300 mm, 3 mm thick, which is a sizable chunk of metal. You'll never achieve this level of chassis heatsinking if you mount four resistors in a reasonable-size enclosure, which is why I believe that additional heatsinking is important.

I tested one of the 8 Ω resistors at its full power rating for 5 minutes, using a QSC RMX 1450 professional audio power amplifier and my Fluke

About the Author

Gary Galo retired in 2014 after 38 years as Audio Engineer at The Crane School of Music, SUNY at Potsdam, NY. He now works as a volunteer in the Crane Recording Archive doing preservation, restoration, and digital transfer of vintage Crane recordings. He is also a Crane alumnus, having received a BM in Music Education in 1973 and an MA in Music History and Literature in 1974. Gary is a widely-published author with more than 300 articles and reviews on both musical and technical subjects, in over a dozen publications. Gary has been writing for *audioXpress* and its predecessors since the early 1980s. He has been an active member of the Association for Recorded Sound Collections (ARSC) since 1989, and a frequent recording and book reviewer for the *ARSC Journal*. He has given numerous presentations at ARSC annual conferences, many of which have been published in the *ARSC Journal.* He was the Sound Recording Review Editor of the *ARSC Journal* from 1995-2012, and co-chair of the ARSC Technical committee from 1996-2014. Gary has also published numerous book reviews in *Notes: Quarterly Journal of the Music Library Association*, written for the *Newsletter of the Wilhelm Furtwängler Society of America, Toccata: Journal of the Leopold Stokowski Society*, and he is the author of the "Loudspeaker" entry in *The Encyclopedia of Recorded Sound* in the US. He has also written several articles for *Linear Audio*. He is a member of the Audio Engineering Society, the Boston Audio Society, and the Société Wilhelm Furtwängler.

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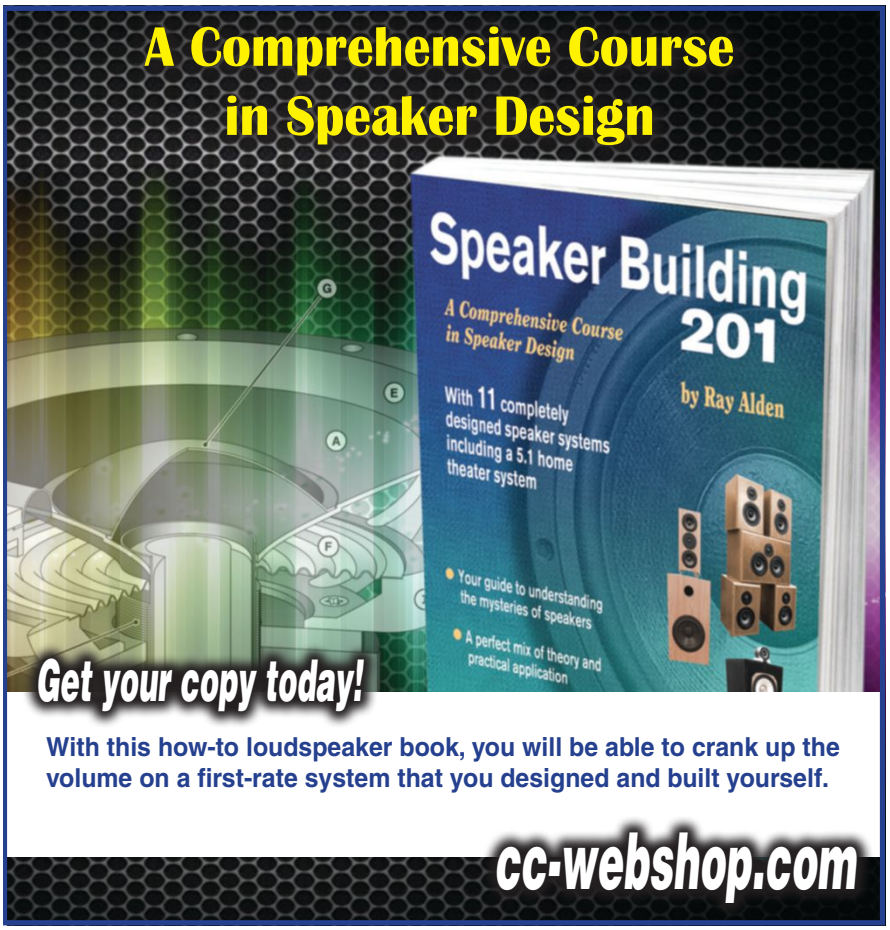
Parts List

Quantity	Part	Source
2	4 Ω, 200 W, non-inductive dummy load resistors	Parts Express 019-025
2	8 Ω, 200 W, non-inductive dummy load resistors	Parts Express 019-030
4	Sure Electronics DC-HS11123 Aluminum Heatsink Kits	Parts Express 320-3282
4	Dual binding posts	Parts Express 090-475
1	2RU10 rack chassis enclosure or MC-20A metal cabinet	Wolgram Engineered Plastics, or from Markertek as the Connectronics CTX-2RU10 or CTX-MC-20A)
1	Small Parts 3003 aluminum perforated sheet, 12" × 24", 0.063" thickness, 14 gauge (Amazon)	
16	6/32 × 1/2" panhead, flanged-head or button-head machine screws	
1 tube	Rawn 10004 silicone heatsink thermal compound (MCM Electronics 20-300) or equivalent	

62 Mini Infrared Thermometer. The RMX 1450 is a pro-audio workhorse rated at 280 WRMS per channel into 8 Ω, so it had no problem delivering 200 W into the load resistor. After 5 minutes at 200 W, the resistor measured 93°C (200°F) on both the side and the top of the metal resistor case. The corner (edge) of the resistor case was a bit hotter, at 102°C (215°F). The heatsink measured 91°C (195°F) along the edge and in the middle of the fins, which indicates good heat transfer from the resistor to the heatsink. My temperature readings are well below the manufacturer's data, but I didn't run the test any longer. If I had, the temperature would have continued to rise well beyond my own comfort level. Immediately after disconnecting the resistor from the amplifiers, the resistance measured 8.02 Ω (LMEM specifies the temperature coefficient as ±260 ppm/°C MAX).

The 93°C measured after 5 minutes is a considerable amount of heat—nearly hot enough to boil water—and the manufacturer's specification of 240°C is well into solder-melting territory. Even if a resistor is designed to withstand such temperatures, other things in its vicinity, including the human performing the test, probably aren't. Exercise extreme caution when making high-power measurements—the risks of fire and personal injury should NOT be taken lightly. As is the case with your kitchen stove and outdoor grill, safety is ultimately in the hands of the operator.

I don't foresee the need for continuous operation at the full power rating of these resistors. If you do, you should strongly consider fan cooling and possibly larger heatsinks. The heatsink fin orientation should be in the direction of the air flow from the fans. The horizontal fin orientation in my layout is ideal for mounting up to four small "muffin"-type fans on the back of the enclosure. A perforated bottom plate with extra-long feet will further improve air circulation through the enclosure and increase the distance between the bottom plate and the adjacent surface. The perforated bottom plate may require some reinforcement to support the weight of the four resistor/heatsink assemblies.



You Can DIY!

TM50x Power Module Tester

I designed the CAL-69A module tester after encountering an obstacle at work with our old TM501 power module. The tester is designed to detect weak filter capacitors and degraded or failed power transistors.



Photo 1: The front panel controls for my CAL-69A power module tester include a power supply selection switch, two reference test points, DMM connection jacks, a test point common, and a LED power indicator.

By
Chuck Hansen
(United States)

Tektronix made more than 100 different functional plug-ins that it used in its TM50x series of power supply modules. These consisted of the TM501 single-bay power module up to the TM506 six-bay power module. Tektronix also produced several audio test equipment plug-in modules for the TM50x series, including audio oscillators, function generators, and a three-bay distortion test set for the TM503 power module.

My project came about because we had a TM501 with old, weak DC filter capacitors that was being used with a Tektronix AM503A current probe amplifier. The excessive ripple on the DC supply affected the accuracy of the current measurements while important development testing was in progress. To determine the problem, I designed and developed what we designated the CAL-69A power module tester (see **Photo 1**). The tester provides test loads on each of the six internal DC sources of Tektronix 50x power modules, to detect weak filter capacitors and degraded or failed power transistors. **Photo 2a** shows a Tektronix TM501A with three Sprague metal can capacitors removed from the module. **Photo 2b** shows the same module with the three new replacement capacitors installed on the TM501 PCB.

This is not a piece of commercial test equipment. It is a one-off unit that we use in the lab where I work to test and verify Tektronix TM50x power supply module operation. I guess you could call it a "brass board" instead of a breadboard. I appropriated the Tektronix chassis from another plug-in module that was no longer repairable. However, the design has no association with Tektronix.

Description

We use the CAL-69A power module tester to check all six power sources in each compartment bay as well as the two unassigned bipolar power transistors—one NPN and one PNP—for the for TM50x series power modules (e.g., TM501, TM502, TM503, etc.). It also enables us to measure the DC ripple across the three aluminum filter capacitors on the three TM50x internal DC supplies (+33.5 VDC, +11.5 VDC, and -33.5 VDC). The switch positions are marked with a ~ sign to indicate they are used for DC voltage and AC ripple check.

In addition, we use Zener diodes to bias the TM50x internal power transistors as emitter followers to provide +22 VDC and -22 VDC test voltages from the two TM50x isolated 25 VAC

windings. A pair of diodes and a filter capacitor is used to provide -11.5 VDC from the same 17.5 VAC CT winding that provides the +11.5 VDC output. The -11.5 VDC also provides power for the front panel LED indicator. In all, six DC voltages can be checked by the Tester.

The CAL-69A tester plugs into the power module compartments like any other Tektronix plug-in module. The tester has no adjustments. It consists of mostly passive components and diodes. The two reference voltages can be checked at the three pin jacks on the right side of the front panel. The tests and test voltage limits are listed in **Table 1**. The DC voltages can be read with an accurate digital multimeter (DMM), the AC ripple voltages can be viewed on an oscilloscope.

The tester can be removed from the TM50x compartment by pulling on the silver pull knob next to the LED. (Caution: Be sure to turn the power module off before inserting or removing the tester.)

To conduct a TM50x power module performance check, we used the following test equipment: a Fluke, a Tektronix or Keysight (Hewlett-Packard, Agilent) DMM or an accurate equivalent, and an oscilloscope (for power module ripple voltage test).

If the TM50x power module fails the AC ripple millivolt test on the +33.5 V, +11.5 V and/or -33.5 V rotary switch positions, the TM50x filter capacitors may have failed. You can use an oscilloscope connected at the + DMM – banana jacks J1 and J21 to measure the peak-to-peak ripple voltage. (Note: A DMM on the AC volts range will measure the ripple voltage in VRMS, so you have to convert to peak-to-peak by multiplying by √2 (1.414).

Circuit Description

Figure 1 shows the schematic. The left side of the schematic shows the TM50x power module (in this case the TM501) and its receptacle connector and pin designations. Note that the "A" side of the TM50x connectors has reference letters that indicate the actual letter markings on the receptacle. The "B" side of the receptacle is marked with the numbers shown on the schematic. Tektronix decided it would be less confusing to show the same number for the two adjacent connector pins. I put the receptacle letters in parenthesis on the schematic.

TM50x 25 VAC Supply, pins 1A(A) and 1B—Pins 1A(A) and 1B provide 25 VAC at 0.5 A. The 20 nF capacitor across the connector provides spike and electromagnetic interference (EMI) filtering. This AC voltage is full-wave rectified in the tester by D1 and the AC ripple is filtered by C1. Resistor R1 bleeds off the DC voltage across C1 when the tester is removed from the TM50x compartment.

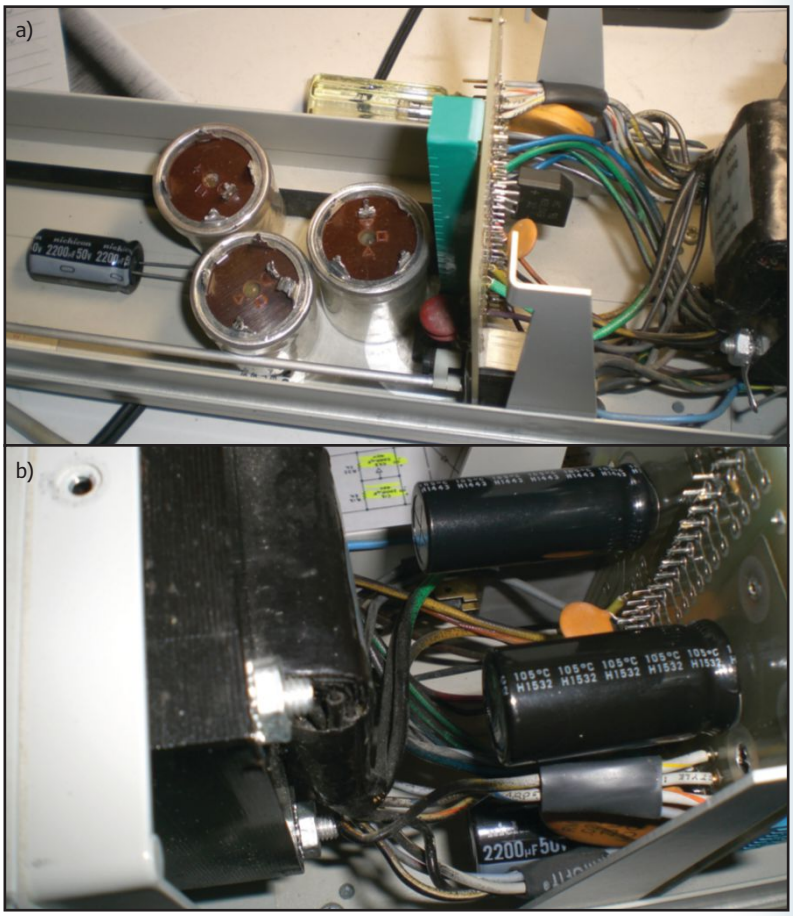


Photo 2: The Tektronix TM501A is shown with three Sprague metal can capacitors removed from the module (a) and the three new replacement capacitors installed on the TM501 PCB (b).

Tests and Test Limits	
Test Jack to DMM (J5, black)	**DC Volt Limits**
J3 Red +22 V	+20.9 to +23.1 VDC
J4 Blue -22 V	-20.9 to -23.1 VDC
Rotary Switch Position	**DC Volt Limits**
+33.5 V	+30.4 to +35.2 VDC
+22 V	+20.5 to +23.5 VDC
+11.5 V	+10.4 to 12.5 VDC
-11.5 V	-10.4 to 12.5 VDC
-22 V	-20.5 to -23.5 VDC
-33.5 V	-30.4 to 35.2 VDC
Rotary Switch Position	**AC Ripple Millivolt Limit**
+33.5 V	100 mVAC maximum
+22 V	200 mVAC maximum
+11.5 V	100 mVAC maximum
-11.5 V	250 mVAC maximum
-22 V	20 mVAC maximum
-33.5 V	100 mVAC maximum

Table 1: The tests and the limits (which need to be repeated for each compartment in a TM50x) are listed.

The DC voltage is used to check the TM-Q12 NPN power transistor mounted to the chassis/heatsink.

Power to the TM-Q12 emitter follower collector is applied to pin 7B through current limit resistor R7. The +22 V base reference voltage circuit at pin 6A(F) consists of R6, Vbe compensation diode D5 and +22 Vref Zener diode VR1. The VR1 reference is fed directly to +22 REF TP test jack J3 for operational check of the tester reference. R15 limits the current in case of an accidental short at J3.

The emitter of TM-Q12 is connected to the tester at pin 7A(H). When switch S1 is not in the +22 position, TM-Q12 is lightly loaded to 1 W by series resistors R8 and R17. In the +22 switch position R17 is shorted by the S1-B2 LOADS section of S1. Q1 is then loaded to 2.2 W by R8. The Q1 emitter follower voltage is read at J1 via S1-A2 VOLTS section of S1 and current limit resistor R5, which introduces less than 0.1% error in the DMM reading.

TM50x +11.5 VDC Supply, pins 2A(B) and 2B—Pins 2A(B) and 2B provide unregulated full-wave rectified +11.5 VDC at 3.6 A. TM-CR30 and TM-CR32 are the rectifiers, TM-C35 is the filter capacitor, and TM-R35 is a bleed resistor for TM-C35 after power is turned off and any plug-ins are removed.

The tester measures the +11.5 VDC voltage out of the TM50x power module via S1-A3. When switch S1 is in the +11.5 position, a 2.6 W load is also placed on that power supply voltage by the combination of TM-R35, and S1-B3 grounding R4 to load down the supply voltage sufficiently to allow a proper ripple voltage test on TM-C35.

TM50x ±11.5 VDC Ground pins 3A(C), 3B, 4A(D) and 4B—The individual ground pins are connected together, along with the ±33.5 VDC ground pins 9A(K) and 9B on the tester's circuit board.

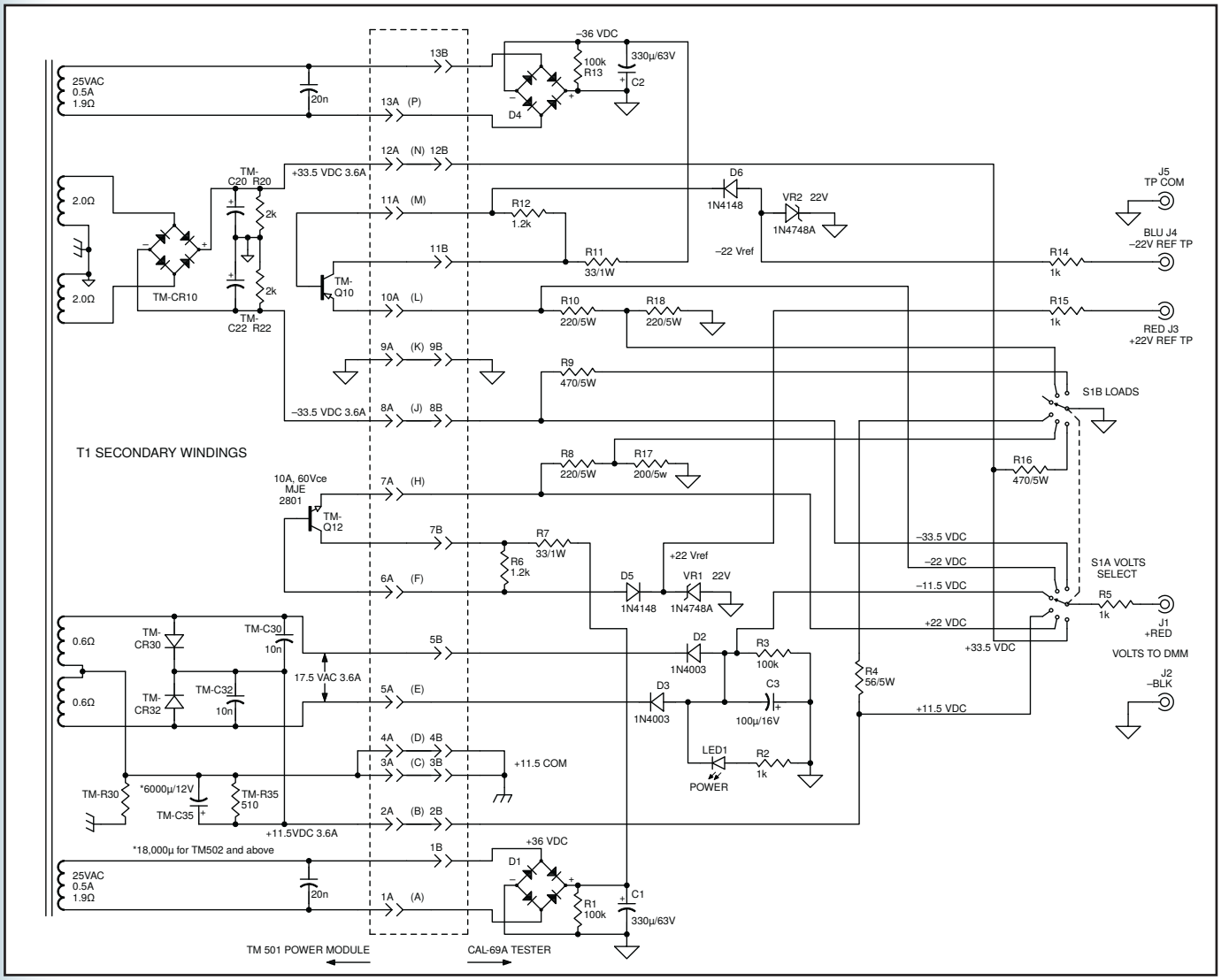


Figure 1: Here is a detailed schematic that should enable you to build your own CAL-69A power module tester.

TM50x +17.5 VAC pins 5A(E) and 5B—This center-tapped AC voltage is used to make the +11.5 VDC unregulated that is output from the TM50x power module at pins 2A(B) and 2B.

In addition, the CAL-69A tester creates a -11.5 VDC supply using D2, D3, filter cap C3, and bleeder resistor R3. This voltage is used to power the red LED1 power indicator on the front panel. It is also measured at S1-A4. No additional DC load is placed on this voltage.

TM50x TM-Q12 pins 6A(F), 7A(H) and 7B—These pins are the connections to TM-Q12 NPN power transistor that is mounted on the TM50x chassis/heatsink.

TM50x -33.5 VDC Supply, pins 8A(J) and 8B—Pins 8A(J) and 8B provide unregulated full-wave rectified -33.5 VDC at 3.6 A. TM-CR10 is the full-wave rectifier, TM-C22 is the filter capacitor and TM-R22 is a bleed resistor for TM-C22 after power is turned off and any plug-ins are removed.

The tester measures the -33.5 VDC directly out of the TM50x power module via S1-A6. When switch S1 is in the -33.5 position, a 2.9 W load is also applied by the combination of TM-R22, and S1-B6 grounding R9 to sufficiently load down the supply voltage to enable a proper ripple voltage test on TM-C22.

TM50x ±33.5 VDC Ground pins 9A(K) and 9B—These individual ground pins are connected together, along with the ±11.5 VDC ground pins 3A(C), 3B, 4A(D) and 4B, on the tester' circuit board.

TM50x TM-Q10 pins 10A(L), 11A(M) and 11B—These pins are the connections to TM-Q1 PNP power transistor that is mounted on the TM50x chassis/heatsink.

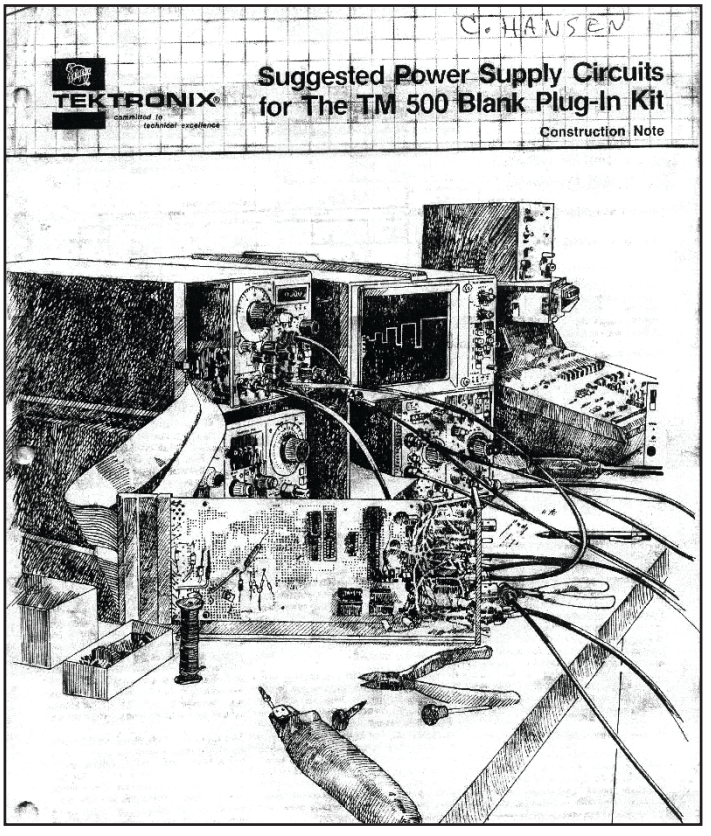


Photo 3: This is the cover of the *Tektronix Power Supply Circuits* manual for the TM 500 Blank Plug-In Kit.

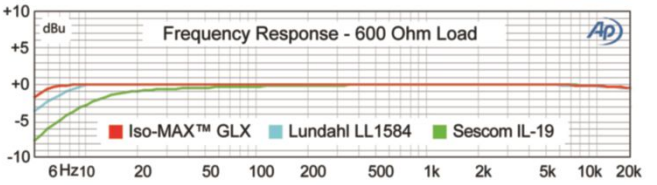
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Distortion @ 20Hz:	.001%	.01%	.1%
Distortion @ +4dB:	.005%	.06%	.3%
Phase shift @ 20Hz:	0°	0°	20°
Ground lift switch:	Yes	No	No
Construction:	Metal	Metal	Plastic
Warranty:	3 Years	1 Year	1 year
MAP Price (estimated street)	$69	$130	$45



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About the Author

Chuck Hansen is an Electrical Engineer who works as a consultant in the aerospace industry. He holds five patents in his field of engineering. Chuck has written two books for Audio Amateur publications and he has more than 260 magazine articles to his credit. He began building vacuum-tube audio equipment in college. He enjoys sailing and playing jazz guitar. He likes to modify guitar amplifiers and effects to reduce noise and distortion, as well as building and restoring audio test equipment.

TM50x -33.5 VDC Supply, pins 8A(J) and 8B—Pins 12A(N) and 12B provide unregulated full-wave rectified +33.5 VDC at 3.6 A. TM-CR10 is the full-wave rectifier, TM-C20 is the filter capacitor and TM-R20 is a bleed resistor for TM-C20 after power is turned off and any plug-ins are removed.

The tester measures the +33.5 VDC directly out of the TM50x power module via S1-A1. When switch S1 is in the +33.5 position, a 2.9 W load is also applied by the combination of TM-R20, and S1-B1 grounding R16 to load down the supply voltage sufficiently to allow a proper ripple voltage test on TM-C20.

TM50x 25 VAC Supply, pins 13A(P) and 13B—Pins 13A(P) and 13B provide 25 VAC at 0.5 A. The 20 nF capacitor across the connector provides spike and EMI filtering. This AC voltage is full-wave rectified in the Tester by D4 and the ac ripple is filtered by C2. Resistor R13 bleeds off the DC voltage across C2 when the Tester is removed from the TM50x compartment. This DC voltage is used to check the TM-Q10 PNP power transistor mounted to the TM50x chassis/heatsink.

Power to the TM-Q10 emitter follower collector is applied to pin 11B through current limit resistor R11. The -22 V base reference voltage circuit at pin 11A(M) consists of R12, Vbe compensation diode D6 and -22 Vref Zener diode VR2. The VR2 reference is fed directly to -22 REF TP test jack J4 for operational check of the Tester reference. R14 limits the current in case of an accidental short at J4.

The emitter of TM-Q10 is connected to the CAL-69A tester at pin 10A(L). When switch S1 is not in the -22 position, TM-Q10 is lightly loaded to 1 W by series resistors R10 and R18. In the -22 switch position R18 is shorted by the S1-B5 LOADS section of S1. TM-Q10 is loaded to 2.2 W by R10. The TM-Q10 emitter follower voltage is read at J1 via S1-A5.

CAL-69A Power Module Tester Parts List

Symbol	Value	Description	PC Span	Manufacturer
C1-C2	330 uF, 63 V, 20%	Aluminum cap, 105°C	2.0	Vishay
C3	100 uF, 16 V, 20%	Aluminum cap, 105°C	1.1	Vishay
D2-D3	1N4003	Diode, 200 V, 1 A	0.4	Fairchild
D1, D4	W02G	Diode bridge, 200 V, 1.5 A	0.3 x 0.4	Vishay
D5-D6	IN4148	Diode, 80 V, 0.5 A	0.5	Fairchild
J1		Banana Jack, Red		Emerson
J2		Banana Jack, Black		Emerson
J3		Tip Jack 0.08, Red		Emerson
J4		Tip Jack 0.08, Blue		Emerson
J5		Tip Jack 0.08, Black		Emerson
(All Resistors, 5% tolerance)				
R1, R3, R13	100 kΩ, 0.5 W	Carbon Film	0.5	Xicon
R2, R5, R14, R15	1 kΩ, 0.5 W	Carbon Film	0.5	Xicon
R4	56 Ω, 5 W	Wirewound	1.3	Dale/Vishay
R6, R11	33 Ω, 2 W	Metal Oxide	1.1	Xicon
R7, R12	1.2 kΩ, 0.5 W	Carbon Film	0.5	Xicon
R8, R10	220 Ω, 5 W	Wirewound	1.3	Dale/Vishay
R9, R16	470 Ω, 5 W	Wirewound	1.3	Dale/Vishay
S1		6P ST Rotary Switch		Lorlin CK1030
VR1, VR2	1N4748A	Zener, 22 V 1 W 5%	0.5	Fairchild
	Vector 3662-2 plug board Plastic knob, 0.75"			Vector (special)

Construction

I salvaged a Tektronix plug-in chassis from a non-working plug-in from Pulse Engineering. I found these are readily available online as non-working units. Other manufacturers who also made non-Tektronix plug-ins are Ball-Efratom, Laser Precision, Metrotek, MOS Aid, Spectracom, Signetics, and others.

Tektronix used to offer custom auxiliary blank plug-in breadboard kits with chassis for people or organizations who wanted to make custom plug-ins compatible with the TM501 and TM502 power modules, but with functions particular to their own purposes. The units, Tektronix models 040-0652-01 and 040-0754-00, respectively, had a Vector-style perf board with a card-edge connector that aligned with the fixed receptacles in the power modules. However, they are no longer offered. **Photo 3** shows the cover of the *Tektronix Power Supply Circuits* manual for the TM 500 Blank Plug-In Kit.

Fortunately, Vector still makes a perf board (Part Number 3662-2) plug board that, with a slight modification, will align with the TM50x receptacle.

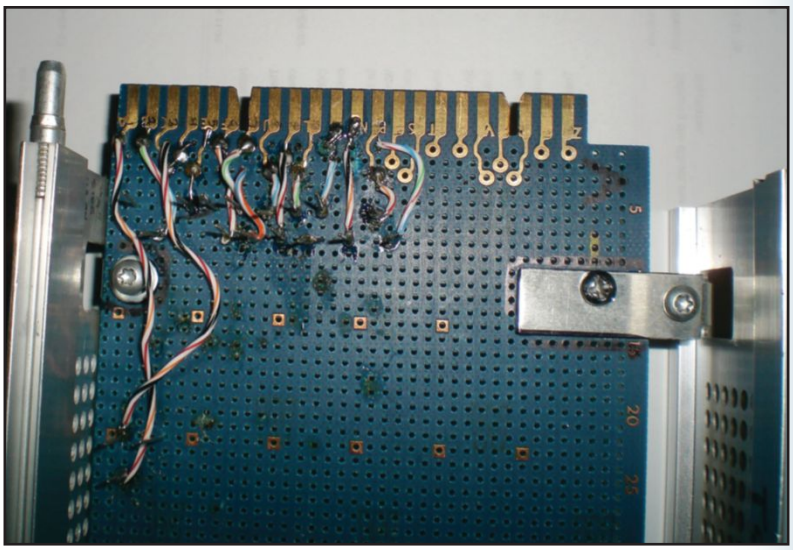


Photo 4: I made several modifications to the Vector plug board.

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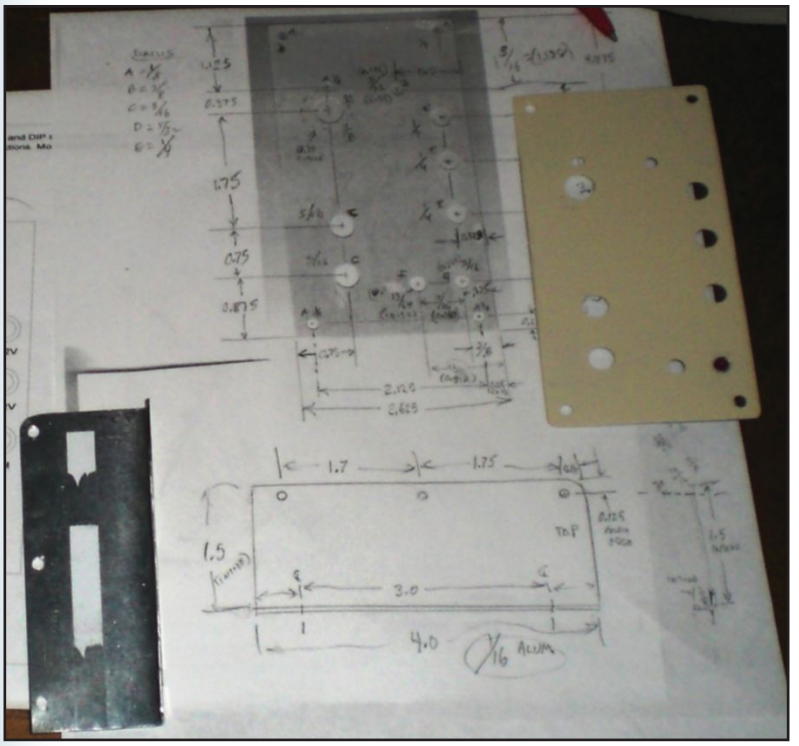


Photo 5: I made a right angle bend to attach to the front control panel. Pictured are the front mounting bracket, the front control panel, hand drawn dimensions, and drill codes.

It is a bit shorter in overall length than the original Tektronix perf board, but that would not be an issue with most custom breadboards that you would want to make. The plug board is a non-stocked special order at Mouser.

The modification requires cutting off the bottom (pin 1, A) side of the perf board (see **Photo 4**). I found the easiest way to do this is to insert two pieces of 1′ × 2′ trim boards between the jaws of a wood-workers vise. The boards need to be a bit longer than the plug board. Clamp the bottom edge of the plug board in the vise right where the cut has to be. Then, score the front and back of the board with a utility knife blade, using the top edges of the boards as a guide. With the board still clamped in the vise, press it to one side and it will snap across the scored lines.

(Caution: You have to score between the rows of holes to prevent cutting off the solder pad for edge connector pin 1. Score the plug board cut line deep enough so it won't accidentally snap off along a row of holes. Use the square solder pads across the plug board to help as a reference location for where to clamp the plug board.)

Next, drill two holes in the bottom of the plug board as shown, to align with the existing PCB mounting tabs. The back edge of the connector must be exactly 0.75″ out from the back edge of the plug-in aluminum chassis sides.

The plug board does not reach the two top mounting pads on the chassis. I made two 0.5″ × 2″ pieces of 1/16″ thick aluminum to connect

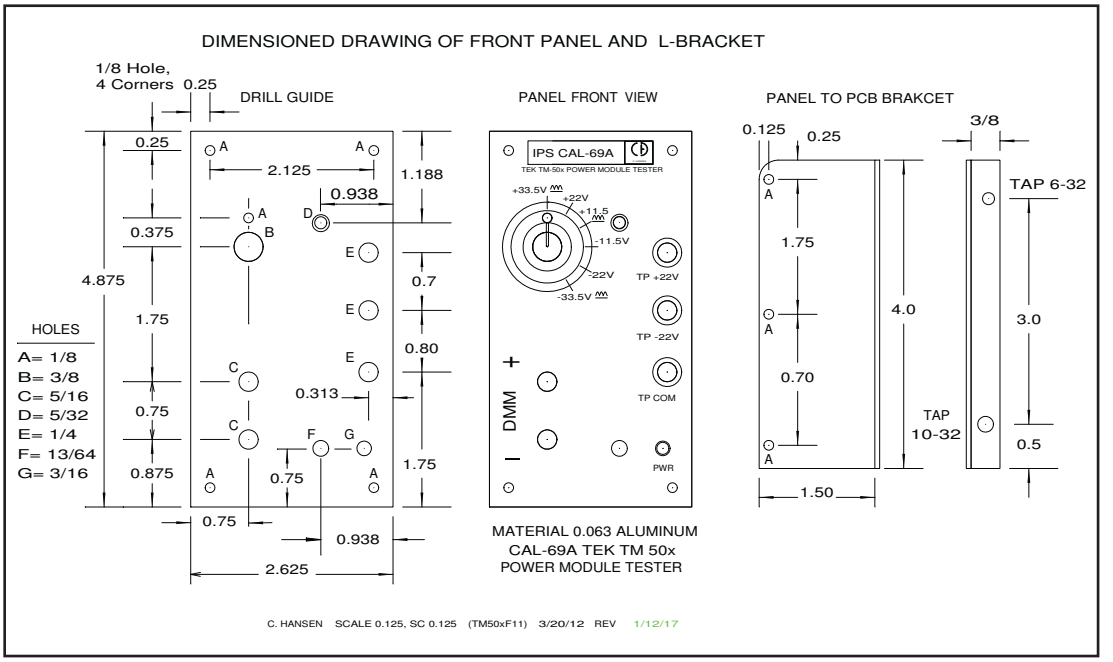


Figure 2: This is a dimensioned drawing of the two aluminum panels (the front panel and the L-bracket).

the upper part of the plug board to the top PCB mounting tabs. Since the plug board is not long enough to reach the front panel of the Tektronix plug-in chassis, I made another aluminum L-bracket to connect the back edge of the plug board to the front control panel. I attached the front end of the bracket to the back of the front panel with one 6/32 flat head screw on top and an 8/32 knurled knob at the bottom. The knurled knob is spaced away from the front panel by a thick #8 washer to be used as a pull to remove the CAL-69A power module tester from the TM50x power module bay. This was previously done with a nylon latch release pull on the bottom-left of the control panel, but time had deteriorated the nylon so much that the front tab broke off from the release mechanism inside the chassis. I removed the remains of the latch so it would not interfere with the tester.

The L-bracket started out as a 2.5″ × 4″ piece of 1/16″ aluminum. I made a 3/8″ deep right angle

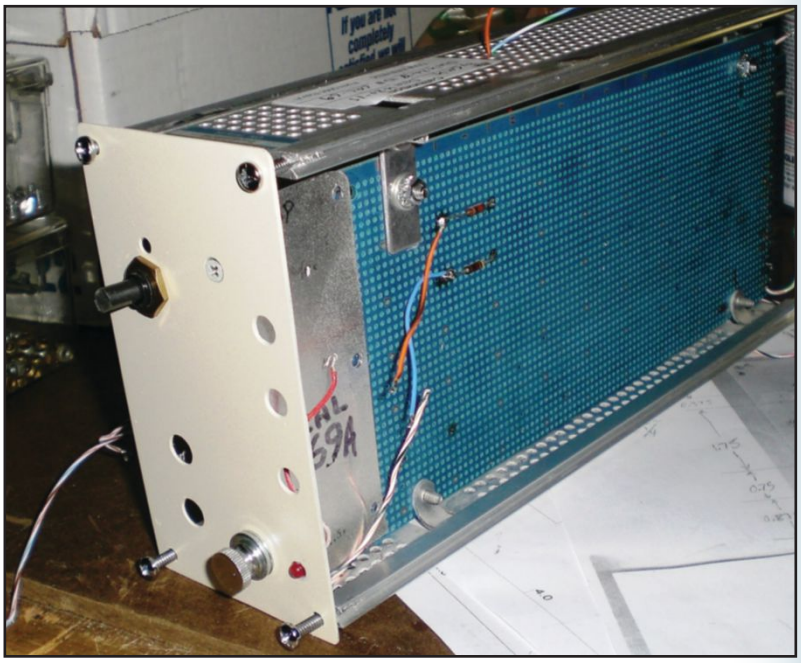


Photo 6: I checked the alignment for the front panel and the front L-bracket hardware.

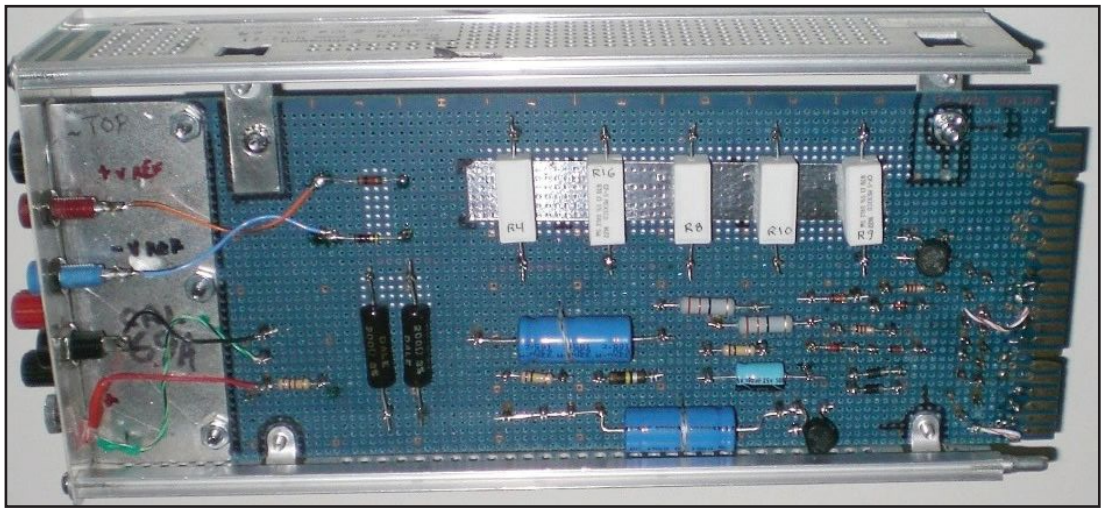


Photo 7: Here you can see the plug board components.

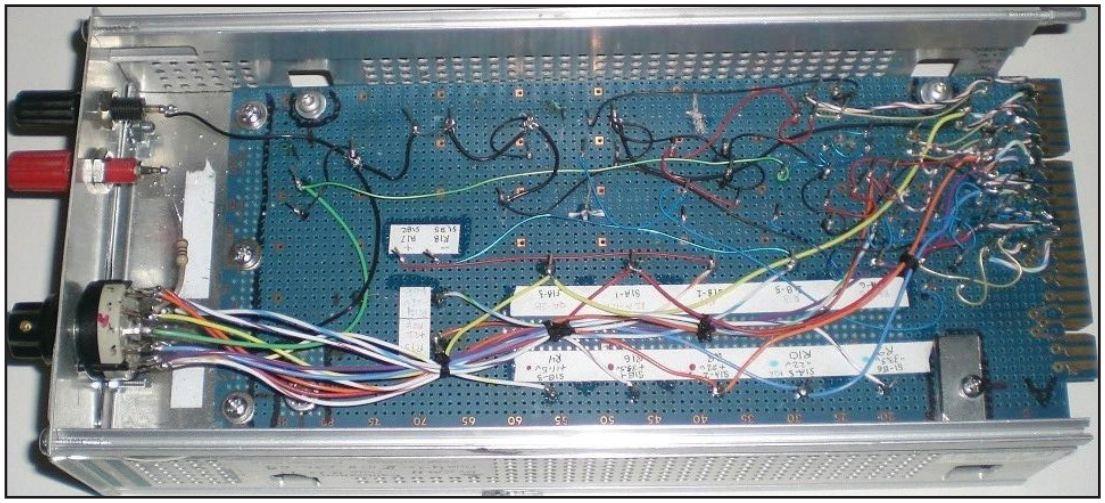


Photo 8: This is the wiring side of the tester.

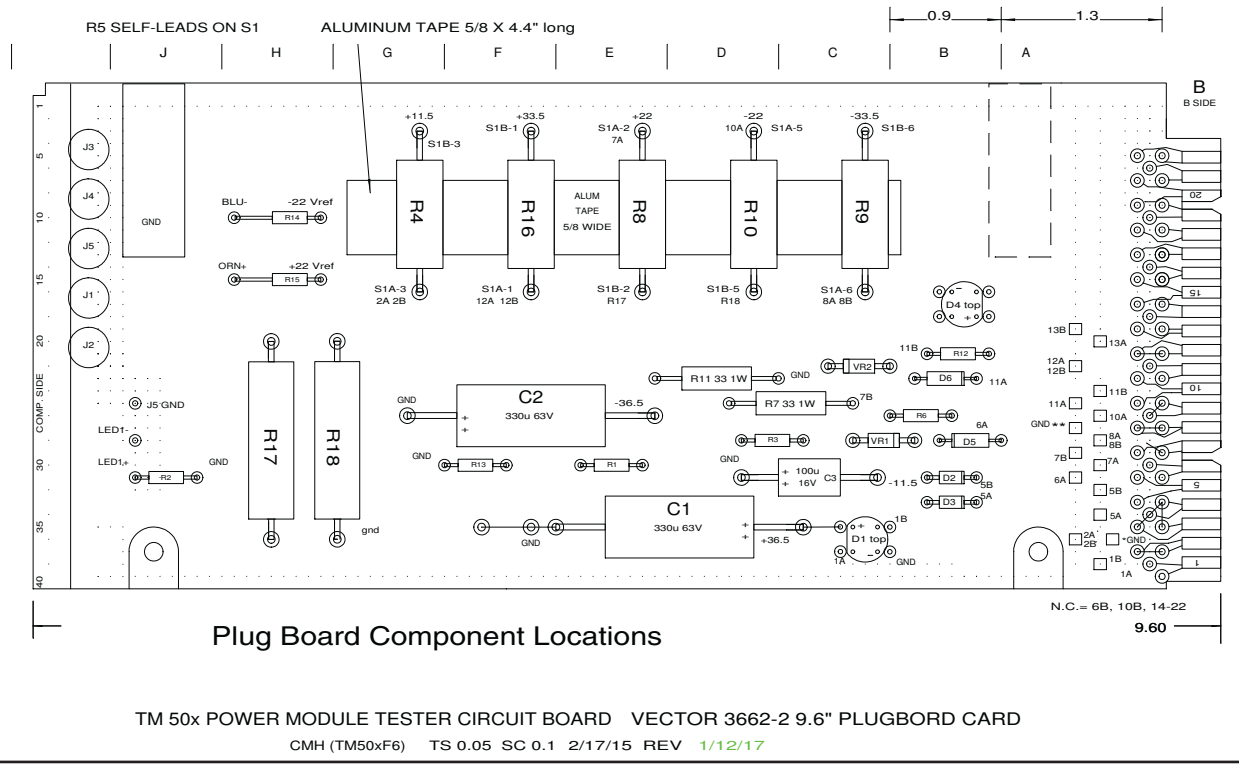


Figure 3: This drawing details the plug board component locations.

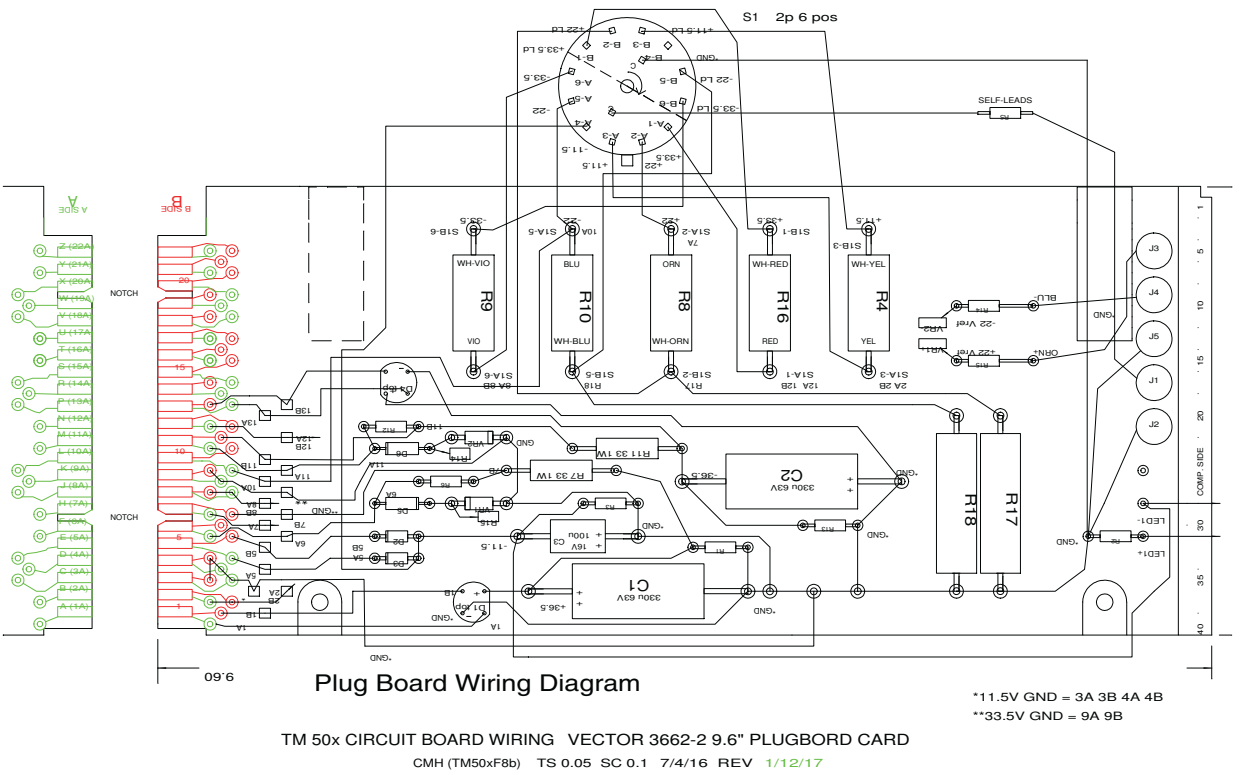


Figure 4: The diagram indicates the proper wiring locations for the tester.

bend to attach it to the front control panel (see **Photo 5**). The front control panel is a 2-5/8" × 4-3/8" flat aluminum plate. A dimensioned drawing of the two aluminum panels is shown in **Figure 2**.

The plug board's card edge has to be notched in two placed to mate with receptacle keys in the TM50x power modules. I did this with my band saw. You can see the notches shown in **Photo 4**, placed between contacts 6 and 7, which is needed for all TM50x power modules. There is another notch between contacts 19 and 20, which is needed for some TM50x power modules. Be careful not to remove any of the gold-plated contact material. Then check that there is no continuity between the two adjacent contacts or between the front and back contacts in the same locations on the plug board.

Photo 6 shows the hardware alignment check I made between the front control panel blank, the L-bracket and the plug board. Once everything was properly aligned, I drilled the L-bracket mounting holes into the back end of the plug board, using the three pre-drilled holes in the L-bracket. I used the existing 6-32 tapped channels in the top and bottom sections of the chassis for my new front panel.

Once I had the front panel, L-bracket, and plug board aligned to the plug-in chassis, I did a fit-check into a TM501 power module and verified the plug board's card-edge connector properly mated with the power module, with no shorts between any adjacent contacts.

Next, I wired the plug board and installed all the perf board components. I used Vector T49 push-in terminal pins since there were no copper pads in the proper locations. I wired the tester from the card-edge contact pads to the first rear row of T49 pins with AWG 24 Teflon covered wire. Some of the TM50x receptacle pins are shorted together for higher current capability, so I shorted card-edge connector pads in the same location together with bare AWG 22 soft-drawn tin-plated copper hook-up wire (Alpha Part Number 298). All of the hard-wired connections from the front panel to the plug board were also made with color-coded Teflon wire.

I made the lettered front panel with my CAD program and printed it to mat-finished clear Chartpak DAF8 permanent adhesive-backed film. The plug board component locations are shown in both **Figure 3** and **Photo 7**. **Figure 4** details how to do the wiring and **Photo 8** shows how the wiring should look once it is completed.

The CAL-69A module tester can be used with the more than 100 Tektronix 500-series plug-ins (see **Photo 9**). I have included an Excel file listing all of the plug-ins in a Supplementary Material file found on the audioXpress website.



Photo 9: This is the completed, calibrated CAL-69A power module tester.

Project Files

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

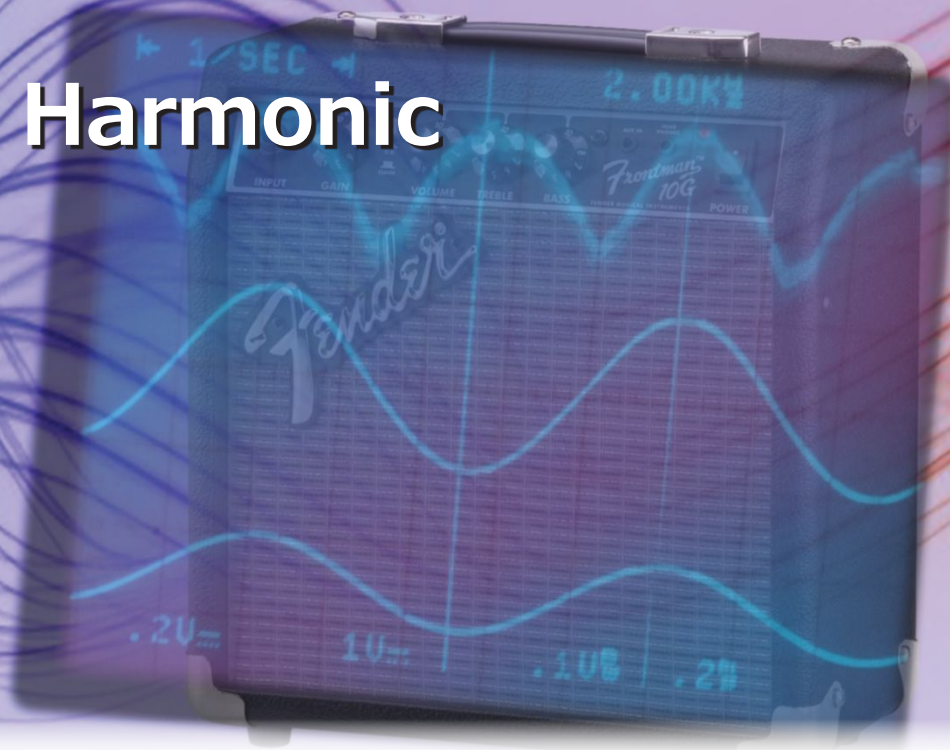
Resources

"List of TM500 and TM5000 equipment," TekWiki, http://w140.com/tekwiki/wiki/List_of_TM500_and_TM5000_equipment.

Tektronix, Inc., www.tek.com

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Measuring Harmonic Distortion



Learn more about harmonic distortion and discover a few techniques you can use to measure it in your own audio signals.

By
Jeff Smith
(United States)

Harmonic distortion consists of frequencies that are integer multiples of an input signal that appear as a result of signal processing. You may or may not want it along with your signal. Either way, it's educational to see and measure it. It helps you understand the relationship between audio as a signal and audio as a sublime rendering of your last electronic dance music (EDM) concert . . . or dulcimer recital.

In this article, I will describe two methods of harmonic distortion measurement. To perform these measurements, you'll need a function generator or low distortion oscillator and an oscilloscope with at least two vertical input channels. You can build a simple circuit to measure signals with more than about 0.1% harmnic distortion. With a low distortion oscillator and a notch filter, you can measure down to about 0.003%. With the right oscillator, notch filter, and spectrum analyzer, you can accurately measure 1 ppm harmonic distortion.

VRMS	Ratio	dBVRMS	Percent
31.6 V	30:1	+30	3,160%
3.16 V	3:1	+10	316%
2 V	2:1	+6	200%
1 V	1:1	0	100%
707 mV	0.7:1	-3	70.7%
501 mV	1:2	-6	50.1%
316 mV	1:3	-10	31.6%
100 mV	1:10	-20	10%
10 mV	1:100	-40	1%
100 µV	1:10,000	-80	0.01%
1 µV	1:1,000,000	-120	0.0001%

Table 1: This chart describes the relationship between VRMS, ratios, decibels, and percent.

Amplitude measurement in decibels expresses the way we hear loudness. It's not a linear function; it's logarithmic. Referenced to 1 VRMS = 0 dBVRMS, the relationships between dBV and RMS volts are:

$$20 \times \log (\text{VRMS}) = \text{dBVRMS}$$
$$10\exp((\text{dBVRMS})/20) = \text{VRMS}$$

A device whose output amplitude is larger than its input is said to have positive gain and is a positive number when expressed as a unit—or dimensionless (e.g., V_{PP}/V_{PP}) ratio in decibels. If the device's output amplitude is smaller than its input, it's said to have negative gain or attenuation and is a negative number in decibels.

Working with amplitude ratios in decibels is sometimes more convenient than in voltage or percent, once you get the hang of it. **Table 1** shows the relationship between VRMS, ratios, decibels, and

percent. Although many of the relationships are not exact, they're almost always close enough to be useful and the inaccuracies are small. In most situations where you would multiply or divide in volts, you just add or subtract in decibels.

Decibels is the ratio of two amplitudes, without a reference point whereas dBVRMS is referenced to 1 VRMS. 1 VRMS = 0 dBVRMS. For example:

- For a power amp with +26 dB gain: +20 dB + 6 dB = (voltage gain of 10) x (voltage gain of 2) = voltage gain of 20

Referenced to 1 V (0 dBV):

- Harmonic distortion of 100 µVRMS = 100 µV/1 V = 0.01% HD = -80 dB
- Harmonic distortion of 50 µVRMS = 50 µV/1 V = 0.005% = 0.01% × 0.5 = -80dB - 6dB = -86dB.
- Harmonic distortion of 400 µVRMS = 400 µV/1 V = 0.04% = 0.01% × 2 × 2 = -80 dB + 6 dB + 6 dB = -68 dB

Most harmonic distortion measurements here are just the ratio of harmonics to signal output level, hence just decibels.

Difference Circuit—Crude But Simple

A simple way to see harmonic distortion on an oscilloscope uses a circuit that subtracts the original signal from your device under test (DUT) output and shows what's left, which are the frequencies and noise that were added. This is useful to get an idea of what the signals look like and for troubleshooting circuits that probably have 0.1% or higher harmonic distortion (e.g., tube power amplifiers). It can be used with music as well as static signals. Some power amplifiers use this idea to determine if the output is significantly different from the input. If a difference is detected, it might flash an LED to alert the user to overload conditions or suggest reduction in amplifier gain.

You'll need a common function generator with a sine wave output. The circuit uses components that will be tailored to your DUT and its environment. The components include op-amps, a power supply for them, a prototyping breadboard, resistors, maybe some capacitors, and an oscilloscope to observe and analyze the DUT input, output, and the difference between them. I'll contribute the circuit equations. Admittedly, using a function generator is not a precise way to measure harmonic distortion, but it'll give you a general idea of your gear's harmonic distortion performance. The circuit takes two forms: one for DUTs whose outputs invert the signal and one for DUTs that don't invert.

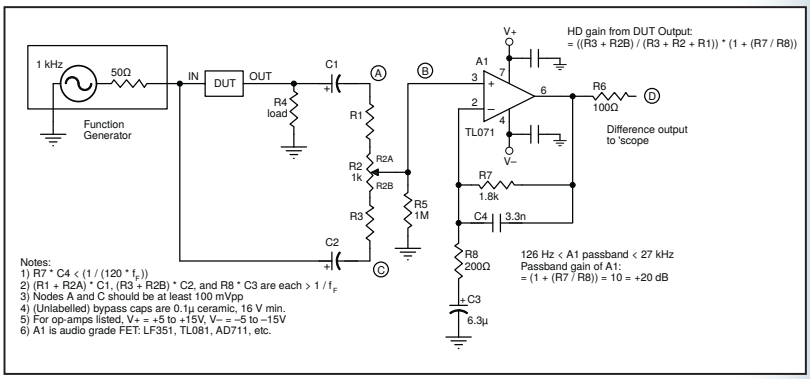


Figure 1: The schematic shows the difference circuit for a device under test (DUT) whose output inverts the input signal.

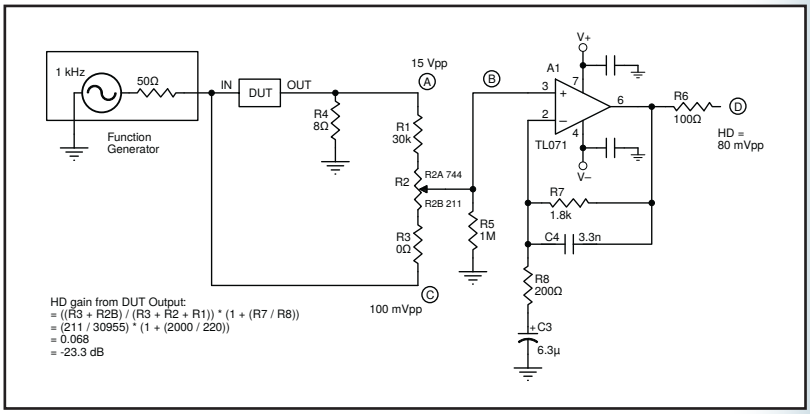


Figure 2: The schematic shows the difference circuit for tube guitar amplifier: an inverting amp.

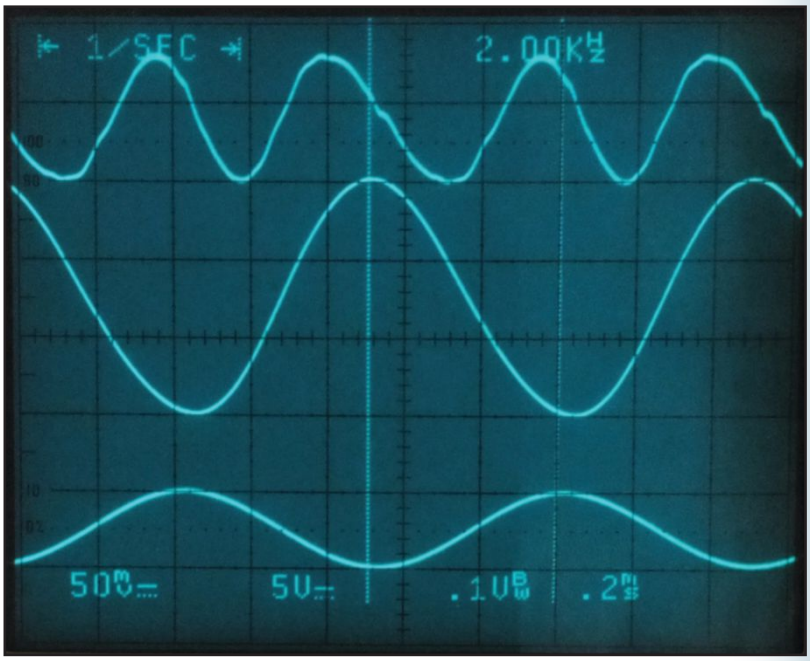


Figure 3: The oscilloscope photo shows the difference circuit waveforms for tube guitar amplifier. (Top: Node D. Middle: Node A. Bottom: Node C)

Difference Circuit for the Inverting DUT

Figure 1 shows the schematic of the difference circuit for inverting the DUT. Signal levels should be at least 100 mVpp. Don't forget the op-amp bypass capacitors. They keep your op-amps stable. Mount them close to the op-amp. You can omit or shunt C1 and C2 if the DC offsets of the generator and the DUT are less than 200 mV. Otherwise, leave them both in. Measure your offset for appropriate polarity.

The idea of R1, R2, and R3 is to balance the contributions of the function generator and the DUT output signals so that the fundamental cancels, and the DUT noise and distortion components are all that's left. R5 just sets the DC bias for A1. Observe point B on the oscilloscope before inserting A1 into the circuit. Adjust R2 for minimum signal amplitude. If the signal is less than 1 V_{PP}, put A1 into the circuit and monitor A1's output for better clarity.

R8 and C3 attenuate 60 Hz noise and DC offset. R7 and C4 attenuate high-frequency noise. These components depend on fundamental frequency (f_F). The values shown are for f_F = 1 kHz. Keep the time constants (R1 + R2A) × C1, (R3 + R2B) × C2, R8 × C3 each greater than $1/f_F$ and keep R7 × C4 less than $1/(120 \times f_F)$ to ensure you're not attenuating frequencies of interest.

Using an oscilloscope, monitor your DUT input and output with music playing to get an idea of the average signal levels. Leave the controls where they are to measure harmonic distortion at your normal listening level. Keep or substitute into the difference circuit whatever resistive load the DUT sees in your system (e.g., 4 to 8 Ω for a power amp, 10 to 100 kΩ for a preamp).

Determine the values for R1, R2, and R3 from the amplitudes at nodes A and C. Compute equal currents from the nodes to the trimpot wiper. Pick resistor values that will result in a current between 100 and 500 µA. For example:

$$\frac{A}{(R1+R2A)} = \frac{C}{(R3+R2B)} = 300\mu A$$

Observe resistor power ratings for high voltage outputs. Although the fundamental will ideally be nulled at the trimpot wiper, the attenuation of the harmonic distortion components is just a voltage divider from node A to Node C, with the wiper being the divider tap. Compute the gain from DUT output to A1's output as:

$$\left(\frac{(R3+R2B)}{(R3+R2+R1)}\right)\times\left(1+\left(\frac{R7}{R8}\right)\right)$$

If there's so much phase shift or time delay that you can't null out the fundamental, you'll need to either test at a different frequency, adjust the tone controls on your DUT, or use a different test method. Any noise at the DUT output that is not from the signal source will appear along with the harmonic distortion.

I used the circuit shown in **Figure 2** to measure the harmonic distortion of 1 kHz through a tube guitar amplifier.

For my tube guitar amplifier with a gain of 150: R1 = 30 kΩ, R3 = 0 Ω. R2's approximate wiper position is such that R2A = 744 and R2B = 211. Then:

$$\frac{A}{(R1+R2A)} = \frac{C}{(R3+R2B)} = \frac{15}{(30744)} = \frac{0.1}{(211)}$$

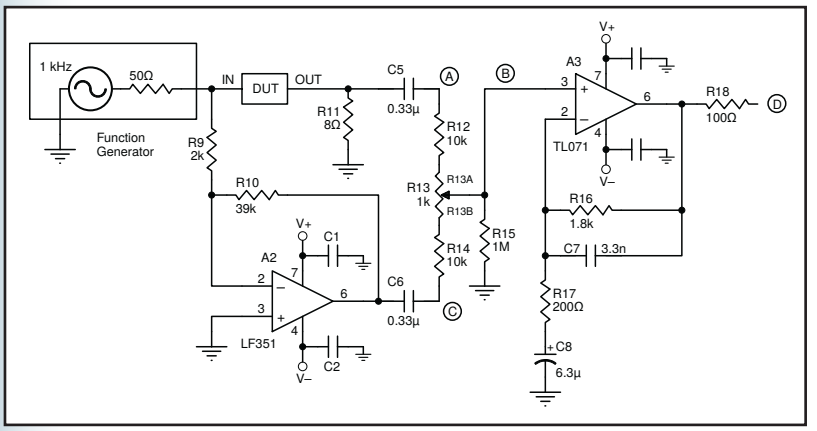


Figure 4: The schematic shows the difference circuit for a solid-state guitar amp: a non-inverting amplifier.

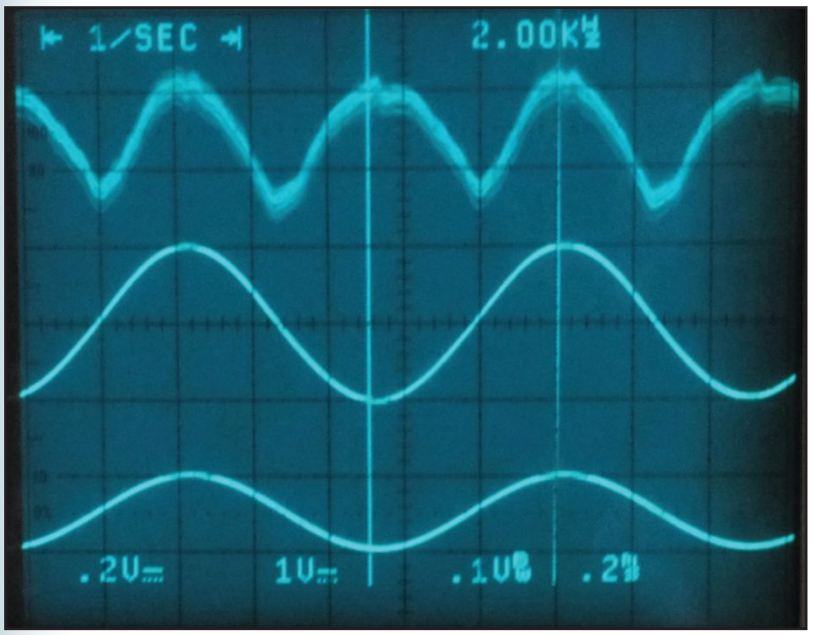


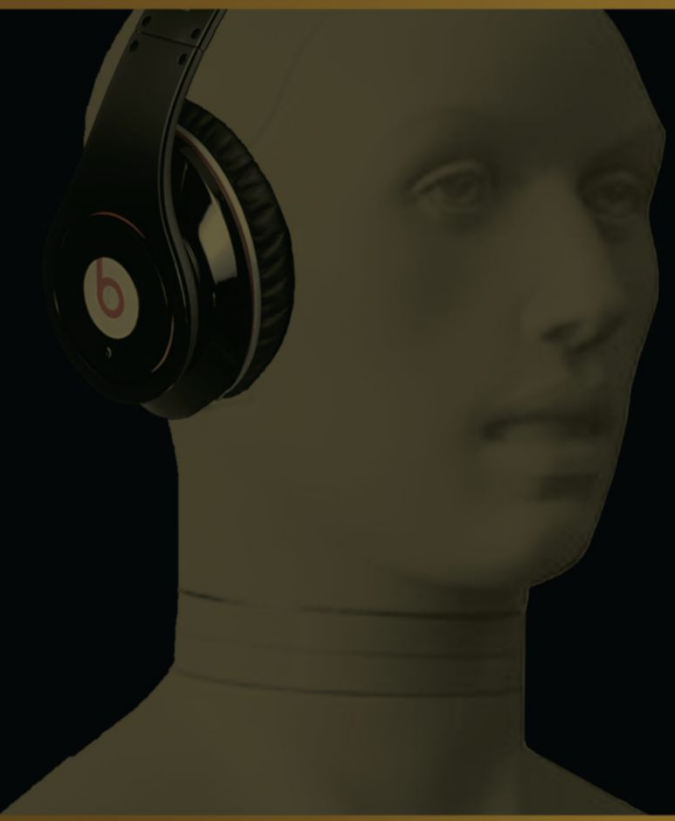
Figure 5: The oscilloscope photo shows the difference circuit waveforms for the solid-state guitar amplifier. (Top: Node D. Middle: Node A. Bottom: function generator output.)

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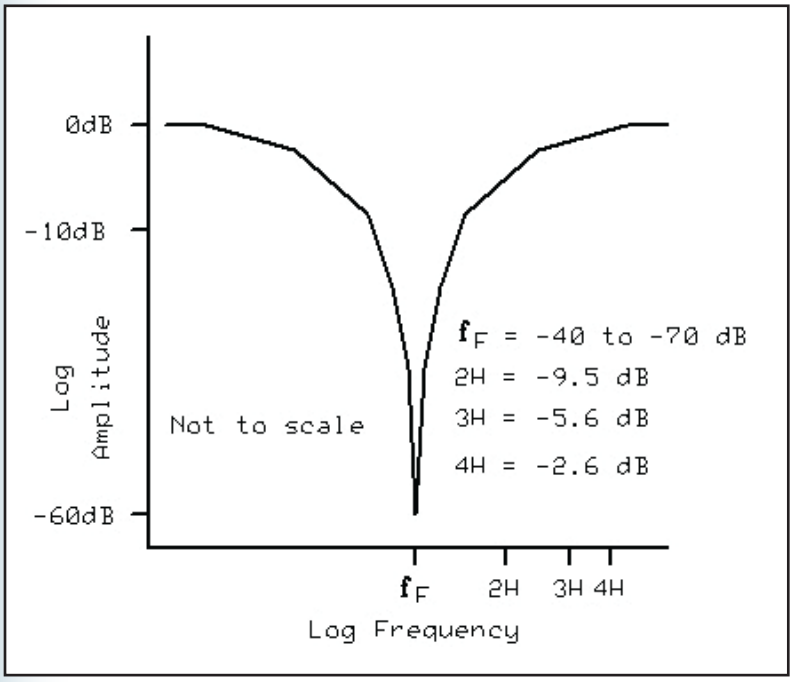


Figure 6: This is the notch filter frequency response. The fundamental frequency is greatly attenuated, the harmonic distortion frequencies not so much.

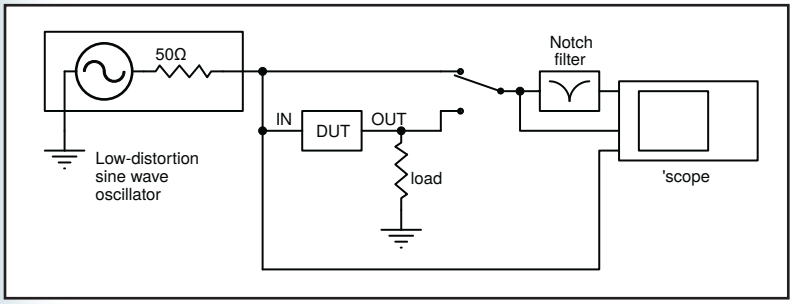


Figure 7: Here is the harmonic distortion test setup with the notch filter.

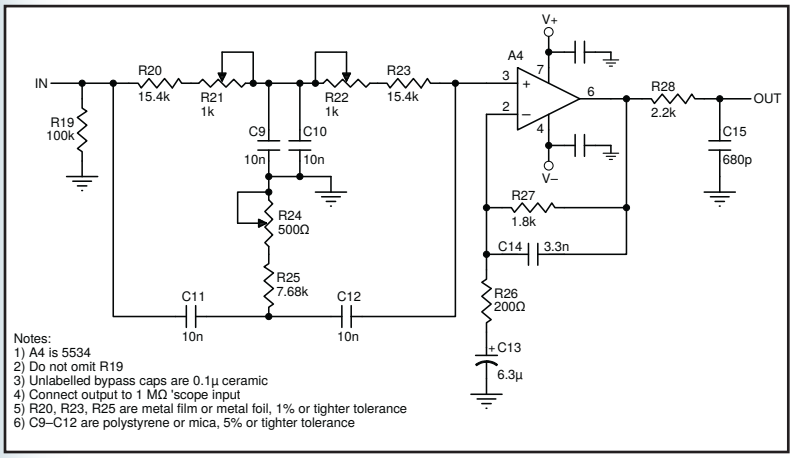


Figure 8: Use this schematic for the 1 kHz notch filter followed by +20 dB gain.

Figure 3 shows the waveforms of the tube guitar amplifier's difference circuit. A1's output is almost pure second harmonic distortion (2 kHz) at 80 mVpp. The gain from the DUT output to A1 output is 0.068. The harmonic distortion at the DUT output is: 0.08/0.068 = 1.17 V

The harmonic distortion referred to the DUT output is: 1.17 V/15 V = 7.8% (-22.1 dB). Rockin!

Difference Circuit for Noninverting DUT

Most of the previous suggestions apply here. Another op-amp is needed for this circuit to invert the generator output. The inverter gain can be set so that its output amplitude more closely matches that of the DUT as long as the inverter output doesn't add distortion to the signal by clipping, slew rate, or bandwidth constraints. Other than amplitude, the inverter output must be an accurate replica of its input. **Figure 4** shows the circuit used to measure a solid-state guitar amp.

Figure 5 shows waveforms of the solid-state guitar amp difference circuit.

Low-Distortion Oscillator

Most function generators have about 1% harmonic distortion, most of which is nulled out in the difference circuit. For harmonic distortion measurements below about 0.1% (-60 dB), you need a low-distortion oscillator. Linear Technology's datasheets describe several low-distortion oscillators designed by Jim Williams. We send this signal through the DUT and then, as before, arrange to . . . remove it. But if the harmonic distortion is so small that an oscilloscope can't display it, we need to add another tool.

Notch Filter

A pure sine wave consists of a single frequency. A notch filter can attenuate that frequency by 60 dB or more without really affecting nearby frequencies. A filter designed and tuned for our purpose will attenuate second harmonic distortion by 9.5 dB (i.e., -9.5 dB), third harmonic distortion by 5.6 dB, and fourth harmonic distortion by 2.6 dB (see **Figure 6**).

The downside of this steep notch is that the frequency of maximum rejection will vary over time, due to the components' coefficient of temperature and temperature gradients between them. The same is true of an analog RC oscillator: the output frequency will drift for the same reasons. There are ways to minimize this problem. For example:

- Use components with low coefficient of temperature

- Minimize temperature gradients between components by waiting for warm-up, physical barrier, potting, Styrofoam, wadded newspaper
- Add a circuit that locks onto the fundamental frequency

Remember that for our purpose, a notch filter removes just one frequency—the fundamental. You'll be left with not only harmonics but also circuit noise, AC line noise, and anything else that's within the passband of the filter. This can be a problem if you need to see harmonics below about -100 dB in the time domain, but it can also be a good thing. It can help analyze ground loops or tell you if your dynamic-dependent signal processor operates differently with and without a signal.

Figure 7 shows the setup with the filter. With the right components, a suitable notch filter is not hard to build. **Figure 8** shows a passive notch filter followed by +20 dB gain and additional filtering. Connect the output to 1 MΩ oscilloscope input.

Xicon 23PS310 10nF 5% polystyrene capacitors look suitable for C9–C12 (see Sources). This design will notch a range of frequencies around 1 kHz. The component values are ideal. If your low distortion oscillator frequency doesn't drift much, you can alter the fixed resistor values so that you can use potentiometers of smaller value, which makes tuning much easier—10-turn wirewound potentiometers are great for this. Watch the fundamental recede into the noise floor on the oscilloscope as you slowly tweak the potentiometers. Feel the analog love.

As with the difference circuit, if your output signal is larger than 10 V_{PP}, make sure the notch is tuned reasonably close to the fundamental by adjusting it first with the switch in the position shown so you don't damage the filter's op-amp.

The notch filter results are less clear with extremely linear amplifiers. With the gain of 10 (A4), the typical oscilloscope waveform of the filtered output will look fuzzy with noise. More gain just gives you a bigger fuzzy waveform. And the waveform may look choppy or stupidly lumpy if it's a combination of two or more frequency components. You can design low-pass and high-pass filters for noise as long as they don't attenuate the frequencies of interest, and bandpass filters to zero in on individual harmonics. It can be done, but shouldn't you get out more?

If you're this far into it and you're willing explore software, go for the hot setup. The best way to see all the details (except phase) on one display with precision and a low noise floor is to add a spectrum analyzer to our test system to monitor the notch filter output.

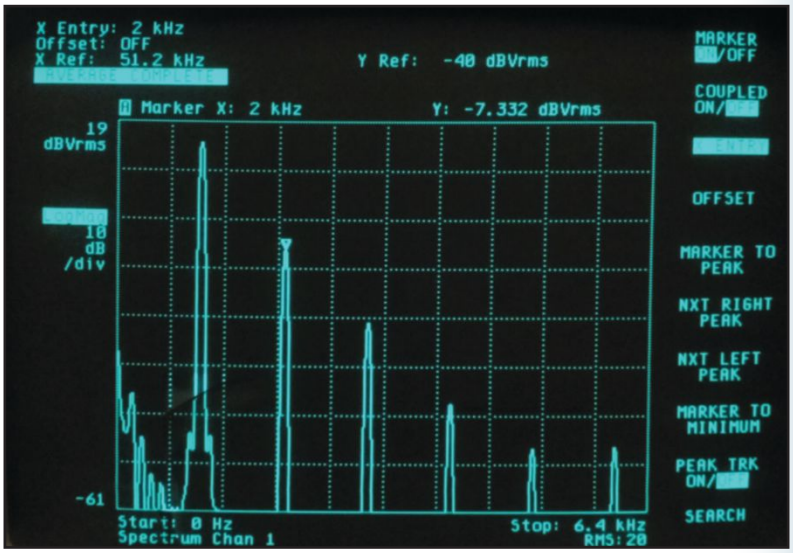


Figure 9: The spectrum analyzer display shows the tube guitar amp output with 1 kHz at +15.0 dBV and 2 kHz at -7.3 dBV.

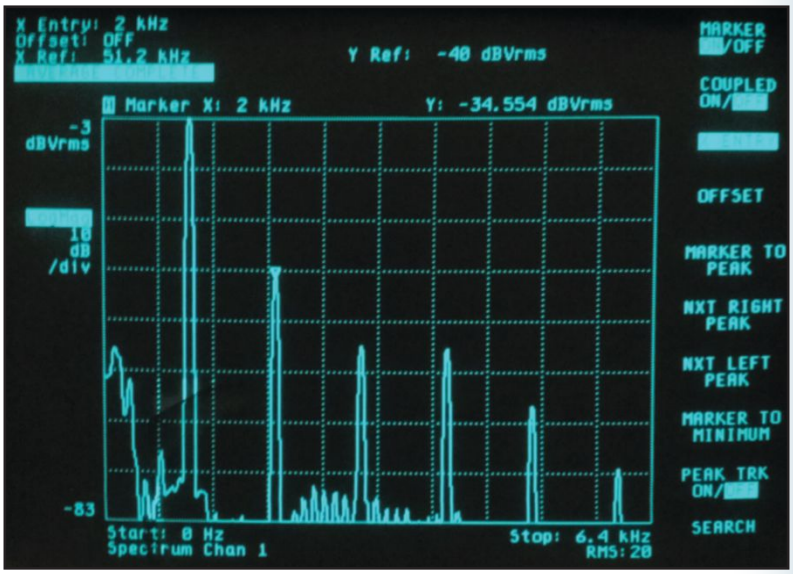


Figure 10: This is the spectrum analyzer display of solid-state guitar amp output.

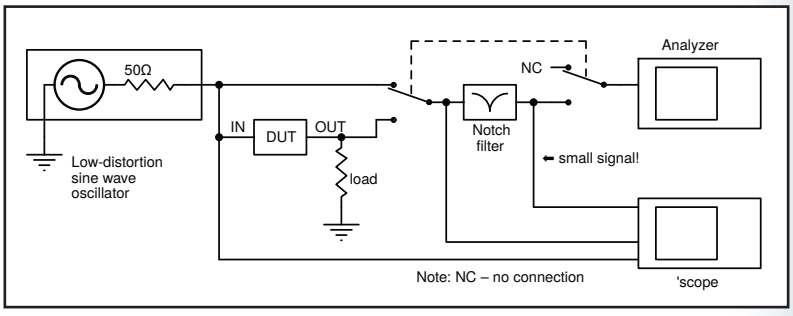


Figure 11: The hot setup consists of a low harmonic distortion oscillator, a notch filter, an oscilloscope, and a spectrum analyzer.

About the Author

Jeff Smith explores retirement by conceiving, designing, building, enjoying, abusing, and then ritualistically destroying his hi-fi, test, and music equipment.

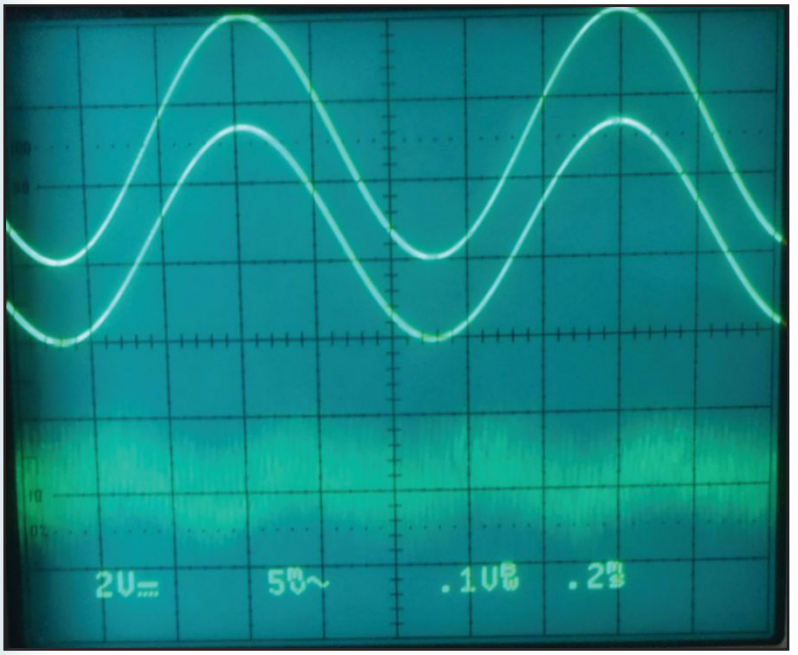


Figure 12: The oscilloscope display shows a solid-state hi-fi amplifier. (Top: Solid-state hi-fi amplifier output, 60 V_{PP}, 10× probe. Middle: Low harmonic distortion oscillator output, 2.7 V_{PP}, 10× probe. Bottom: Notch filter output: 2 mV_{PP}.)

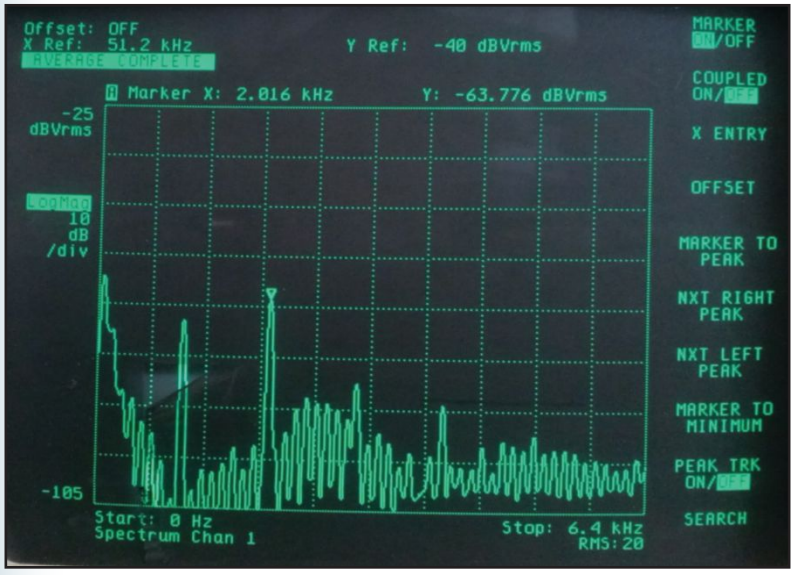


Figure 13: The spectrum analyzer display shows the hi-fi ampifier output with the fundamental notched out.

Spectrum Analyzer

Because of AC line noise and high gain amplifier noise, it gets difficult to see or measure harmonic distortion below about 1 mV on an oscilloscope without additional gain and filtering. And if you're looking at the waveform, it takes some practice to determine the amount each harmonic contributes if no single harmonic dominates the signal.

A spectrum analyzer displays amplitude on the vertical axis and frequency on the horizontal axis. Noise floor displays as a rough horizontal line and each frequency component is like a pole sticking out of the ground. Each harmonic is easily located and measured via text, indicating the cursor's position as long as they're high enough above the noise floor (at least 6 dB) to ensure the amplitude is unaffected by noise. You can use it in place of or in addition to your oscilloscope.

Figure 9 shows the spectrum analyzer display of the tube guitar amplifier's output. **Figure 10** shows the spectrum analyzer display of the solid-state guitar amplifier's output. Do the analyzer results corroborate the difference circuit results? One way to check is to refer to the chart shown in **Table 1**.

Even an old, used but working hardware audio Fast Fourier Transform (FFT) analyzer such as mine (the HP35660A) costs as much as a decent power amplifier. But, there are many software spectrum analyzers available. If you try one, I recommend that you compare and corroborate the difference circuit output waveform with the analyzer result. I tried a few in the past but did not verify their accuracy.

There's a caveat for measuring harmonic distortion using just an analyzer. Even the best analyzers generate their own harmonic distortion, typically -80 dB below the fundamental. So if you're troubleshooting harmonic distortion below about -70 dB, the analyzer distortion will mask your DUT measurement.

The dynamic range—the range between the DUT output level and about 3 dB above the noise floor of the oscillator/analyzer—is greatly increased by attenuating the fundamental with the notch filter, then sending this signal to the analyzer input. If the filter attenuates the amplitude of the fundamental by 60 dB, the analyzer's dynamic range is increased by 60 dB. If the analyzer alone has an 80 dB dynamic range, the notch filter and analyzer test system together have a 140 dB dynamic range—a huge improvement. Problem solved: caveat empty (see **Figure 11**).

Always monitor the notch filter output on an oscilloscope before connecting to the analyzer. Make sure you understand and set your analyzer input sensitivity. Perform the switch function shown in

Figure 11 with cables and reduce the amplitude of the fundamental by tuning the notch before connecting the cables in the other switch position. Both software and hardware analyzers use analog components to see your real-world signal so make sure you don't overload the analyzer input. You don't want to replace your PC sound card or radio frequency front end. Oscilloscopes are more forgiving.

Figure 12 shows the oscilloscope display of the hot setup measuring a 275 WPC solid state hi-fi amp output with an 8 Ω load. The notch filter output is mostly second harmonic distortion, but how much? It can't be accurately measured without the analyzer.

Figure 13 shows the analyzer display of the solid state hi-fi amp output with fundamental notched out. The marker displays second harmonic distortion, clearly above the noise, at -63.8 dBVRMS.

More Oscillator

What is the setup shown in **Figure 11** capable of measuring? **Figure 14** shows the oscilloscope display of an extremely low distortion oscillator output into the notch filter and the filter output. Where's the harmonic distortion? How much better can we measure with the analyzer (see **Figure 15**)?

The analyzer shows the notched fundamental and harmonics plainly above the noise floor:

- 1 kHz = -70 dBVRMS (notched), 2 kHz = -99.4 dBVRMS, 3 kHz = -103 dBVRMS—with 13 V_{PP} (+13.2 dBVRMS) sine wave output
- 2 kHz = -13.2 - 99.4 - 20 + 9.5 = -123 dB
- 3 kHz = -13.2 - 103 - 20 + 5.6 = -130 dB

The hot setup is a test system that can accurately measure individual harmonics below –120 dB (1 ppm). You can get an idea how steep the notch is on a spectrum display by tuning the filter while watching the frequency components. The fundamental's amplitude will bob up and down as you search for the deepest notch, but the amplitudes of the harmonic levels barely change at all.

In Summary

You can use a function generator and the circuits shown to accurately measure harmonic distortion from the painfully audible to 0.1% without big investment in time or money. With a better oscillator and a notch filter, you can measure somewhat lower. An excellent oscillator, a notch filter, and a spectrum analyzer enables you to measure to 1 ppm or less.

Author's Note: Thanks to the audioXpress *staff and Oliver Masciarotte for all their assistance in the preparation of this article.*

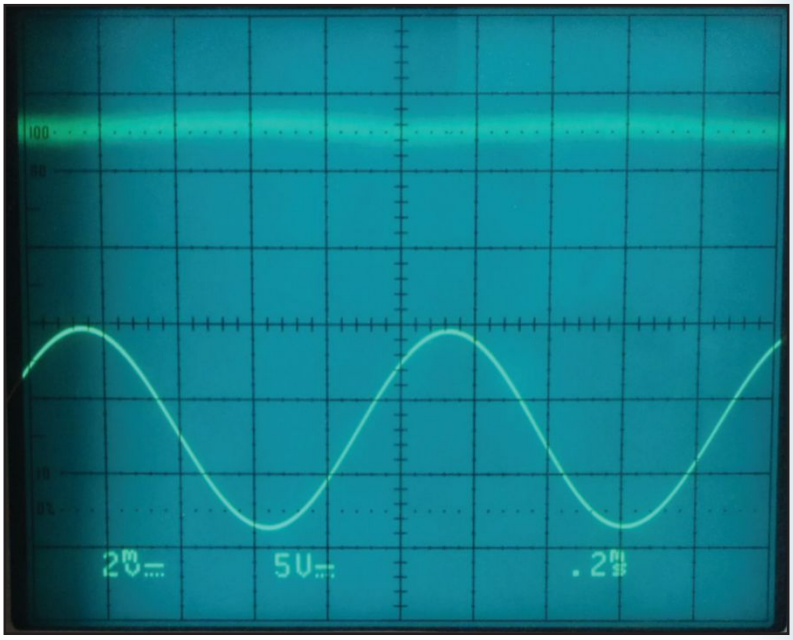


Figure 14: The oscilloscope display shows a low distortion oscillator. Top: Notch filter output, ~200 μV_{PP}. Bottom: Low harmonic distortion oscillator output, 13 V_{PP}.

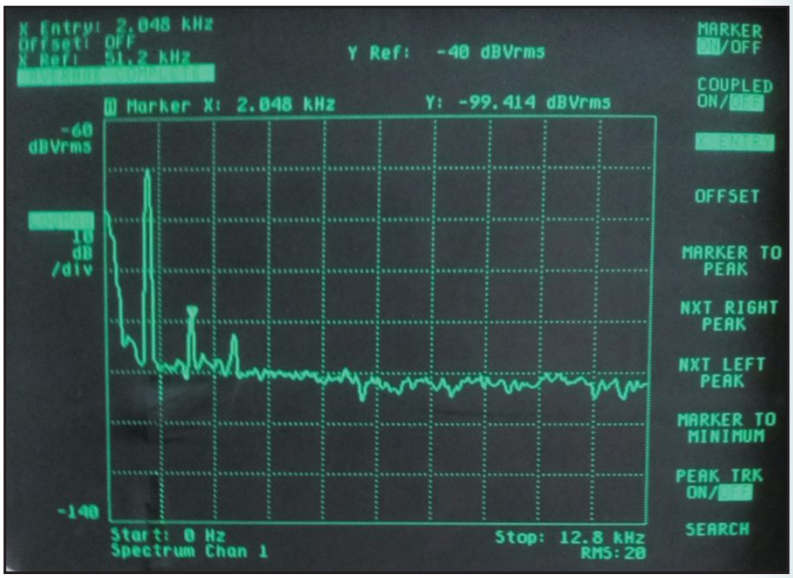


Figure 15: The analyzer display shows the notch filtered output of a low-distortion oscillator.

Resources

"Plastic Film Capacitor Data," Larry Davis, 1998, www.interfacebus.com/Capacitance_Temperature_Change.html.

Resistor Materials, Resistor Guide, www.resistorguide.com/materials.

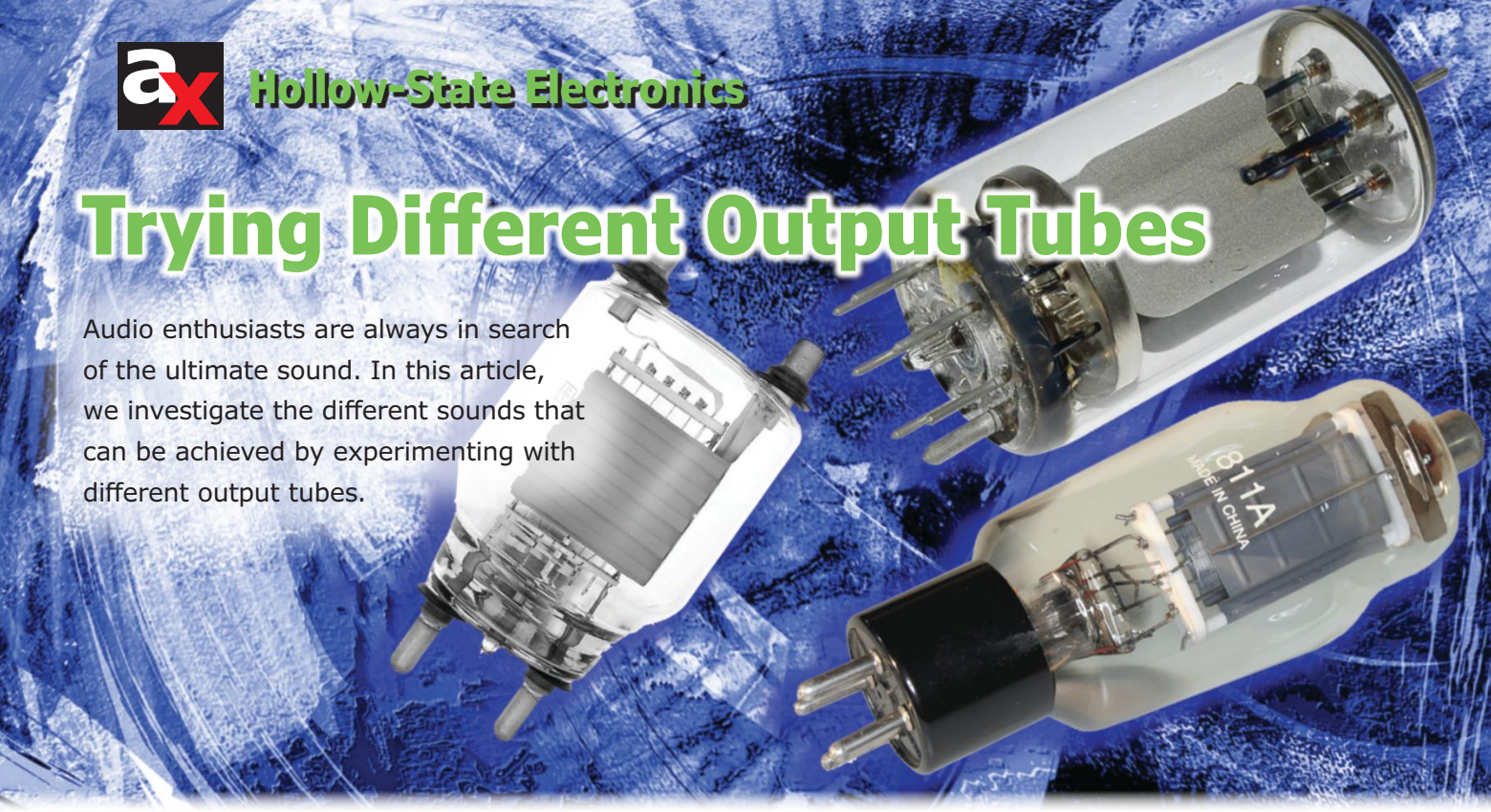
"Ultralow Noise Precision High Speed Op-Amps," LT1028 datasheet, Linear Technology, http://cds.linear.com/docs/en/datasheet/1028fd.pdf.

Source

Xicon 23PS310 polystyrene capacitors
Mouser | www.mouser.com/ds/2/449/23p-341148.pdf

Trying Different Output Tubes

Audio enthusiasts are always in search of the ultimate sound. In this article, we investigate the different sounds that can be achieved by experimenting with different output tubes.



By
Richard Honeycutt
(United States)

One of the most common "tweaks" made to audio power amplifiers is a change of output tubes. In some cases, this process is as simple as taking out the old tubes and plugging new ones into the same sockets. At the other end of the spectrum is the rebuild project that involves rewiring the output stage, and perhaps replacing the output transformer. The goal, in any case, is usually to see if the amplifier's sound can be improved. Often, this goal is elusive since differences among different types of output tubes—at least, ones that are interchangeable—may not have much effect on the sound. This fact has not prevented many enthusiasts from developing keen preferences for what they believe to be the "sound" of a particular model of output tube.

Determining the Differences

The only accurate way to determine a difference in sound between two models of tubes would be to use ABX testing. This would involve a test subject (listener) hearing music on known tubes A, then hearing music on known tubes B, followed by hearing music on unknown tubes X. If the subject can reliably identify X as either A or B, then we conclude that there is a perceptible difference between the samples. If the subject cannot reliably identify X as either A or B, a perceptible difference is pretty much ruled out.

In order for such testing to clearly show a difference between the sounds of two different tubes, the test must be "blind": The subject must not be able to find out except by listening whether X is A or B. This is a difficult test to perform for output tubes, because of people's very limited aural memory. Switching between two different speakers can be accomplished almost instantaneously, so in ABX speaker tests, the subject does not have to remember the sound of the first speaker for very long. But in the case of tubes, warm-up time necessitates a silent gap between listening to different tubes, making the task of judging them much more difficult and less accurate for the subject to accomplish.

Despite the challenges of accumulating objective evidence of differences between the sounds of different tubes, many hollow-state aficionados enjoy trying to do so. Since the choice of one "sound" over another is subjective, there is nothing to lose by unscientifically picking your "favorite" type of output tube, except the expense of buying a few tubes to try.

Swapping output tubes should only be done with a knowledge that certain properties of the original type and those of the new type are compatible. Random swapping predicated upon "Hey, it fits the base when I plug it in!" can result in significant damage, requiring major amplifier repairs. There are three critical specifications to be considered: base diagram (socket type and pinout), maximum plate voltage, and maximum plate dissipation.

Most base diagrams (or terminal diagrams) for

tubes are standardized among manufacturers. Each diagram is identified by an alphanumeric sequence. For example, many beam power tubes and power pentodes use an octal (8-pin) base, but the assignment of pins to the tube's internal elements varies from one terminal diagram to another. A 6L6 beam power tube uses base diagram 7AC, as does a 6550 (see **Figure 1**), but a 7027 power pentode uses an 8HY base diagram (see **Figure 2**). The differences are that the 7AC base has no connection to pin 1, and an absent pin 6; whereas the 8HY base has pins 1 and 4 connected internally to the screen grid, and pins 5 and 6 connected internally to the control grid. Often, manufacturers of amplifiers that have 6L6s use pin 1 or pin 6 on the socket as a spare solder lug or tie point for circuitry not directly connected to the output tube. In such cases, if you plug a 7027 into the socket, you get unintended connections.

Real Performance Differences

Real performance difference between different output tubes could result from the types having differences in transconductance (g_m) or different mu in power triodes. These are both related to gain, and could be audible in terms of sensitivity or output power level with given control settings, or in terms of distortion: Higher gain predisposes an amplifier to having a lower clip point. Non-audible, but still real performance differences include maximum power dissipation and maximum plate voltage and current, which can affect tube life.

In the 1970s, at least one retailer billed the 5881 beam power tubes as "industrial 6L6s." However, although both the 5881 and the 6L6GC are beam power tubes using the 7AC base, the 5881 is 2 29/32" high, not including the pins. The 6L6GC is 4 1/16" high, not including the pins (see **Photo 1**). The reason for the difference in physical size is a difference in electrical capability: The 6L6GC is rated at 450 V on the plate (500 V for the ultra-linear connection) and 30 W maximum plate dissipation; whereas, the 5881's maximum ratings are 400 V and 23 W, respectively. Unwitting amplifier owners who swapped their 5881s for 6L6GCs may have heard a difference in sound, but they paid for it in terms of shorter tube life.

Output Tube Types

There are three types of output tubes used for audio amplifiers: power triodes, power pentodes, and beam power tubes, which are essentially power tetrodes with beam-forming plates instead of a suppressor grid. Power pentodes and beam power tubes can usually be connected as triodes, but the

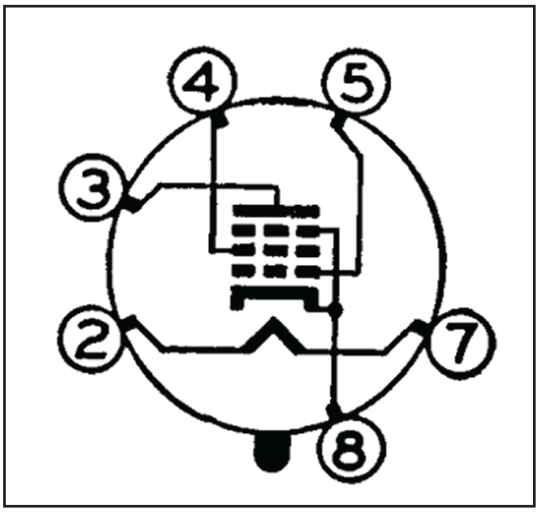


Figure 1: The 6L6GC tube uses a 7AC base.

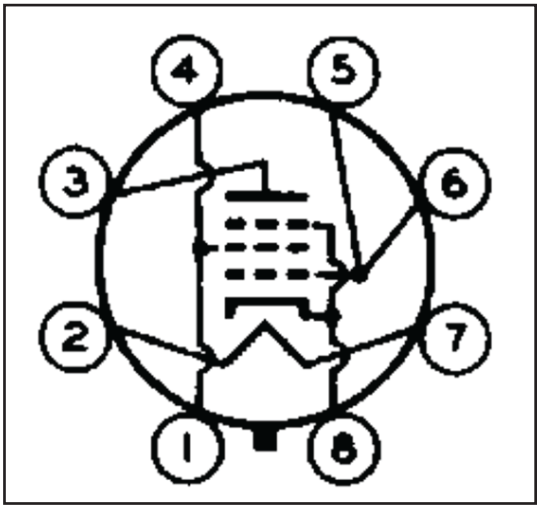


Figure 2: The 7027A tube uses an 8HY base.



Photo 1: The 5881 and the 6L6GC output tubes differ in size.

Triode Types	Base
833A	2-pin with grid and plate caps (see **Photo 2a**)
211	4-pin
2A3	4-pin
AA32-B-S	4-pin
AA52-B-S	4-pin
AA62	4-pin
G8WRB	4-pin
811, 811a	4-pin with plate cap (see **Photo 2b**)
RGN2004	European 4-pin
845	jumbo 4-pin
6C33	Septar 7 (see **Photo 2c**)

Table 1: Many power triodes use 4-pin bases, although some use less common styles.

Pentode Types	Base
6EH5	7CV
6F6	7S
6K6	7S
6CA7	8ET
7591	8KQ
7189	9BL
6BQ5	9CV
6GK6	9GK
7868	9RW

Table 2: There are almost no models of power pentodes that share the same base style with other tube models.

Beam Power Types	Base
5881	7AC
6550	7AC
7581	7AC
6L6 family	7AC
6V6	7AC
6W6	7AC
6Y6	7AC
6AQ5	7BZ
6DS5	7BZ
6146	7CK with plate cap
6AS5	7CV
6CU5	7CV
12AC5	8GW
7027	8HY
7355	8KN
6973	9EU
12AB5	9EU
6GC5	9EU

Table 3: One limitation on swapping beam power tubes is base style.

surrounding circuitry is different than would be used for the same tube in a pentode connection. Thus, changing among triodes, beam power tubes, and pentodes is not usually a matter of unplugging one tube and plugging in another. Some redesign may be required.

Table 1 lists power triodes that can be used for audio, sorted by base style. The 6C33 is rather unique, being a twin triode with a common cathode. They were designed as voltage regulators for the Russian military, but now find some use as audio power amplifiers (see Resources). The G8WRB was introduced by Svetlana, and is a direct replacement for the 811 (see Resources).

A table listing single triodes suitable for audio applications can be found on the Angelfire website (see Resources). The AA32, the 52, and the 62 tubes are unique products whose data can be found on the Ayon Audio website (see Resources). Always remember that having the same base is no guarantee that one tube is a direct replacement for another. **Table 2** lists audio power pentodes, sorted by base style.

Table 3 lists audio beam power tubes, sorted by base style. Many beam power tubes and power pentodes have European-type numbers as well. For instance, the beam power tubes were first called "kinkless tetrodes" in Europe, and had numbers such as KT66 (6L6GC), KT77 (6CA7/EL34), and KT88 (6550). Electro-Harmonix builds a KT90, which is a plug-in upgrade for a KT88. There was also a KT100 tube made in China a few years ago. The EL34 (6CA7),

Resources

Angelfire, Lycos, Inc., www.angelfire.com/electronic/funwithtubes/single_triodes.html.

"Ayon Super Triode Power Tubes," Ayon Audio, http://www.ayonaudio.com/ayon/ayon-super-triode-power-tubes.html.

M. Boele, "Datasheet 6C33C-B 6S33S-V," Rev 1.1, 2005, www.jogis-roehrenbude.de/Russian/6C33C/6C33C-B-6S33S-VExtendedDatasheetMB.pdf.

D. Grimwood, "Electron's Triode Operation of Audio and RF Modulator Power Tubes," 2008, www.oestex.com/tubes/triode.htm.

F. Langford-Smith, *Radiotron Designers' Handbook*, RCA Victor Division, Radio Corporation of America; 4th edition, 1953.

Svetalana Technical Devices, "Svetlana Technical Data 572B High-Mu Power Triode," www.g8wrb.org/data/Svetlana/pdf/572B.pdf.

although originally manufactured by Mullard as a pentode, was also produced by other manufacturers as a beam tetrode instead.

Distortion Signatures

Many audio enthusiasts and musicians prefer the distortion signature of triodes to that of pentodes or beam power tubes. However, most power triodes have directly heated filaments, making them more susceptible to hum. High-power triodes are often more expensive and harder to procure than beam power tubes or power pentodes. Therefore, it is common to use beam power tubes or pentodes in the so-called "triode connection." In this connection, the screen grid is connected directly to the plate. The suppressor grid of a pentode or the beam-forming plates of a beam power tube can be connected to the cathode.

With the tube properly connected, it can be used in a circuit just as a triode would. The maximum ratings should, of course, not be exceeded. Even though some tubes have higher maximum ratings for the triode connection, it is safer from both the thermal and the parasitic-oscillation standpoints to keep the operation under the maximum for pentode or beam power use.

Further information on triode-connected pentodes and beam power tubes can be found in the venerable *Radiotron Designers' Handbook*, or on this website: www.oestex.com/tubes/triode.htm.

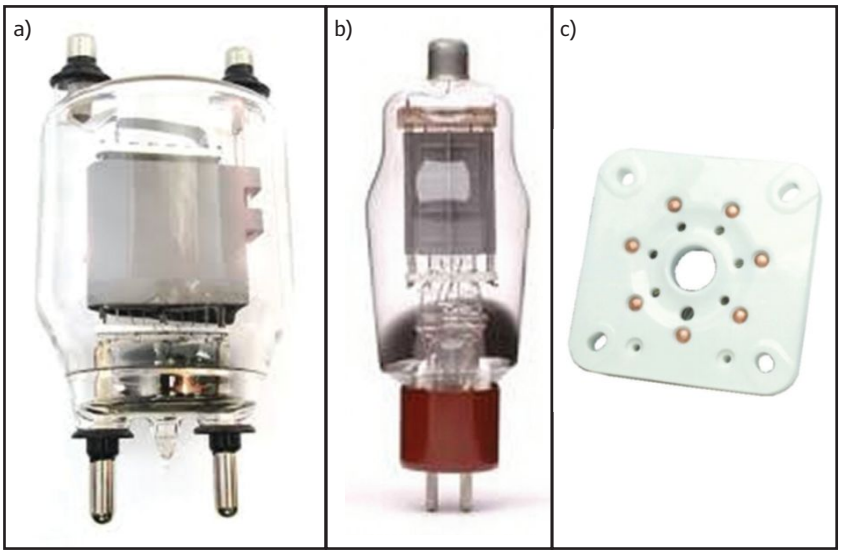
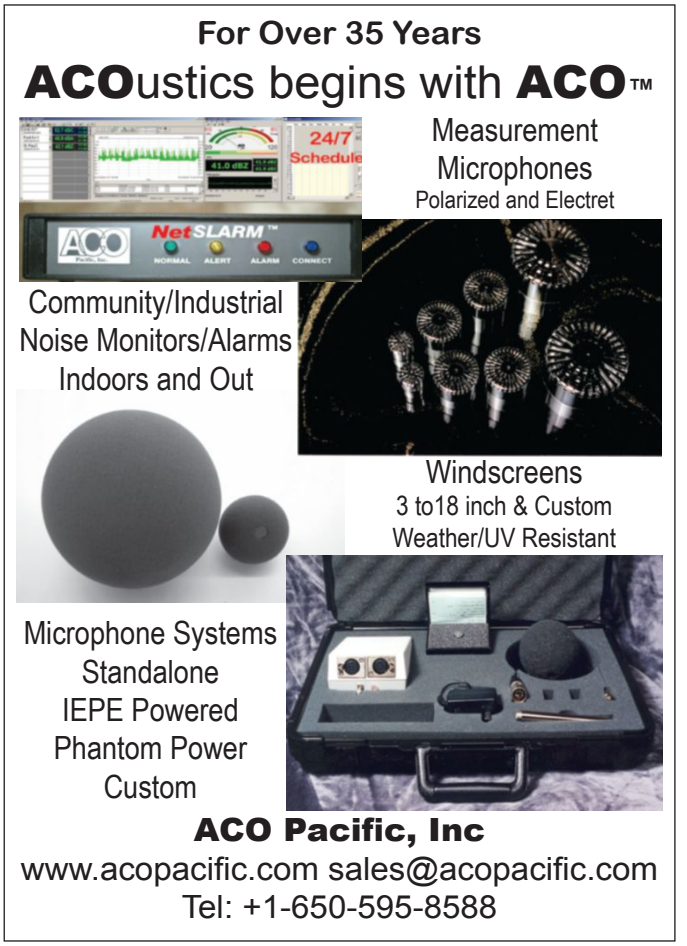


Photo 2: The bases of different tubes greatly vary. The 833A tube uses a 2-pin base with grid and plate caps (a). The 811a tube uses a 4-pin base plus a plate cap (b). The Septar 7 base (c) fits the Russian-made 6C33, which was used in the power supply of the MIG 25's main radio.





Industry Calendar

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

March 14–17, 2017
electronica China 16th International Trade Fair for Electronic Components, Systems, and Applications

China Import & Export Fair Complex, Guangzhou, China
Shanghai New International Expo Centre, China
http://electronica-china.com

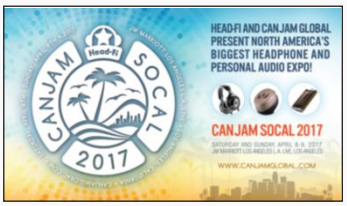
electronica China 2016 continues to be the premier showcase of electronics innovation in Asia, offering a multi-faceted experience to the visitors. It gets bigger and better every year. Founded in 2002, electronica China is now one of the leading platforms for electronic components, systems and applications.

electronica China takes place in conjunction with productronica China (dispensing, EMS, materials, SMT, test and measurement, automation, and component manufacturing), Photonics China and SEMICON at the Shanghai New International Expo Centre. The combination of these shows and market segments creates attractive synergies and added value for visitors, covering the electronics industry's entire value chain.

electronica China, is part of the electronica network —including the parent trade fair electronica Munich, held at Messe München every two years in November, and electronica India, which has been held in India every year in September since 2000. In 2016, electronica China combined 62,000 m² of exhibition space, attracting 61,455 trade visitors and 1,088 exhibitors from 23 countries.

April 8–9, 2017
CanJam SoCal 2017

The JW Marriott Los Angeles, L.A. LIVE, Los Angeles, CA
www.head-fi.org
www.canjamglobal.com/socal2017



CanJam is a network of headphone and personal audio shows, produced by Head-Fi.org, the world's largest community for headphone and personal audio enthusiasts. Since 2006, CanJam SoCal has been a well-established event for one of the fastest growing segments of the audio industry, and is dedicated to everything that goes on and around our ears. CanJam's not only about headphones and in-ears, but also increasingly about all the audio players, preamps, and electronics that complete the experience.

CanJam is expanding worldwide (Singapore, London, etc.), but the SoCal edition is a must-attend, this year moving from Orange County to the heart of Los Angeles. Attendees are encouraged to bring their own music to get the best possible experience. And with products in all price ranges, educational seminars, and prize giveaways, there is something for everyone at CanJam.

April 4–7, 2017
Prolight+Sound Frankfurt



April 5–8, 2017
Musikmesse

Exhibition Centre, Frankfurt, Germany
www.musikmesse.com
www.prolight-sound.com
www.messefrankfurt.com



Prolight+Sound 2017 and Musikmesse 2017 will enable Messe Frankfurt to fine tune the model going forward for these two important shows.

While Prolight+Sound 2016 was clearly a successful event for pro audio and lighting companies exhibiting in the modern Hall 3, everything else about this traditional Frankfurt fair needed some work—from the layout of the remaining Prolight+Sound halls to the disconnect with Musikmesse located in the extreme end of the gigantic Frankfurt Messe complex to the absence of key brands and the significant drop in visitors, trade press, and traffic between the two shows.

For 2017, Messe Frankfurt has implemented significant changes, starting with a new organizational structure for optimum linkage of the two shows. Michael Biwer, previously responsible for Prolight+Sound, is now responsible for both shows, which will return to a more coordinated model in 2017. Following those conceptual developments, Frankfurt Messe hopes to, once again, attract the ideal environment for product presentations, business, and recruiting.

Basically, Prolight+Sound will maintain and continue to develop the new components for the next edition in 2017, maintaining opening days from Tuesday to Friday, even bigger presentations, and an extensive outdoor exhibition area, now centrally located on the Agora, in the center plaza. However, the essential traffic flow from Musikmesse to Hall 4.1 of Prolight+Sound will continue to be the biggest challenge. For that reason, Musikmesse will be open from Wednesday to Saturday (from 10:00 AM to 6:00 PM) to increase the synergistic effects between the two events, held concurrently on three days.

Altogether, the two trade fairs attracted more than 100,000 visitors from 130 countries and around 2,000 exhibitors from 60 countries in 2016. Still, essential problems remain, such as the closed B2B Hall 11.1 area and the excessive dispersion of halls in the gigantic Messe complex, with an artificially imposed "instrument category" segmentation. However, some key brands such as Ibanez and Tama have decided to return to the show, which is a good sign.

April 21–23, 2017
AXPONA



North America's Premier Event for Audiophiles and Music Lovers
Westin O'Hare, Chicago, IL
www.axpona.com

Now in its eighth edition, Audio Expo North America (AXPONA) is successfully promoted by JD Events and is growing exponentially, becoming one of the best high-end consumer audio shows in the US. This three-day event features multiple hotel floors packed with listening rooms, exhibit floors, seminars, and live musical performances. The Ear Gear Expo is entirely dedicated to headphones, while the Marketplace allows visitors to discover components, accessories, vinyl, and more.

The 2016 edition welcomed 5,934 attendees and 400 exhibitors. Attendance was up 16% over 2015 with 137 students at the show as part of the new Student Day Program.

In addition to a series of special events, live concerts, and educational seminars, attendees had the opportunity to experience the very best in sound in 111 listening rooms by top manufacturers in the industry. In all, more than 400 leading manufacturers, retailers, and brands gathered at AXPONA. AXPONA 2016 also unveiled The Manufacturers Showcase, which featured a museum-like display of the newest products and innovations from 18 manufacturers.

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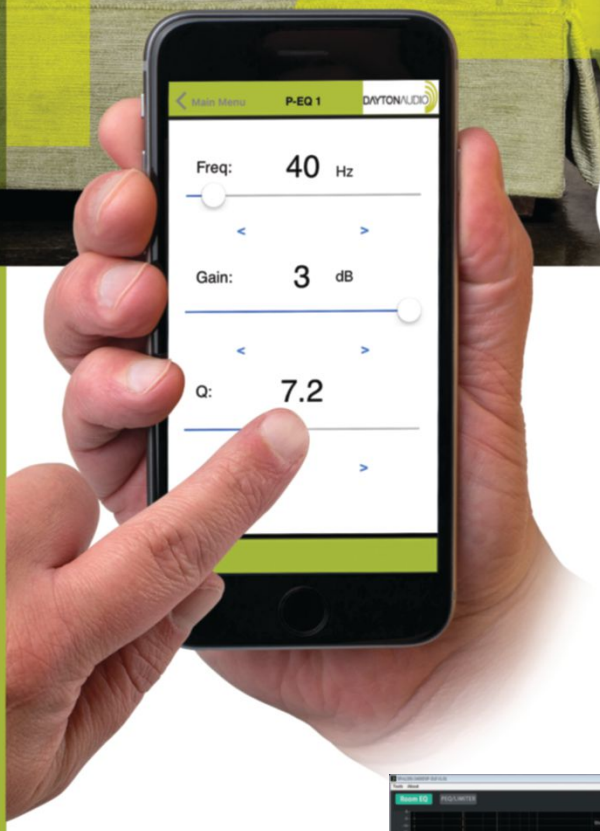
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