INNOVATIONS IN AUDIO • AUDIO ELECTRONICS • THE BEST IN DIY AUDIO

www.audioxpress.com

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Test and Measurement Focus

Market Update
The Latest Audio Test and Measurement Gear
By J. Martins

Practical T&M
Measuring Speech Intelligibility in Audio Systems
By Joe Begin (Audio Precision)

Practical T&M
Attenuators for Measurement
By Jonathan Novick

Fresh From the Bench
miniDSP EARS Affordable Headphone Test Fixture
By Stuart Yaniger

Speaker Builder
Loudspeaker Efficiency The Truth Behind the Numbers
By Geoff Hill

Audio Electronics
Discrete High-Output Current Buffer with JFET Input
By Dimitri Danyuk

Hollow-State Electronics
Noise in Vacuum Tube Circuits
By Richard Honeycutt

MARCH 2019

PARTS CONNEXION

The Authority on Hi-Fi DIY

Enter code: **AX0817**
in the comments box at
checkout to receive
25% Off
your next order.

Your #1 Source for
Vacuum Tubes, DIY Parts,
Audiophile Accessories
&
Premium HIGH-END Audio Gear
(Discontinued • Demo • B-Stock)

CAMBRIDGE AUDIO • MONITOR AUDIO •
CARDAS • REGA • FOCAL • WHARFEDALE •
MUSIC HALL • CYRUS • DEVIALET •
ACOUSTIC ZEN • QED • BDI • SOLID TECH •
APOLLO • TARGET • GRADO

www.partsconnexion.com

Sales: order@partsconnexion.com
Inquires: info@partsconnexion.com

Toll-Free - US/Canada
905-631-5777 **1-866-681-9602** 905-681-9602

5403 Harvester Rd, Unit 1, Burlington, Ontario, Canada L7L 5J7

Debit, Visa, Mastercard, Amex, PayPal, EMT, Money Order & Bank Draft

HAMMOND MANUFACTURING®

MANUFACTURING & DESIGNING
TRANSFORMERS & ENCLOSURES
SINCE 1917

USA
Tel: (716) 630-7030
Fax: (716) 630-7042

Canada
Tel: (519) 822-2960
Fax: (519) 822-0715

UK
Tel: 01256 812812
Fax: 01256 332249

Australia
Tel: 8-8240-2244
Fax: 8-8240-2255

www.hammondmfg.com

March 2019 **ISSN 1548-0628**

www.audioxpress.com

audioXpress (US ISSN 1548-0628) is published monthly, at $50 per year for the US, at $65 per year for Canada, and at $75 per year Foreign/ROW, by KCK Media Corp., P.O. Box 417, Chase City, VA 23924, US Periodical Postage paid at Chase City, VA, and additional offices.

Head Office:
KCK Media Corp.
P.O. Box 417
Chase City, VA 23924, US
Phone: 434-533-0246
Fax: 888-980-1303

Subscription Management:
audioXpress
P.O. Box 417
Chase City, VA 23924, US
Phone: 434-533-0246
E-mail: custserv@audioxpress.com
Internet: www.audioxpress.com

Postmaster: Send address changes to:
audioXpress
P.O. Box 417
Chase City, VA 23924, US

Advertising:
Strategic Media Marketing, Inc.
2 Main Street
Gloucester, MA 01930, US
Phone: 978-281-7708
Fax: 978-281-7706
E-mail: audioxpress@smmarketing.us

Advertising rates and terms available on request.

Editorial Inquiries:
Send editorial correspondence and manuscripts to:
audioXpress, Editorial Department
P.O. Box 417
Chase City, VA 23924, US
E-mail: editor@audioxpress.com

Legal Notice:
Each design published in audioXpress is the intellectual property of its author and is offered to readers for their personal use only.
Any commercial use of such ideas or designs without prior written permission is an infringement of the copyright protection of the work of each author.

© KCK Media Corp. 2019
Printed in the US

Measurements to the Masses?

Welcome to our March edition dedicated to audio test and measurement. In this edition, we are featuring a variety of audio signal analyzers; solutions that address acoustical measurements and room compensation; solutions that can help brands characterize and tune hybrid noise cancelling headphones; solutions to quality test the acoustic responses of the microphones and speakers in smart speakers and connected devices; and to test loudspeakers, headphones, and audio systems in vehicles. We even discuss a computer simulation application of a head-and-torso mannikin used for evaluating the performance of in-ear and on-ear headphones!

And as we discuss test and measurement solutions, I am glad that we are able to feature a review by Stuart Yaniger about the new Earphone Audio Response System (EARS), which miniDSP introduced as an affordable test jig for headphone measurements. *audioXpress* was among the first publications to announce online the introduction of this miniDSP solution, basically a plug & play USB audio streaming device, featuring dual low noise calibrated mics with interchangeable silicon ears. We always closely follow what this Hong Kong company is doing and admire its cutting-edge work at the intersection of the latest audio industry trends and the interest of the audio communities, from DIYers to commercial integrators and developers—much like *audioXpress*.

When miniDSP announced EARS for $199, we were not surprised to see the immediate reaction of fear around the industry. With the growing debate around home-made measurements, DIY-calibration and personalization of headphones, it was clear that miniDSP had identified a market need. Of course, many immediately denounced the fact that EARS could not be compared to professional lab-grade headphone measurement solutions incorporating ear simulators and standardized simulated pinnae that typically cost more than $10,000, fearing that this would just encourage more online activists to start publishing unreliable homemade measurements to substantiate their opinions about products. Of course, the fact that miniDSP lists the EARS system under the "Measurement Gear" category just reinforced those fears.

The truth is that, in just a few months, we have witnessed an upswing in activity from headphone enthusiasts using EARS to compare and interactively equalize headphones, some targeting their own subjective response curves, others simply trying to emulate existing industry references, including the Harman "preferred headphone target response." In general, those enthusiast communities praised the miniDSP device's possibilities "regardless of its absolute accuracy," as a way to compare headphone measurements in an apples-to-apples manner, instead of using "improvised" personal measurement rigs.

Again, this is a formidable challenge for the audio industry. As if homemade measurements of questionable origin and unknown methodology spread across the Internet weren't bad enough, now more people are being encouraged to use an affordable "measurement" solution that doesn't conform to any established standards. That's why we are proud to feature our own review of the EARS system and have Yaniger carefully examine what can be done with this apparatus, what the trade-offs are, and conduct some actual measurements with it—and with proper lab-grade test gear, including Audio Precision's AECM206 headphone test fixture, which was also reviewed by Yaniger in *audioXpress* April 2018. In fact, anticipating this review, Yaniger had already experimented with the EARS systems when he reviewed the 1MORE Triple Driver headphones in our February 2019 edition.

Why are we doing it? Because, like or not, the miniDSP EARS system is now available globally and people are using it—establishing a de-facto standard with all its inconsistencies. So, better provide some actual data on the actual virtues and defects of such a system and explain how and when to use it. Our much esteemed and highly missed late author, Ron Tipton, effectively started working on the EARS review before falling fatally ill. Tipton was experimenting with a pair of Sennheiser headphones using its DIY HRTF Head (described in *audioXpress* April 2018) and the miniDSP EARS system, using the free Room EQ Wizard (REW) software to effectively analyze what he could do with the Sonarworks software for headphones that corrects their frequency response. He was quick to note the inconsistencies in the results he was getting but was notably excited with the possibilities. And this is something that is common with many of our readers, including the many high-level product designers currently working on headphone measurement and calibration solutions.

J. Martins
Editor-in-Chief

The Team

President:	KC Prescott	Associate Editor:	Shannon Becker
Controller:	Chuck Fellows	Graphics:	Grace Chen
Editor-in-Chief:	João Martins	Advertising Coordinator:	Nathaniel Black

Technical Editor: Jan Didden

Regular Contributors: Bruce Brown, Bill Christie, Joseph D'Appolito, Vance Dickason, Jan Didden, Scott Dorsey, Gary Galo, Gerhard Haas, Chuck Hansen, Richard Honeycutt, Charlie Hughes, Mike Klasco, Ward Maas, Oliver Masciarotte, Nelson Pass, Christopher Paul, Bill Reeve, Fernando Rodrigues, Steve Tatarunis, Stuart Yaniger

OUR NETWORK

audioxpress · circuit cellar · LIS Loudspeaker Industry Sourcebook · Voice Coil

SUPPORTING COMPANIES

ACO Pacific, Inc.	58	Menlo Scientific, Ltd.	29
All Electronics Corp.	66	miniDSP, Ltd.	45
Audio Precision, Inc.	21	NTi Audio, Inc.	35
Avel Lindberg, Inc.	47	Parts ConneXion	2
AXPONA 2019	25	Parts Express International, Inc.	68
CRY Sound Electronics Co., Ltd.	37	PCB Piezotronics, Inc.	15
Hammond Manufacturing, Ltd.	3	Redco Audio	55
HEAD acoustics, Inc.	33	RØDE Microphones	11
Hypex Electronics BV	63	Scantek, Inc.	65
iSEMcon, LLC	45	Studio Six Digital	47
KAB Electro-Acoustics	55	Triad Magnetics	67
KLIPPEL GmbH	57	Virtins Technology	41
Linear Integrated Systems	59	Warkwyn Associates	66

NOT A SUPPORTING COMPANY YET?

Contact Peter Wostrel (audioxpress@smmarketing.us, Phone: 978-281-7708, Fax: 978-281-7706) to reserve your own ad space in the next edition of the magazine.

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.

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.

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.

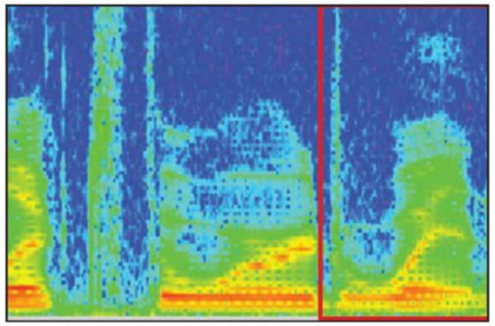
Steve Tatarunis has been active in the loudspeaker industry since the late 1970s. His areas of interest include product development and test engineering. He is currently a support engineer at Listen, in Boston, MA, where he provides front-line technical support to the SoundCheck test system's global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

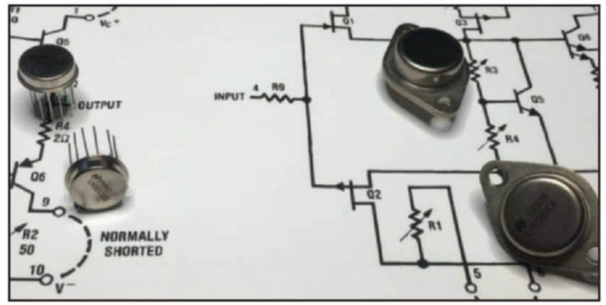
Contents

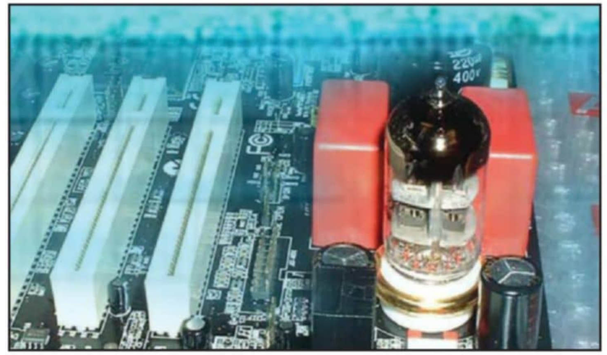
Volume 50 – No. 3

March 2019

audioxpress







Edition Focus—Test & Measurement

MARKET UPDATE

8 Exploring the Frontiers from R&D to QC

By J. Martins

In this Market Update, *audioXpress* highlights some of the latest solutions now available to the audio industry, including updates to some of the most established toolkits as well as new solutions.

30 Examining the miniDSP EARS Headphone Test Fixture

By Stuart Yaniger

Learn more about the capabilities of "EARS," the Earphone Audio Response System, which miniDSP says brings headphone measurement capability to "the masses."

38 Measuring Speech Intelligibility in Audio Systems

By Joe Begin

Joe Begin, of Audio Precision, examines the measurement methods currently in use for the objective—and repeatable—evaluation of speech intelligibility.

42 Attenuators for Measurement

By Jonathan Novick

In this article, Jonathan Novick discusses external attenuators and compares the impact different attenuators' thermal noise will have on measurements of distortion.

Features

48 Loudspeaker Efficiency

The Truth Behind the Numbers

By Geoff Hill

This article explores why we don't see more high-efficiency designs in common use and reminds us that it is the total system efficiency that matters.

52 Build a Discrete High Output Current Buffer with JFET Input

By Dimitri Danyuk

Discover ways to provide better isolation between input and output terminals and lower input noise current using complementary JFETs from Linear Integrated Systems.

Columns

HOLLOW-STATE ELECTRONICS

60 Noise in Vacuum Tube Circuits

By Richard Honeycutt

Departments

4 From the Editor's Desk

5 Client Index

66 Industry Calendar

Websites

www.audioxpress.com
www.voicecoilmagazine.com
www.cc-webshop.com
www.loudspeakerindustrysourcebook.com
www.linuxgizmos.com

@audioxp_editor audioxpresscommunity
linkedin.com/company/audioxpress

Market Update

Exploring the Frontiers from R&D to QC

For this special edition of *audioXpress*, which focuses on Test & Measurement, we compiled a Market Update highlighting some of the latest solutions now available to the audio industry, including updates to some of the most established toolkits as well as new solutions. All the selected solutions have been introduced in the last 12 months and address specific market requirements for all types of signal and acoustical measurements, research and development, repeatable processes for quality control, high-speed production tests, or troubleshooting at the test bench.

By
J. Martins
(Editor-in-Chief)

Listen, Inc.

New Possibilities with SoundCheck 16

At the end of 2017, Listen announced the release of SoundCheck 16, its flagship audio measurement software. With a consistent schedule of product updates since the original launch in 1995, this new release of SoundCheck includes a bundle of new features that expand its capabilities and simplify ease of use.

Following the introduction of version 14 in 2015, when Listen brought its powerful test and measurement platform (also for Mac OS X) as a native 64-bit application, the Boston-based company followed through with version 15, launched in 2016. Version 15 expanded its testing capabilities with additional external control, calibration, audio mixing, and hardware integration, and introduced the first in the company's redesigned virtual instruments, the multimeter.

With version 16, SoundCheck's functionality was further enhanced with several new optional modules including a Real-Time Distortion Analyzer, Frequency Counter, and Strip Chart Recorder. The Real-Time Distortion Analyzer provides continuous real-time measurement of output distortion, including total harmonic distortion (THD) and THD+N, THD and THD+N residual level, and signal-to-noise and distortion (SINAD). The Frequency Counter offers an accurate and clear visual indication of frequency, determining the dominant signal in a selected signal path and returning a precise frequency measurement. The New Strip Chart Recorder enables the multimeter, distortion analyzer and frequency counter to plot measurements over time—a modern day equivalent of connecting a paper chart recorder to a classic, stand-alone hardware instrument. The multimeter, distortion analyzer, and frequency counter can all now save results to the memory list, enabling an instantaneous measurement to be recorded without having to run a sequence, and enabling saved values to be used within sequences.

Listen's completely redesigned THD+N analysis algorithm is compliant with the AES17 standard and is more accurate and significantly faster. High accuracy is achieved, even with short and/or low-level test signals, through the use of a synthetic notch filter—an extremely precise digital notch filter. A traditional notch filter option is also offered to enable correlation with measurements made on legacy systems. The algorithm also includes high- and low-pass filters for controlling the measurement bandwidth and for filtering noise in electronic measurements.

A new Sweep Equalization option offers smoother equalization to minimize the transient response in the device under test, resulting in shorter test times and smoother sounding sweeps; this is useful for microphone testing where a source speaker needs to be equalized. In addition, the multimeter has been upgraded, the setup simplified, and several additional control features added.

The upgraded multimeter has a new, fixed or auto-tracking bandpass filter option. This is useful for measuring a specific signal in the presence of background noise. In addition, a new "Linear Repeating" averaging mode is available. In this mode a linear average is made and then repeated. This is useful when, for example, we would like to plot a repeating 1 second Leq of the environmental noise level.

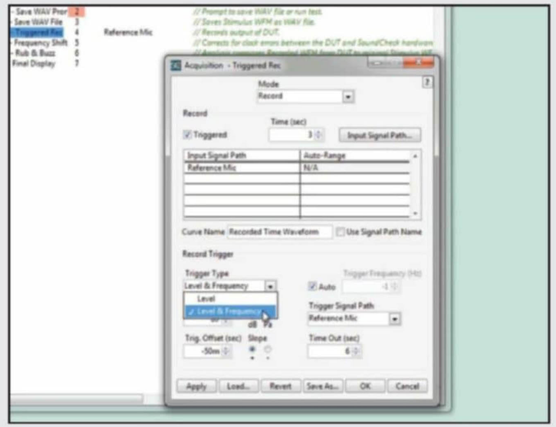
There are also improvements on the usability side including increased setup automation and control via Python. An included example (which can be used as a starting point and modified) demonstrates the simplicity of controlling SoundCheck using Python. Compatible with Python 2 and 3, this script opens SoundCheck, loads and runs the "Complete Test" sequence, and passes results back to the Python script for further processing.

SoundCheck 16 simplified ease of use and introduced many new measurement routines, including a completely redesigned THD+N analysis algorithm, which is compliant with the AES17 standard and is more accurate and significantly faster.

Simplified Setup

SoundCheck now automatically scans for new devices and setting changes when opened, and updates the Hardware Editor, Audio Hardware, and Listen Hardware Tabs. This makes setup much faster. The ability to also manually adjust hardware setup and preserve third party hardware settings offers flexibility and makes it easier to integrate Listen and third party hardware. To simplify audio interface setup, sample rates are restricted to the selection valid for that device. If changes are made during a SoundCheck session, a simple "refresh" will scan for changes and update the settings.

Software installation for multiple systems is also simplified. When pointing to a folder of multiple status.dat files in the setup wizard, the correct status.dat file is automatically selected based on the connected hardware key. A new conversion tool facilitates updating custom VIs from an older version to the latest version of SoundCheck by automatically re-linking the appropriate VI and CTI files, saving time over manually doing it.



With SoundCheck 16.1, Listen introduced a new frequency trigger that enables frequency-based triggering of acquisition from an external source, using a pilot tone at the beginning of a test sweep.

New Frequency Trigger for Open-Loop Testing

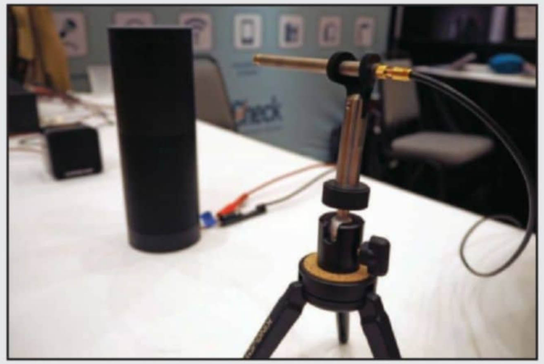
Already in October 2018, Listen announced the release of SoundCheck 16.1. This software release contains exciting new features for all developers and manufacturers working with connected audio devices, providing a test routine for smart speakers and other voice-controlled devices. The new software is available free of charge to all registered users of SoundCheck 16.

Most significant in the SoundCheck 16.1 release is the inclusion of a new frequency trigger. This allows frequency-based triggering of acquisition from an external source using a pilot tone at the beginning of a test sweep. This function offers improved accuracy over previous level-based triggering when capturing responses from a device where we don't have direct access to the microphone or speaker, especially in noisy environments. This technique optimizes open-loop test methods for smart speakers and other voice-controlled devices.

SoundCheck now also fully integrates with the Mentor A2B interface for testing automotive audio connected via the Analog Devices A2B digital bus. The Mentor Analyzer, which handles the transmission of signal into and out of the bus, is viewed as an ASIO interface by SoundCheck, enabling SoundCheck to read/write to the device and analyze any transducer connected to the A2B bus. A custom VI permits control of the Mentor A2B interface configuration via SoundCheck. This means that it can be controlled from within a SoundCheck sequence (e.g., loading configurations and starting/stopping ASIO streams). This makes it an ideal R&D or production line test solution for automotive audio, or for anyone testing transducers connected via A2B bus.

Also available (for an additional cost) with this new release are two optional sequences—background noise simulation to the ETSI ES 202 396-1 standard and the TIA-920B dual-bandwidth telephone test standard.

Market Update



Since the release of SoundCheck 16, Listen has been introducing test solutions to help developers and manufacturers working with smart speakers and other voice-controlled devices.

The background noise simulation sequence (part number 3121) is a cost-effective alternative to dedicated background noise generation systems. It calibrates a 4.1 speaker array to conform to the ETSI ES 202 396-1 Standard, providing an equalized, calibrated playback solution to test devices in a standardized and repeatable way. The sequence includes a library of real-world binaural recordings from the standard, and custom or user-defined binaural recordings may also be used. Applications of this sequence include evaluating ANC, noise suppression, voice recognition testing, SNR optimization of microphones, beamforming directionality studies of microphone arrays and more.

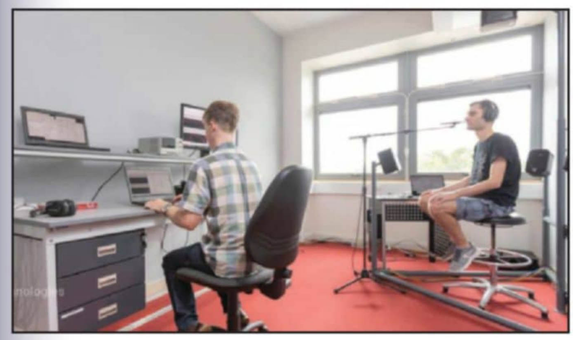
www.listeninc.com

SoundChip

Soundstation Array for Hybrid Noise-Cancelling Headphone Development

SoundChip recently introduced Soundstation Array, a spatially-limited diffuse sound field measurement platform designed to help brands characterize and tune hybrid noise-cancelling headphones. Soundstation Array leverages SoundChip-sponsored post-graduate research into the minimum requirements for generating a sound field that is sufficiently diffuse for ensuring repeatable and reproducible measurements of noise-cancelling headphones, without the need for an anechoic or reverberation chamber.

Rather than target strict conformity with International Standard definitions of diffusivity, Soundstation Array takes a more pragmatic approach that has been proven to provide an effective means for obtaining laboratory-grade results within a small office environment. Soundstation Array effectively assists acoustic engineers working on the development of challenging noise cancelling systems, enabling repeatable and reproducible measurements of feedback, feed-forward, and hybrid noise-cancelling headphones.



Soundstation Array has the capability to render a stand-alone, spatially-limited diffuse sound field for the precise measurement and characterization of hybrid noise-cancelling headphones.

Soundstation Array comprises a robust steel frame with adjustable speaker brackets, allowing four two-way loudspeakers to be located at the vertices of a tetrahedron. Coupled with a subwoofer, a multichannel sound card, an amplifier, and a software-based signal generator, Soundstation Array provides engineers with an acoustic field large enough to accommodate a standard Head-and-Torso-Simulator (HATS), or even a human test subject, and diffuse enough to return stable measurements up to 3 kHz.

The solution renders a spatially-limited diffuse sound field with broadband, statistically decorrelated noise at each source, and can be supplied with an optional SW-based equalizer to minimize the difference between adjacent third-octave bands (31.5 Hz to 8000 Hz).

Soundstation Array can also be operated within any type of environment, with no restriction on the proximity of locally reflecting surfaces, and enables performance comparisons to be made from measurements taken in different room environments. The frame comprises internal cable runs, ensuring neat installation and safe usage, and it can be easily assembled/dismantled using a single hex key. The height of the four loudspeaker brackets may be adjusted to suit different test fixtures.

www.soundchip.ch/platforms/soundstation-array

RØDE Test
FUZZMEASURE
The world's most advanced acoustic measuring software

POWERFUL AUDIO AND ACOUSTIC MEASUREMENTS
FuzzMeasure is an audio and acoustic measurement tool to produce, analyze, and publish beautiful graphs. It delivers a comprehensive suite of features for professionals in research, acoustics, live sound, room design, and pro audio. By combining sophisticated technology with an elegant user interface, FuzzMeasure offers unparalleled usability and flexibility.

rodetest.com
RØDE MICROPHONES

Market Update

NTi Audio

New Omnidirectional Sound Source and Amplifier for Insulation and Acoustical Measurements

Measuring the sound insulation and reverberation time in multi-roomed buildings can become pretty tiring. Carrying a heavy amplifier and a loudspeaker from room to room is also cumbersome. To better support the work of audio professionals and acousticians in their daily work, NTi Audio has developed a new lightweight, yet robust omnidirectional sound source, the DS3 Dodecahedron, together with the powerful PA3 Power Amplifier.

Despite its low weight of only 7.5 kg, the new DS3 Dodecahedron Loudspeaker consistently delivers up to 121 dB sound power level for extended periods of time, using a dozen 5" speakers. The high output is suitable for testing in rooms with lots of background noise, and the sturdy grids and housing are designed to take the knocks of everyday usage, even in hostile conditions (e.g., construction sites).

The new DS3 Dodecahedron Loudspeaker is delivered in a set with a speaker stand and the PA3 Power Amplifier, a carrying bag, and a Speakon cable—offering a complete signal source for room and building acoustic measurements. The dodecahedral design arranges 12 high-quality dynamic drivers for a uniform omnidirectional radiation of the acoustic test signal.

And, NTi Audio knows how a reliable remote control can ease the workload, so the new solution was thoroughly tested. The user can switch the signal ON only when necessary and OFF without repeatedly walking around. And there should be no concerns about the quality of the measurements, as this new Dodecahedron Speaker Set is compliant with

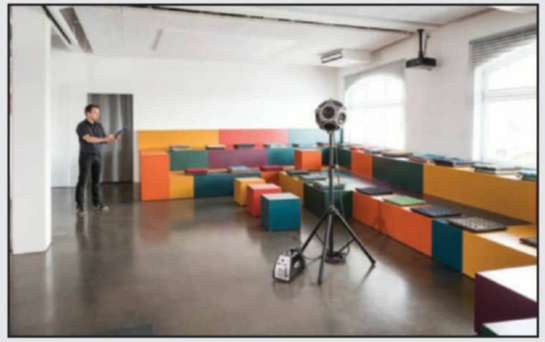
The Dodecahedron Speaker Set is an ideal choice for the experienced acoustic consultant, who is looking for a powerful, yet reliable and lightweight solution for professional measurements. Despite its low weight of only 7.5 kg, the new DS3 Dodecahedron Loudspeaker delivers up to 121 dB SPL for extended periods of time, and is driven by the new PA3 Power Amplifier, which has been optimized for high performance and ergonomic operation.

ISO16283, ISO140, and ISO3382 standards. Also, clients will appreciate the professional reports for Airborne, Impact, and Facade sound insulation measurements produced by the combination with the XL2 Acoustic Analyzer and the Sound Insulation Reporter PC-Software.

The DS3 is driven by the new PA3 Power Amplifier, which has been optimized for high performance and ergonomic operation. The illuminated buttons, together with the angled front and rear panel, provide a comfortable direct access to the control elements from a standing position. The built-in signal generator includes vital test signals (e.g., an equalized pink noise for an acoustically-flat frequency response from the DS3). The equalization ensures that the ISO 16283-1 requirement of ≤ 8 dB difference between contiguous one-third octave bands in the emitted sound is met.

Exel Building Acoustics Kit for Easy Sound Insulation Measurements

NTi Audio's latest Exel Building Acoustics Kit consists of the instruments and software required for professional airborne, impact, and facade sound insulation measurements. With the latest enhancements, Building Acoustics consultants can utilize this comprehensive package from NTi Audio for remote-controlled measurement. For simultaneous



The Exel Building Acoustics Kit consists of the omnidirectional loudspeaker and power amplifier, together with the XL2 Sound Insulation Reporter PC-based software application. Designed for acoustic consultants, this comprehensive tool uses data gathered by the XL2 Acoustic Analyzer, and returns graphical analysis of all measurement positions.

measurement in the source and receiving rooms, Wi-Fi communication takes over from the more traditional cable connection methods.

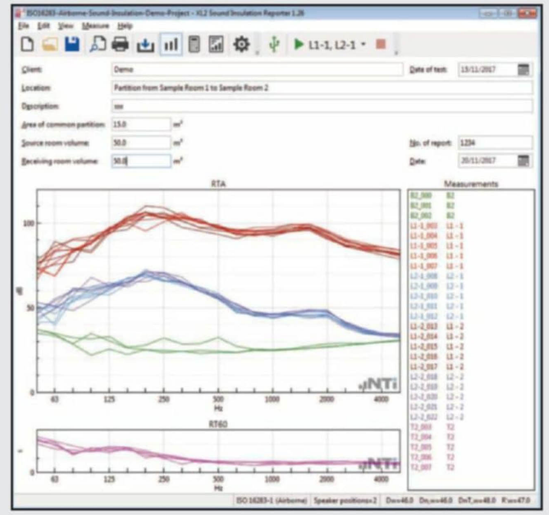
NTi's XL2 Acoustic Analyzer, in combination with the "Sound Insulation Reporter" PC-software, offers a professional and intuitive building acoustics measurement solution. The sound reduction index, a measure of the sound insulation provided by a partition between two rooms, is determined precisely and in accordance with the relevant ISO or American Society for Testing and Materials (ASTM) standards. In order to determine the sound reduction index, an excitation signal is generated (from an airborne or impact sound source) on one side of the partition and the reduction in sound across the partition is determined by measuring the sound levels with an XL2 in both the source and receiving rooms.

For consultants infrequently called on to carry out sound insulation measurements, one XL2 with its associated microphone would be a sufficient system configuration for the assessment. The process is to first measure the average level in the source room, then measure the average level in the receiving room. Finally, connecting the XL2 to a PC via the USB interface enables users to manually load the results into the Sound Insulation Reporter PC-software for immediate reporting.

Where there is an increasing requirement to undertake sound insulation tests, investing in a second XL2 and microphone may offer significant benefits for both site-based and laboratory-based measurements. The consultant can then simultaneously connect both XL2s to one PC via USB. Set up one microphone in the source room and one in the receiving room, and perform both measurements simultaneously with a single push of a button in the Sound Insulation Reporter software.

Having both source and receiving room results at the same time, gives an immediate indication of the sound insulation performance of the partition under test. If there are any notable deviations from the anticipated performance, the cause can be dealt with immediately and the measurement retaken. Performing measurements in a

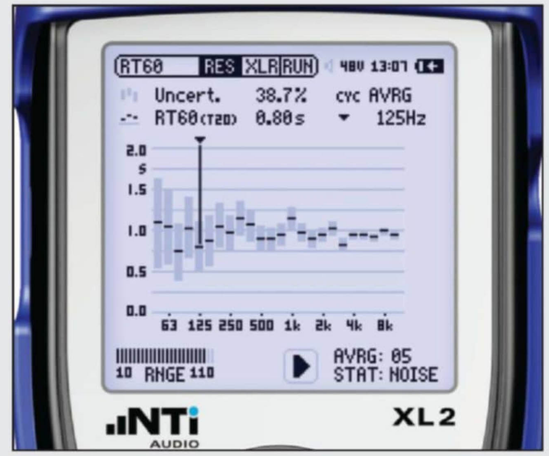
The XL2 Sound Level Meter quickly and easily takes one-third-octave level and reverberation time measurements required for the rating of insulation against airborne, impact, and façade sound transmission. And the XL2 Sound Level Meter and the new DS3 Dodecahedron Speaker can be controlled remotely.



Measurement view of the Sound Insulation Reporter Software

one-at-a-time sequence (e.g., with the Basic Configuration), means that such anomalies will only become apparent at the end of the process, which can be time consuming if there are many such anomalies that need attention. In addition to reducing measurement time, simultaneous measurements also increase accuracy by negating the effect of varying background levels.

Running the microphone cable between adjacent rooms may be too far for the cable length available, too difficult, or downright impossible. If, for example, there is no underfloor cable channel, then leaving the connecting door between the rooms even slightly open to run the cable can have a significant detrimental influence on the results. A Wi-Fi Access Point can solve this. The XL2 along with the associated microphone can be positioned in the remote room and connected via USB to a Wireless USB Device Server. The Sound



Here is an example of the reverberation time average results screen with the XL2 Sound Level Meter.

Market Update

Insulation Reporter software then simultaneously receives both wireless and USB signals from the two (or even more) XL2s. Eliminating cables in this way can often be essential for good measurement integrity, while also offering notable improvements in efficiency.

The latest version of the "Sound Insulation Reporter" software also makes it possible to determine the airborne sound insulation of facades or facade components. For example, the airborne sound insulation of windows can be quickly assessed and compared with alternative window types or installation methods. The assessment is carried out in accordance with ISO16283, ISO140, ASTM E966, GB/T 19889, or SIA 181 standards.

The "Sound Insulation Reporter" PC-software can be installed freely on any number of computers, so that measurements can be evaluated onsite on a notebook PC as well as post-processed on the office PC. For building acoustic assessments the "Extended Acoustic Package" and "Sound Insulation Reporter" options are required on the XL2. For users who only occasionally perform building acoustic measurements, an alternative to the permanent license for "Sound Insulation Reporter" is an inexpensive annual license. For the remote control of the XL2 and live display of the spectra on the PC, the "Remote Measurement" option is also required on the XL2. The minimum XL2 firmware version for automated sound insulation testing is V3.33 or higher. This is available to download for free to registered users.

www.nti-audio.com

NTi Audio

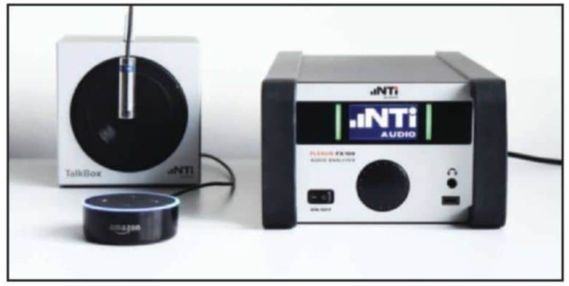
Smart Measurements for Smart Devices Using the Flexus FX100 Audio Analyzer

Responding to the challenges that were brought to the audio industry by the sweeping popularity of connected speakers and smart home devices, Swiss test and measurement specialist NTi Audio announced an open-loop test solution for its FX100 Audio Analyzer system.

Devices (e.g., smart speakers, mobile devices, personal assistants, smart thermostats, smart security cameras, and so forth) have all entered the consumer market in force during the last couple of years. More and more household appliances are also getting connected to the home networks. Most of those devices have an audio interface, such as built-in loudspeakers or microphones to interact with the user. Some of them even deliver an impressive level of audio quality. Manufacturers of smart devices need to stay ahead of the game and must run tests to ensure the quality of the acoustic responses of the microphones and speakers in those connected devices. A quality testing procedure is also something that is necessary for repair and refurbishment of such devices.

When testing loudspeakers and microphones as separated components, the stimulus signal and the measured signal going to and coming from the component are directly connected from the loudspeaker or the microphone to the Audio Analyzer system. This is called a "closed-loop" test setup. However, in smart devices, where the device has been completely assembled, there are usually no designated interfaces that enable access to the back of the components. Injecting a test signal into the loudspeaker under test, or measuring a test signal acquired by the device's microphone cannot be achieved in a closed loop.

The solution is to use an "open-loop" test setup. For the loudspeaker test, the test signal is not injected by the analyzer, but rather produced by the device under test (DUT) itself. For the microphone test, the results are not immediately received by the analyzer, but rather first recorded by the DUT itself. This open-loop solution is possible by using the NTi Audio "External GlideSweep" measurement technique. This allows the FX100 Audio Analyzer to synchronize to a played-back signal, perform the measurements, and visualize the data. Values measured include frequency response in decibels for sound pressure level (dBSPL); distortion response in percentage; sensitivity at 1 kHz in dBSPL; total harmonic



Setup example for smart device measurements. NTi Audio produced a tutorial using an Amazon Echo Dot device to demonstrate how to test the device's loudspeaker and microphone array using an open-loop configuration test sequence.

CLARITY.
FLEXIBILITY.
TRUE SOUND.

MODEL 426A14
2-IN-1 PHANTOM POWERED MICROPHONE PREAMPLIFIER

- 156 dB 1% THD and flat frequency response for high definition recording and modeling instruments
- Quickly change between ½" and ¼" 0V, IEC 61094-4 compliant microphones
- Fast rise times and transients for superior impulse responses

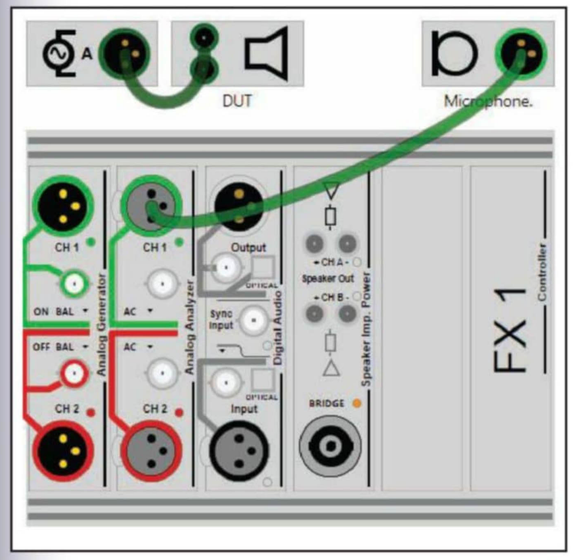
PCB PIEZOTRONICS
MTS SYSTEMS CORPORATION

1 800 828 8840 | pcb.com/phantom

MTS SENSORS

MTS Sensors, a division of MTS Systems Corporation (NASDAQ: MTSC), vastly expanded its range of products and solutions after MTS acquired PCB Piezotronics, Inc. in July, 2016. PCB Piezotronics, Inc. is a wholly owned subsidiary of MTS Systems Corp.; IMI Sensors and Larson Davis are divisions of PCB Piezotronics, Inc.; Accumetrics, Inc. and The Modal Shop, Inc. are subsidiaries of PCB Piezotronics, Inc.

Market Update



This is an open-loop test configuration with NTi Audio's FX100.

distortion (THD) at 1 kHz in percentage; and steepness (Rub & Buzz) response in Pascals per second (Pa/s). For the microphone, values measured in the test include frequency response in decibels as referenced in voltage (dBV); distortion response in percentage; sensitivity at 1 kHz in dBV; and THD at 1 kHz in percentage.

For the open-loop test sequence, NTi Audio uses its Flexus FX100 Audio Analyzer to play the voice commands and audio test signals, and analyze and report on the results; an M2010/M2015 Class 1 measurement microphone to pick up the test signal coming from the Smart Device loudspeaker; and the NTi Audio Talkbox calibrated loudspeaker to play the audio test signal when the Smart Device's microphone is under test and also play the voice commands to trigger the Smart Device to play the audio test signal when the Smart Device's loudspeaker is under test.

All the analysis and reporting are generated using the FX100 Control Software for PC. This software processes the measurement sequence, visualizes and logs the measurement results, and it can also be used to produce the test signals in WAV or MP3 format. The solution enables the creation of sequence modes for individual measurements, including tolerance handling and Pass/Fail results, creating a test workflow that is easy to understand for the operator.

www.nti-audio.com

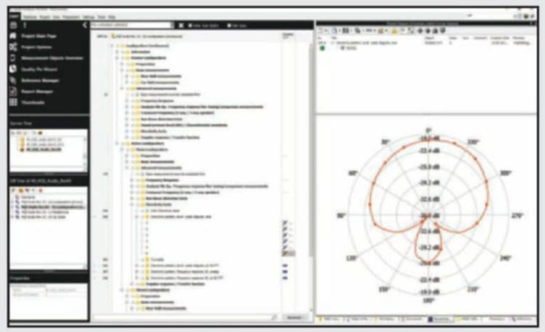
HEAD acoustics

HQS-Audio for Comprehensive Testing of Loudspeakers, Headphones, and Audio Systems in Vehicles

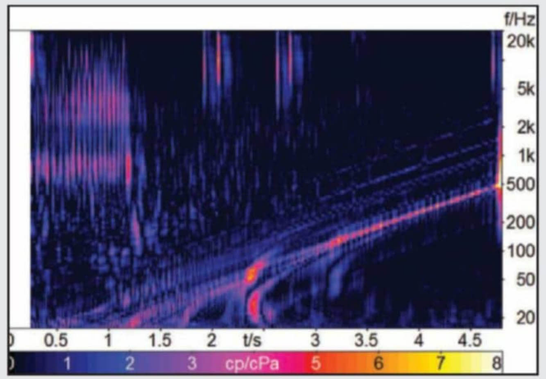
HEAD acoustics, one of the world's leading companies in the field of voice and audio quality optimization, has introduced HQS-Audio. HQS-Audio is a new tool for its Advanced Communication Quality Analysis (ACQUA) analysis system that provides a whole set of targeted electroacoustic measurements and helps to comprehensively test the quality of audio devices. Whether it be loudspeakers, headphones, or audio systems in vehicles, HQS-Audio provides independent test runs for each application, consisting of various electroacoustic measurements, including Thiele-Small (T-S) parameters.

HQS-Audio is an essential tool for manufacturers as well as for R&D divisions to test and optimize audio devices in a targeted manner. Besides the analysis of frequency response and signal-to-noise ratio (SNR), HQS-Audio offers extended tests of distortions including measurements of intermodulation distortion as well as total harmonic distortion (THD) including noise (THD+N). All are part of the database as well as Relative Approach analysis.

Based on the hearing model, the Relative Approach algorithm can be used to highlight pulse-like noises and is ideally suited as an additional measurement to detect Rub & Buzz effects caused by misalignment of a loudspeaker's voice coil. As an additional highlight, HQS-Audio offers special performance tests for headphones with passive and active noise filtering (noise cancelling/ANC). This type of test verifies the tested headphones' ability to isolate the listener from the surrounding background noise. This is done either by passive noise isolation or by using modern signal processing techniques (e.g., active noise suppression). In combination with the high-precision HRT I (HEAD acoustics' remote-operated turntable), users are able to measure directivity



The HQS-Audio database for the analysis system ACQUA is perfectly suited to test and optimize the quality of audio devices.



The Relative Approach analysis helps to detect the Rub & Buzz effects of loudspeakers.

characteristics of passive and active loudspeakers. Last but not least, HQS-Audio offers the calculation of T-S parameters.

HQS-Audio uses various test signals such as realistic speech signals or logarithmic sine sweeps. Since the database is perfectly suited for testing smart speakers and other active speakers, both mono and stereo reproduction is taken into account. All measurements in the database can be conducted fully automated. This enables manufacturers to analyze and optimize their audio devices quickly and comprehensively. Users are able to configure individual analyses intuitively and to adapt them according to the DUT. All measurement data is recorded and documented in ACQUA. The entire database of HQS-Audio is clearly structured and provides users with useful information on individual measurements.

HQS-AudioBasic for Testing Loudspeakers

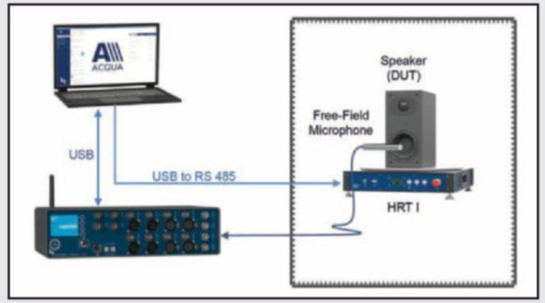
With HQS-AudioBasic, HEAD acoustics provides a product variant of HQS-Audio that solely focuses on electroacoustic tests of loudspeakers. HQS-AudioBasic is a subset of the HQS-Audio database, which contains all traditional measurements for testing and optimizing active and passive loudspeakers.

In combination with the HRT I turntable, users are able to test directivity characteristics of passive and active loudspeakers.

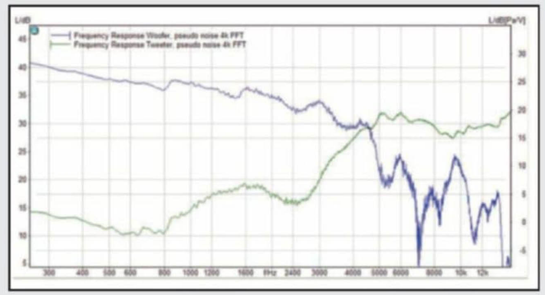
This includes measurements with a baffle and/or enclosure with a free-field microphone.

Used in combination with HEAD acoustics' modular hardware platform labCORE, which includes a multi-channel signal generator, a multi-channel analyzer front-end, and the HRT I—a high-precision turntable enabling users to perform automated, orientation-dependent acoustic measurements—the HQS-AudioBasic software enables basic measurements such as frequency response, SNR, SPL, THD+N, electrical polarity, and impedance.

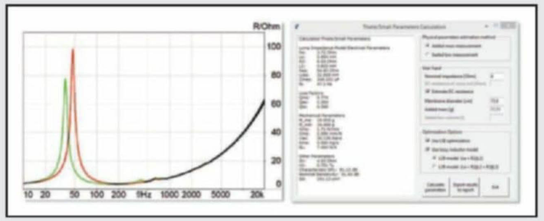
Advanced measurements include the visualization of crossover frequency for two- and three-way loudspeakers, directivity pattern, level-polar plot (sine/broadband noise), frequency response 3D (broadband noise), impulse response, transfer function, phase response, group delay, coherence, impulse response time windowing, cumulative spectral decay (CSD), two-tone intermodulation distortion, and distortion including Rub & Buzz, T-S parameters, and directivity. Traditional Rub & Buzz is measured starting from the 10th



Measurement setup for loudspeaker tests



Visualization of crossover frequency with HQS-AudioBasic



Examples of Thiele-Small parameters with HQS-AudioBasic

Market Update

harmonic, possible only for low frequencies, or with 3D relative approach—impulsive distortion—for measurements in audible frequencies (20 Hz~20 kHz).

T-S parameters are calculated on classical electrical impedance curve methods and with nonlinear optimization, including inductor model selection, and allow free-air, added mass, and sealed volume methods. T-S parameters require the installation of the coreOUT-A2 (code 7750) module on the labCORE analog front-end.

All measurements are implemented according to European Standards (EN) 60268-5 and EN 60268-21, with test signals generated dynamically depending on the DUT's properties, including band limited pink noise, filtered 24 dB/octave, and level adjusted.

Directivity patterns of speakers can be measured in combination with the HRT I turntable, allowing for generation of polar diagrams with levels according to different angles of direction, and generating 3D frequency response spectrograms versus direction and level.

www.head-acoustics.com

HEAD acoustics

Fully Automated Testing of Voice-Controlled Smart Home Devices

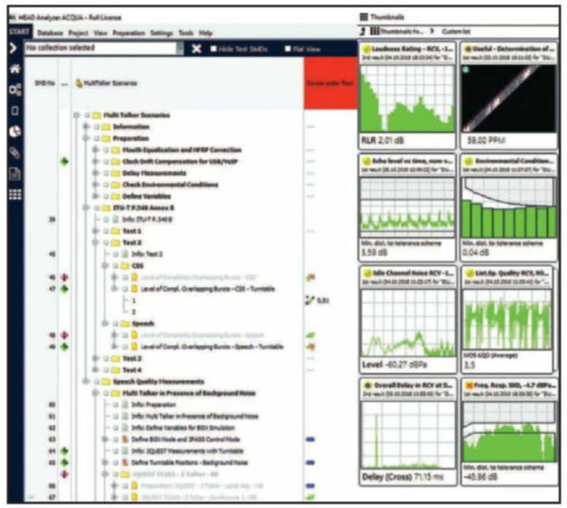
Apart from offering closed-loop measurements of devices connected to labCORE via Bluetooth, USB, or analog, using the HQS-Audio, which also enables open-loop (triggered) measurements when there is no possible connection with the DUT, HEAD acoustics also created the HQS-SmartHome software, specifically for testing communication and audio quality of voice-controlled smart home devices.

The database for the ACQUA analysis system contains an efficient test series for analyzing and optimizing smart speakers, speakerphones, and conferencing devices. HQS-SmartHome now provides a wide range of realistic and reproducible measurements and simulations for research and development phase up to production.

With this new software, HEAD acoustics now offers a solution ideally suited to ensure the high quality of smart home devices, which enables users to simulate realistic background noise and reverberation and perform directivity measurements, since voice-operated smart home devices are used in various situations and rooms with different acoustic room properties.

Settings with one single talker as well as with multiple talkers are taken into account. And since talkers may move around the room and interfere with each other, HQS-SmartHome also features a test series for multiple talker scenarios.

For simulating the different positions of moving speakers, directivity measurements are implemented in HQS-SmartHome.

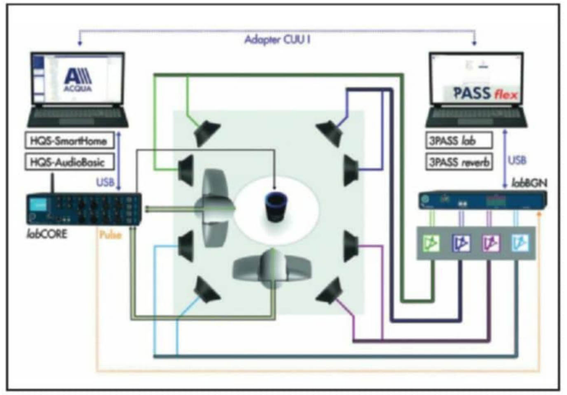


The database HQS-SmartHome for the analysis system ACQUA provides realistic and fully automated measurements for testing and optimizing communication quality of voice-controlled smart home devices, speakerphones, and conference devices.

In addition, the database offers tests for evaluating speech and transmission quality under realistically simulated background noises for various everyday situations, enabling users to determine speech quality in the presence of reverberance. Thus, acoustic room properties are correctly simulated in the test laboratory.

These measurements require the background noise simulation systems 3PASS lab or 3PASS flex as well as the option 3PASS reverb. Additionally, the database allows users to reliably test the performance during double talk.

For simulating the different positions of moving talkers, directivity measurements are implemented in HQS-SmartHome, performed in combination with the high-precision turntable HRT I. Furthermore, HQS-SmartHome provides traditional



This measurement setup of HQS-SmartHome consists of two artificial head measurement systems positioned around the DUT, ACQUA including HQS-SmartHome, and further software and hardware components.

Comsol

Brüel & Kjær HATS Modeled with COMSOL Multiphysics Software

Engineers at GN Resound used the boundary element method for acoustics to create a simulated version of a physical Head-and-Torso Simulator (HATS), which is a critical tool in evaluating the performance of acoustic devices that interact with the human body, including headsets, hands-free communication devices, and hearing aids.

Representing the human body's influence on airborne sound signals is no straightforward task. But in order to evaluate the performance of acoustic devices that interact with humans, it is a necessary challenge to undertake. In situ electroacoustic testing is a standard method of performance evaluation, but the irregular shapes and extensive variations of the torso, head, pinna, and ear canal often pose quite the headache for acoustic engineers.

To make acoustic data collection as accurate as possible, sound and vibration measurement equipment manufacturer Brüel & Kjær developed the Head-and-Torso Simulator (HATS), a manikin that accurately reproduces the geometric features and acoustic properties of an average adult head and torso (see **Figure 1**). Composed of built-in mouth and ear simulators, the manikin is ideal for performing electroacoustic tests on headsets, headphones, hands-free communication devices, hearing aids, helmets, and hearing protectors. Usually, the ear simulator is a half-inch microphone connected to a preamplifier, while the mouth simulator is a high-compliance loudspeaker with a sound pressure distribution that simulates a talking average adult human mouth.

GN Resound has been using a HATS for many years in its measurement setup to investigate the effects of the geometry of ears on the perceived sound at the eardrum and to determine directionality characteristics due to the scattering by the ear-and-head geometry. Recently, a mathematical model for HATS has been constructed in the COMSOL Multiphysics software. The task was carried out by Senior Engineer René Christensen, using the boundary

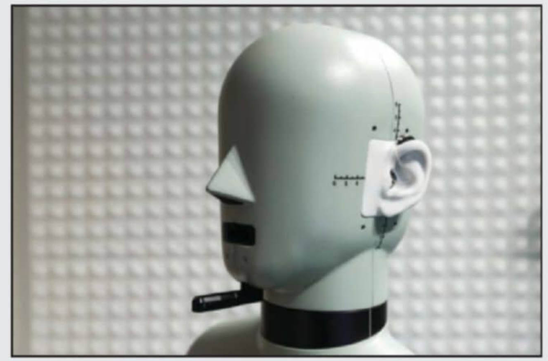


Figure 1: Head-and-Torso Simulator (HATS) being used to test an integrated helmet system

element method (BEM) functionality. Once the model was complete, several colleagues saw an opportunity to utilize this simulation tool instead of a manikin for their respective purposes.

Why the Boundary Element Method?

The combined influence of the torso, head, pinna, and ear canal on airborne sound signals and the acoustic field around the head is known as the Head-Related Transfer Function (HRTF). "For a given point in space away from the head, you can describe how a sound originating from that point will arrive at the eardrum," explains Christensen. This is a deterministic approach that allows the simulated sound to be visualized for different microphone placements (see example points of interest shown in **Figure 2**), aiding engineering decisions and enabling the exploration of how the head and torso affect the total sound pressure (see **Figure 3**). The challenge with acoustics simulations that take place over meter-scale distances is that the commonly used finite element method (FEM) is computationally expensive and requires considerable amounts of RAM.

For Christensen, who performs most of his simulation work on a laptop, this was a severe constraint. But with

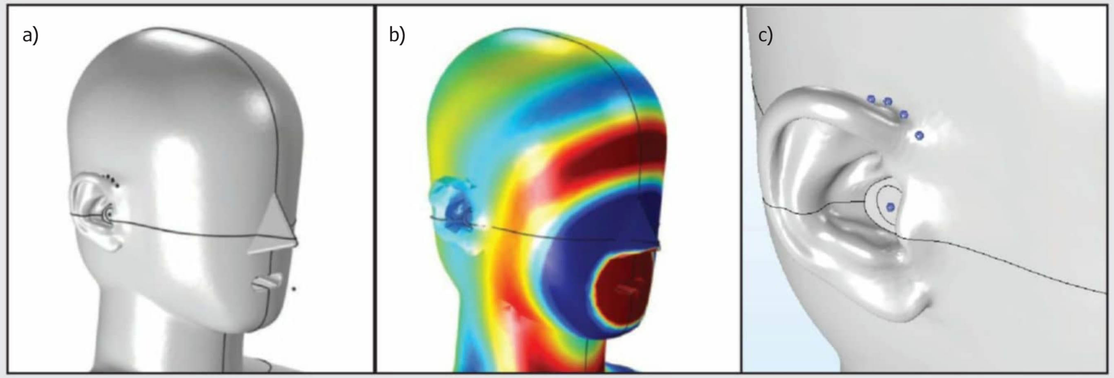


Figure 2: (a and b)—"Near-ear" points where the mouth-to-point near-ear transfer function is evaluated. The mouth-to-point near-ear transfer function is a more specific type of head-related transfer function. (c)—Sound pressure distribution is shown at 3200 Hz. The red corresponds to high, positive pressures, while the blue corresponds to high, negative pressures.

the COMSOL Multiphysics software he was able to choose which method to use and, in this case, he took advantage of the boundary element method (BEM) functionality available for acoustics to run much more efficient simulations. While BEM demands more computation per degree of freedom than FEM does, BEM usually requires significantly fewer degrees of freedom to obtain the same level of accuracy over large volumes. BEM allows you to extract sound pressure values at any point in the domain while only meshing and performing calculations on a surface, providing a substantial computational advantage. The entire volume needs to be meshed instead with FEM, which is feasible when the near-field is of interest. With the use of BEM, Christensen could now keep his simulations within his computational resources and perform them on his laptop. "It's a lot easier to set up with the boundary element method," Christensen says. "If you have a good geometry, you can just create a surface mesh which is such a luxury. Usually you would use a volume mesh and have to do things like resolve the air and apply radiation conditions to limit reflection, but this is all implicitly incorporated in BEM."

Talking and Listening

The beauty of HATS is that it can be used to model situations with several sources, an example being if a user speaks and additional sounds are present, as is often the case with a headset user talking in a busy office environment. Hearing aids are especially complicated because they consist of both microphones and loudspeakers, and this can lead to feedback. "There is an extra layer of complexity because there are two microphones in a typical hearing aid, and both of them pick up sound and vibration," Christensen explains. "Some of this stems from the hearing aid itself, and so there is a potential for feedback, something that will often result in a horrible noise."

With BEM, the simulation is relatively straightforward because of how little physics needs to be set up. In such instances, an acceleration boundary condition is applied on the mouth to mimic a loudspeaker. HATS itself is assumed to be rigid, and any radiation conditions that would ensure acoustic waves are not reflected are automatically fulfilled with BEM. From there, the sound pressure field around the head and torso is calculated, and once the simulation has been run, several microphone positions can be investigated with post-processing. If a hearing aid geometry is present,

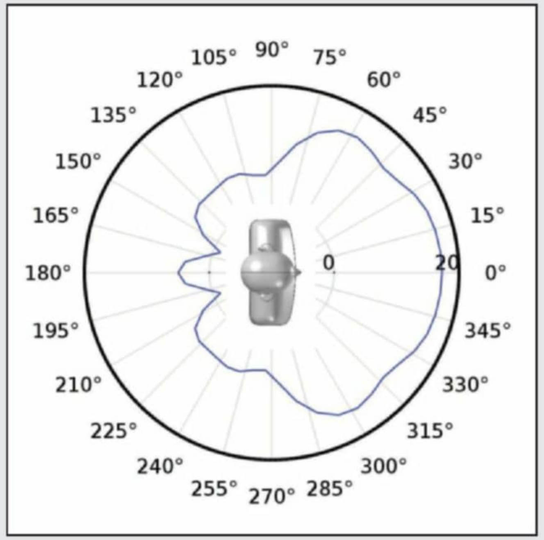


Figure 3: The polar plot of total sound pressure for 1 m radius at 3200 Hz

Calibrated
Electro-Acoustic Test

Measurement microphones.

Integrated amplifiers & power supplies.

Analyzers & test software.

Measurement results you can trust.

Audio precision

www.ap.com

mechanical designers can indicate all feasible positions, and the optimal one can be determined with the information extracted from the simulations. "With BEM, there is really not much physics to set up," Christensen says. "Since we are interested in mouth-to-ear relations, I apply an acceleration in the mouth and extract pressure at points in front of the mouth or in and on the ear, and that is it. The only physics is acoustics."

Future Use

Christensen's HATS model is of supreme value to him and his colleagues because once it has been solved, there is a tremendous amount of information available to extract using various post-processing tools. Different configurations (e.g., domes that help aids fit snugly or ports that connect the inner ear to the exterior) can be investigated, new transfer functions can be derived for assorted points, and it can be prepared for use with other software.

Christensen also says that as he continues to improve his acoustic models, not just for HATS but vibroacoustics in general, he will continue to use COMSOL Multiphysics. "I like the software because it has a very familiar and intuitive feel, and you can add or modify equations to accommodate your needs, which I often take advantage of," he explains. "The software is very beneficial to someone like me who tries out physics described by mathematics not necessarily already available in it. I like to ask myself, 'Can you give me an equation that describes this problem?' And if so, then you can probably set it up and solve it in COMSOL."

www.comsol.com

Dayton Audio

Exploring What's Possible with the OmniMic V2 and DATS V2 Computer-Based Audio Test & Measurement Systems

Not long ago accurate and detailed loudspeaker measurements required expensive and specialized equipment and/or lots of time and patience. Today, thanks to Dayton Audio's OmniMic V2 and DATS V2 measurement systems, these once hard to come by measurements can be made quickly. Even a novice speaker building hobbyist with any Windows-based computer, laptop, or even tablet can easily make accurate measurements. The combination of these two affordable systems make it simple to measure Thiele-Small (T-S) parameters, frequency response, impedance, impulse response, distortion, and so much more in just a matter of minutes.

The Dayton Audio DATS V2 is a compact and remarkably accurate impedance measurement device that derives individual driver parameters and other data to design enclosures and passive crossovers predictably. The software will run a sweep from 1 to 20,000 Hz and quickly plot an impedance response using 30,000 data points. An accurate and detailed impedance measurement is an extremely important resource for speaker designers. First, the impedance response is needed to derive a driver's T-S parameters, which in some cases, can vary quite a bit from what is on the spec sheet. These measurements are required to design an enclosure for an accurate and predictable response. A driver's electrical parameters (Fs, Qes, Re, Le, etc.) are easily measured by connecting the DATS V2 hardware to the driver's terminals and running the software sweep. By adding a little mass to the driver, DATS V2 can calculate the mechanical parameters (Mms, Vas, Bl, Cms, etc.) and all the T-S parameters are ready to export into the user's favorite enclosure modeling software.

In order to design a passive crossover network that will perform as intended, the driver's impedance response is a necessity. All of the classic crossover formulas and simple calculators out there assume that drivers have linear impedance, but unfortunately, that just isn't the case. Users can design professional quality crossovers in no time by importing the impedance response from DATS V2 into their favorite design software. In addition, the DATS V2 system can make incredibly accurate measurements of resistors, capacitors, and inductors. Plus, the software can even compare a known good driver to other drivers for a quick quality control check (pass/fail) when dealing with multiple drivers.

Accurate electro-mechanical parameters are only half of the equation. Actual acoustic measurements are also necessary to the speaker design process. The Dayton Audio OmniMic V2 system provides fast, comprehensive, and accurate acoustic measurements that are useful for everything from speaker design to room treatment. Making impulse and frequency response measurements are as simple as installing the easy-to-use software and downloading the unique microphone calibration file.

Dayton Audio's DATS V2 is a complete and easy-to-use audio test system toolkit.

The OmniMic V2 offers the ability to measure the response of drivers inside the intended enclosure, making the need to compensate for baffle diffraction unnecessary. These measurements further simplify crossover design and ensure that the intended final frequency response will be almost exactly as predicted. Using the system, users can quickly measure finished speakers to identify that all drivers are operating in phase and ensure the crossover is performing as expected.

Dayton Audio's OmniMic V2 is a must-have omnidirectional acoustic measurement system and the new hardware provides fast, comprehensive, and accurate results.

The OmniMic V2 measures harmonic distortion just as easily, making it simple to compare drivers for intended applications or determine just how low to cross over a midrange or tweeter.

Beyond speaker design, the OmniMic V2 will also help set up entire systems for optimal performance. Using the frequency response measurement, users have the ability to EQ a system while watching the results of the equalization in real time. This real-time analysis makes it easy to tweak a system for a desired sonic character. Moreover, the OmniMic V2 can measure the actual characteristics of a room. The RT60 measurement capability aids design considerations in room damping and diffraction applications and sees the effects thereof. Bass decay measurements help in subwoofer placement by identifying room modes. For the casual home user, professional speaker designer, or the seasoned pro-sound installer, the OmniMic V2 acoustic measurement system is key to guaranteeing that we are getting the most from any speaker system.

This description only scratches the surface of what is possible with the DATS V2 and OmniMic V2 measurement systems. With the surprising capability of these two systems, users discover new uses and techniques for them constantly. Dayton Audio provides technical support and guidelines to help anyone become familiar with the use and interpreting the results of these measurement systems. Furthermore, with so many of these popular systems currently in use, there is a huge community of experienced users out there to provide tips and support to beginners.

www.daytonaudio.com

Audiomatica

New Clio 12

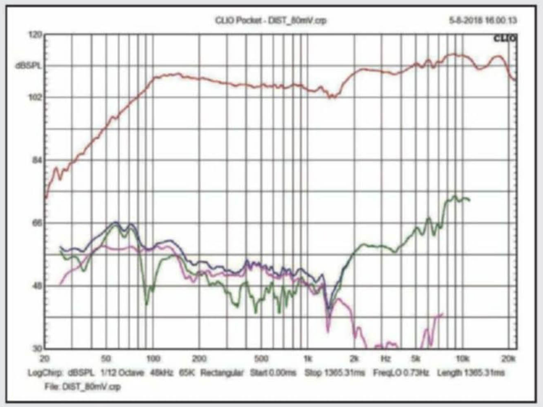
Audiomatica designs, manufactures, and consults for the professional and consumer audio, acoustics, and computer industries. Products include the CLIO system, which is a complete range of measuring equipment and accessories. Inside the CLIO product range we find two PC-based electroacoustics analyzers, the CLIO 12, and the CLIO Pocket.

CLIO 12 is the newest measurement software for the CLIO system and controls the latest Audiomatica FW-02 USB Interface (24-bit at 192 kHz). The software offers a professional suite for electrical and acoustical measurements running under Windows from XP to 10 (USB hardware driver is fully compatible with Windows 10).

The CLIO 12 software is available in two versions. The CLIO 12 Standard is a laboratory-grade software with most of the features, while the CLIO 12 QC version adds a Quality Control Processor for state-of-the-art testing and controlling a production line with a versatile script to handle test sequences. It also provides the best solution for complex routines such as Rub & Buzz testing, while adding a TCP/IP measurement server to offer state-of-the-art features for anyone in need to create custom applications. CLIO 12 QC also adds some particular applications like 3D measurement analysis. The CLIO 12 Standard software was recently expanded with a direct and indirect STI/STIPA measurement tool and Cycle Wavelet processing.

Market Update

Compared to other measurement systems, CLIO 12 concentrates the power of many different instruments into a single one. Three different measurement techniques are available for system identification and characterization, including MLS & LogChirp analysis using either pseudo-random noise or logarithmic chirps as stimuli; Sinusoidal Sweeps using sinusoidal signals; and fast Fourier transform (FFT), real-time analysis (RTA) and "Live" Transfer Function, enabling users to choose any stimulus, even music. And while other instruments offer one single possible measurement choice, CLIO 12 allows visualizing frequency response, impedance, or other parameters from different points of view.



Clio Pocket is Audiomatica's multiplatform personal measurement system, which comes complete with CLIO Pocket software for Windows and macOS X, the CLIO CP-01 Audio Interface (USB 2.0), a microphone, and accessories, all housed in a convenient carrying case. The new CLIO Pocket version 2 software added distortion measurement using the LogChirp measurement menu (pictured at left).

www.audiomatica.com

Virtins Technology

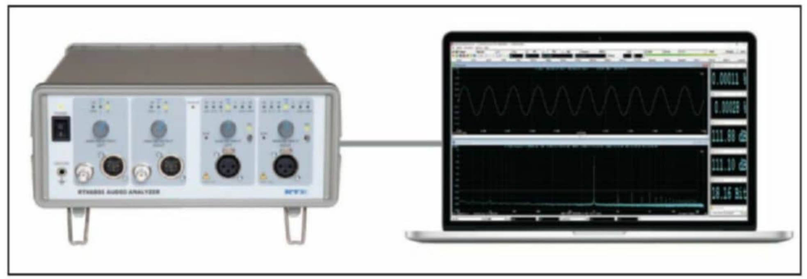
RTX6001 Sudio Analyzer Powered by Multi-Instrument

Multi-Instrument, developed by Virtins Technology, has been around for years as a powerful multi-functional virtual instrument software. It supports a variety of hardware ranging from sound cards, which are available in almost all computers, to proprietary ADC and DAC hardware (e.g., NI DAQmx cards, VT DSO, VT RTA, and VT IEPE). The software has recently added RTX6001 audio analyzer onto its list of supported hardware.

RTX6001, developed by RTX (www.rtx.dk), is a PC-based high-performance audio analyzer with a 24-bit ADC and 32-bit DAC, a maximum sampling rate of 192 kHz and a bandwidth of 0~90 kHz. It has two balanced XLR inputs with seven input ranges: -20 dBV (±141.4 mV), -10 dBV (±447.2 mV), 0 dBV (±1.414 V), 10 dBV (±4.472 V), 20 dBV (±14.14 V), 30 dBV (± 44.72 V), 40 dBV (±141.4 V), and two balanced XLR outputs with three output ranges: -20 dBV (±141.1 mV), 0 dBV (±1.414 V), and 20 dBV (±14.14 V). It also provides two single-ended BNC outputs at half of the balanced voltages. These input and output ranges should be sufficient to test a wide range of audio gear without additional amplification or attenuation.

RTX6001 boasts extremely low distortions and noises. The typical loopback residual THD and THD+N at 1 kHz are 0.0001% and 0.0003%, respectively, making it one of the top performers in the market. The loopback frequency response flatness is remarkably ±0.01 dB from 20 Hz to 20 kHz, and ±0.2 dB from 10 Hz to 80 kHz. RTX6001 communicates with a PC through an isolated USB 2.0 port. Thus, its measurement section is isolated from the noisy PC environment. The combination of RTX6001 and the Multi-Instrument provides a cost-effective yet powerful solution for audio tests and measurements, at only a fraction of the cost of other audio analyzers with similar capabilities and performances.

To help new users to quickly adapt, Multi-Instrument comes with a set of pre-configured settings for various tests. Twenty of them can be loaded through a single mouse click in a user configurable toolbar. These tests include fractional octave analysis, measurements of THD, THD+N, SNR, SINAD,



The RTX6001 is a PC based 24-bit 192 kSPS 90 kHz high-performance audio analyzer with dual-channel balanced inputs and outputs and an isolated USB port, seamlessly integrated with the Multi-Instrument software to form a completed multi-function high-performance audio analyzer.

AXPONA
AUDIO EXPO NORTH AMERICA
APRIL 12-14, 2019 • CHICAGO
RENAISSANCE SCHAUMBURG
HOTEL & CONVENTION CENTER

HEAR IT ALL.

Immerse yourself in the AXPONA experience. Indulge in discovering the best in loudspeakers, headphones, turntables, amplifiers, streaming services and so much more. Explore over 185 listening rooms, Ear Gear Expo, The Record Fair plus enjoy seminars, product demos and concerts. Connect with fellow enthusiasts, designers and industry experts.

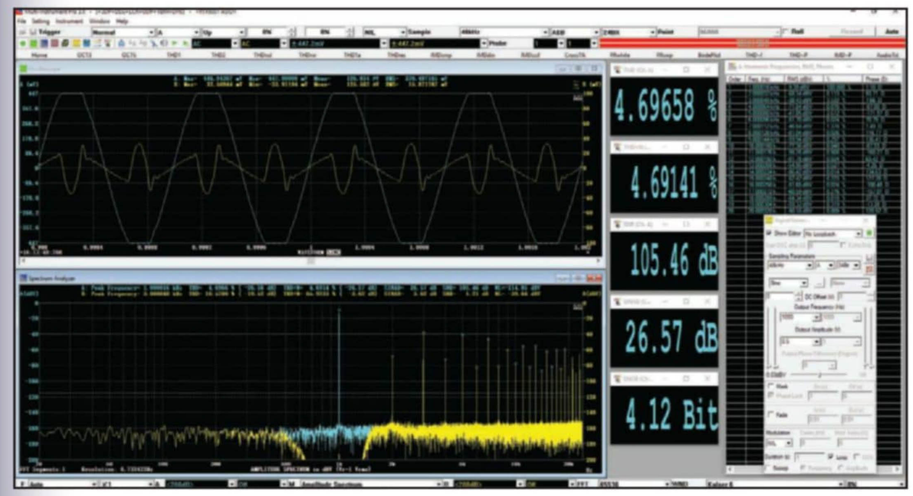
Audition the largest gathering of high-fi products in North America. It's all in one place over three incredible days! See you in Chicago this April!

Streaming Sponsor: Qobuz

**Tickets on Sale Now!
For information, visit
www.AXPONA.com**

Learn about exhibiting at AXPONA 2019. Contact Mark Freed: mark@jdevents.com, 203-416-1927

Market Update



A clipped 1 kHz sinewave vs. its residual harmonics and noises

Noise Level, ENOB, IMD SMPTE/DIN/CCIF, crosstalk, frequency response, Bode plot, THD vs frequency, THD vs power, and an automated audio parameter test.

One pre-configured test that is not commonly available in other audio analyzers is the waveform comparison between a sinewave and its residual harmonics and noises (i.e., with the fundamental removed). The test uses a FIR filter to remove the fundamental and thus no external notch filter hardware is needed. A clipped 1 kHz sinewave vs. its residual harmonics and noises is shown in the screen above. The latter was obtained by attenuating the fundamental of the original sinewave by more than 120 dB using a FIR band stop filter. The time delay introduced by the FIR filter is automatically corrected by the software in order to make the two waveforms time-aligned for comparison. The clipped sinewave is used in this example for evaluation purposes as its residual harmonic waveform is already known.

Once users get familiar with the software, they can start to design their own tests by modifying various parameters. Taking the frequency response test as an example, there are many ways to do it in Multi-Instrument. It can be done with a white noise, MLS, linear-frequency-sweep, multi-tone, and impulse stimulation from the signal generator and narrow-band FFT analysis in the spectrum analyzer.

It can also be done with a pink noise/logarithmic-frequency-sweep stimulation from the signal generator and octave analysis in the spectrum analyzer. The software has a test scheduler called Device Test Plan. It can be used to configure and perform a sequence of test steps. Some preconfigured device test plans are provided (e.g., THD and THD+N vs frequency, crosstalk vs frequency, etc.). The aforementioned frequency response test can also be conducted using a frequency-stepped-sinewave stimulation in the Device Test Plan. Furthermore, it is possible to convert the above frequency response tests to impedance vs. frequency tests in the software. Multi-Instrument pioneers the incorporation of some new distortion tests such as the psychoacoustically based distortion "GedLee Metric" and non-coherent distortion.

Multi-Instrument supports more than 200 Derived Data Points (DDP) including 16 User Defined Data Points (UDDP). A DDP is a quantity derived from the raw ADC samples (e.g., maximum, mean, minimum, RMS, and THD). A UDDP can be defined by a mathematical expression consisting of other DDP and UDDP. The use of UDDPs expands the measurement capability of the software greatly as it enables the software to measure various quantities directly without hard coding measurement algorithms for them.

RTX6001 supports remote and local control modes. By default, it works under remote control mode and is fully controllable from Multi-Instrument. In view of the fact that some users may prefer the look and feel of hardware knobs and switches to their software alternatives, RTX6001 also provides a set of hardware knobs and switches on its front panel. They can be enabled if the user opts to delegate control from Multi-Instrument to the local control panel. Under the local control mode, the software can sense the positions of the hardware knobs and switches and acts accordingly when a change is detected.

In additional to being used as an audio analyzer, the combined system can be used to perform non-audio tests such as power supply output impedance vs. frequency, power supply ripple rejection, etc. The test capability of Multi-Instrument is highly expandable. Many tests that have to be done through custom programming using Matlab, Labview or C++ can actually be performed through configuration without coding in Multi-Instrument. Moreover, Multi-Instrument can work as an ActiveX automation server which can be integrated seamlessly into an ActiveX client program.

Multi-Instrument can be downloaded for 21-day fully functional free trial using a PC sound card as the ADC and DAC device.

www.virtins.com | www.rtx.dk

Klippel

What's New with the Klippel Analyzer System

Klippel has released a new software update for the Klippel Analyzer System, unifying both R&D and QC applications that have been formerly available as separate hard- and software system packages. Klippel dB-Lab has been the base platform for all software modules so far, but the unified release and common hardware platform provides completely new application opportunities and improved workflow in product development and manufacturing. Additionally, new software modules and major features in existing modules have been added.

Unified Hard- and Software Framework for R&D and QC

In modern audio system manufacturing, research and development (R&D) should be closely connected to quality control (QC) in pre-production and end-of-line (EOL) testing for optimal product performance, maximum yield, and lasting customer satisfaction.

The new unified release of the dB-Lab framework software benefits from the latest modular Klippel Analyzer 3 hardware, which can be tailored to the particular requirement of the application while supporting the full range of Klippel software modules. Extension cards such as Laser (includes the microphone), XLR, Speaker, and Amplifier Card provide a high level of scalability and flexibility.

Klippel R&D modules (e.g., LSI or TRF) and Tasks of the QC module (e.g., Air Leak Detection) can now be used with the same analyzer hardware platform and even used in the same batch run sequence, extending the test system's capabilities. Since the requirements in EOL testing are still different from lab applications, the software comes in two dedicated distributions that are fully compatible, but include different tools (e.g., QC Start for operator-oriented EOL test management). Test results, templates, settings, and limits can easily be exchanged for optimal communication and comparability between design and manufacturing. Therefore, the same tests can be run on prototypes in the lab as well as in pre- and mass production.

Since compatibility with earlier software releases and existing analyzers is ensured, longtime users of the Klippel systems will feel completely familiar with the software's look and feel, while benefiting from the new capabilities. The Klippel software license infrastructure has been revised. Therefore, a new USB license dongle will be provided in order to update from older versions.

New Features for Complex QC Tests

Modern audio devices (e.g., smart speakers, soundbars, and wireless headsets) are getting more complex through high-integration, multi-channel in- and output, wireless signal transmission, and extensive signal processing (e.g., virtualization, beam steering, and active noise cancellation). Consequently, the test system needs to be highly flexible in order to reflect those capabilities in quality control.

In combination with the Klippel analyzer hardware, the QC framework module handles all kinds of audio devices

(ASIO or Windows Direct Sound), either as a DUT (e.g., a USB headset) or a test interface (e.g., a sound card). Multiple QC operations can be run in a test batch covering different input and output devices, transmission channels, sample rates, codecs, and so on.

Using the capabilities of the KA3 hardware, up to eight channels (4× microphone and 2× impedance channels) can be simultaneously measured. This significantly speeds up testing of passive stereo systems (e.g., headphones) or multi-microphone setups (e.g., speaker cabinet air leak detection).

To simplify the test setup of complex sequences, single test steps in the QC test sequence may be activated/deactivated as needed. The new, free sequence control add-on feature allows repeating or skipping individual test steps dynamically, based on test verdicts or user interaction.

To get a practical insight into those capabilities, the new Application Note 73 dedicated to quality control of passive, digital, and wireless headphones and headsets provides some good examples. **Figure 1** shows an exemplary hardware setup sketch for a Bluetooth headphone test based on KA3 hardware, a Bluetooth interface and additional, speakers for active noise cancellation (ANC) performance check. All application notes can be downloaded from the Klippel website at www.klippel.de/know-how.

New Time-Frequency Analysis

The new Time-Frequency Analysis (TFA) module is a powerful tool for investigating the spectral content of audio signals over time. Both, arbitrary WAVE files or signals recorded by Klippel measurement modules may be analyzed. For this purpose, the tool provides wavelet analysis, auditive

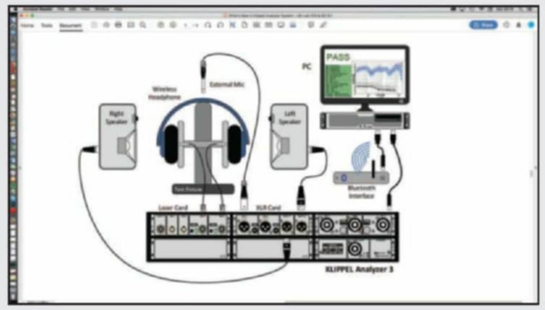


Figure 1: Hardware setup sketch of a complex QC test for Bluetooth headphones with ANC.

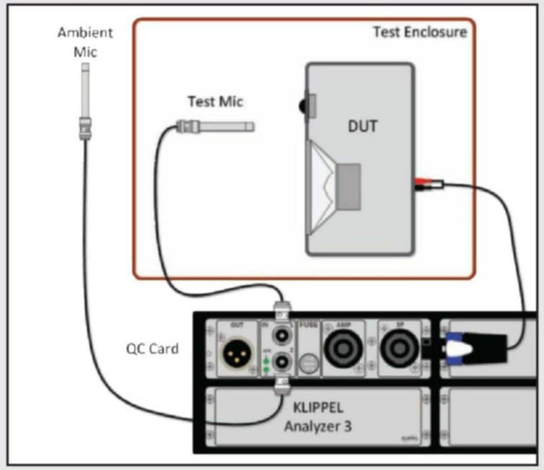


Figure 2: EOL test example setup for a passive loudspeaker using the KA3 with QC Card

filter bands or short-time Fourier transform (STFT). The results may be displayed in 3D plots (time slices), waterfall diagrams, or sonographs with high temporal resolution due to interlaced analysis.

A useful application is the analysis of the impulse response and nonlinear distortion generated by an audio system. The sonograph reveals nonlinear signal components generated by the test stimulus that are at much higher frequencies than the excitation frequency.

For defect analysis, the TFA allows for the investigation of temporal and spectral fine structure in the chirp sound pressure response of a bad speaker. While deterministic defects (e.g., coil bottoming) result in higher order harmonics, random defects cause impulsive sounds—the energy is distributed over the entire frequency band for a very short time. The information provided by the TFA is very helpful in optimally setting Rub & Buzz filters (harmonic order, bandwidth, etc.) for the EOL test.

QC Card for Klippel Analyzer 3

The QC Card is a novel extension card for the Klippel Analyzer 3 hardware framework (see **Figure 2**). This cost-efficient all-in-one card solution was tailored to meet essential needs of transducer and audio system testing at the end of the production line and quality control.

Due to the integrated 45 W amplifier, most speaker types can be tested without the need for costly additional amplifier and cables, providing extremely compact test setups. However, for high power applications, an external amplifier can be optionally looped in, using the balanced line output and speakON input.

The single speaker output channel provides premium Klippel features such as four-wire, high-power voltage and current sensing with switchable sensitivity, an electronic relay, and fuse protection.

Two BNC microphone inputs with software switchable IEPE constant-current supply and adjustable input gain are dedicated to testing frequency response, distortion, Rub & Buzz, air leakage noise, and many other parameters while ensuring full ambient noise immunity.

Improved Testing of Bluetooth Devices

Klippel has released a minor software update for dB-Lab 210/QC 6.1 with extended capabilities for testing Bluetooth-enabled audio devices such as portable and smart speakers or true wireless ear- and headphones in both QC and R&D software framework.

A new dedicated measurement mode was added to the Transfer Function Measurement (TRF) module for synchronized measurement of frequency response, distortion and many other parameters in presence of long and varying delays as occurring in wireless audio devices. Additionally, the stimulus signal can be exported to audio files for open-loop testing of devices with no signal input (e.g., smart speakers and smartphones).

Also, the Near Field Scanner system (NFS) benefits from the update for quasi-anechoic directivity scanning of wireless speakers in normal rooms (see **Figure 3**). Extended with a stationary reference microphone, phase accuracy is ensured to provide accurate directivity data.

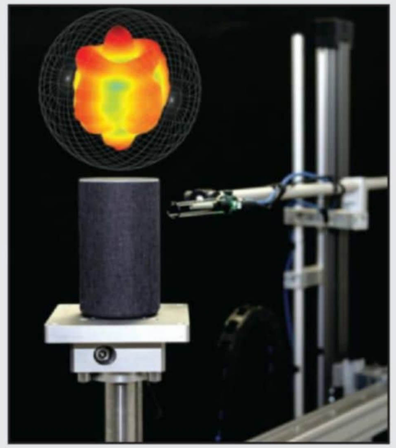


Figure 3: Klippel NFS quasi-anechoic directivity scan of a smart speaker (e.g., balloon plot).

For efficient EOL testing of Bluetooth-enabled devices, the latest software update improves third-party sound device handling and adds automated device pairing for the Klippel QC framework. Devices can be paired based on their friendly name or address that may be imported from a text file, QR code or NFC tag. Also, simple pairing is provided.

In combination with the advanced, yet simple features of the External Synchronization (SYN) option, testing digital and wireless audio devices in both closed and open-loop setups is made easy and robust.

www.klippel.de

Engineering Consulting for the Audio Community

Celebrating 36 YEARS 1983-2019

MENLO SCIENTIFIC

- Liaison Between Materials, Suppliers, & Audio Manufacturers
- Technical Support for Headset, Mic, Speaker & Manufacturers
- Product Development
- Test & Measurement Instrumentation
- Sourcing Guidance

MENLO SCIENTIFIC

USA • tel 510.758.9014 • Michael Klasco • info@menloscientific.com • www.menloscientific.com

Examining the miniDSP EARS Headphone Test Fixture

(Photo courtesy of Cynthia Wenslow)

In this article, our esteemed contributor explores the capabilities of "EARS," the Earphone Audio Response System. miniDSP says its EARS brings headphone measurement capability to "the masses" with its affordable measurement rig for headphones and IEMs, which miniDSP has built using the same technology as its popular UMIK-1 calibrated measurement microphone. Alternatively, one could use EARS as a binaural microphone with USB audio streaming.

By
Stuart Yaniger
(United States)

Over the years, I've done extensive testing of amplifiers, preamplifiers, signal processors, loudspeakers, microphones, and more, for which there are pretty standard and well-accepted test methods for all of these categories. Their basic performance parameters are likewise well-known. For example, amplifiers should have flat frequency response, low noise, low source impedance, low distortion, and the like, all of which are objectively and repeatably measurable across nearly any type of test equipment. Loudspeakers are a bit harder, but we know how to measure anechoic and quasi-anechoic response, on and off axis, and any good test microphone will give similar results. Moreover, the targets in all of these measurements are well defined and (basically) universal.

Challenges Associated with Headphone Testing

Here's the main problem with headphone testing: It's not transportable into the intended environment. What I mean by that is if I put a set of loudspeakers in your room and measure a gated frequency response (which removes room reflections), it will be the same as the gated frequency response of that loudspeaker in my room. Likewise, the measured in-room response will be the same for any listener in the room (at least for a given seating position!). This is not true for headphones—the frequency response on my head with my ears will not be the same as the frequency response on your head with your ears since the size, shape, and consistency of ears is not constant and the ears are a significant part of the acoustic cavity being measured. And worst of all, that difference is intrinsic, it cannot be gated out.

So how then can we characterize a headphones' frequency response? The short answer is, we can't. We can only characterize their frequency response in conjunction with a particular test fixture. This leads to two new problems—which test fixture is "correct" and what should the target response be?

One approach to the first problem is using a coupler that simulates the acoustic load of an ear (though that begs the question of which ear). Over the years, there's been several "standard" ear couplers, including National Bureau of Standards (NBS) 9A, International Electrotechnical Commission (IEC) 60318, IEC 60711, and American National Standards Institute (ANSI) S3.25-1989.

More recently, the IEC 60268-7 standard has become dominant, calling out the type of artificial ears, the couplers (which simulate the response of ear canals), and test signals. This approach does solve the issue of portability, so that two different measurers can compare their results. And it gives sort of a rough idea what the response will be with average ears, but of course, that response will not be correct for any individual user. Mike Klasco (president of Menlo Scientific, Ltd.) and João Martins (editor-in-chief of *audioXpress*) presented an excellent review of current commercial offerings last year (see Resources). It's well worth reading to see the type of equipment used in serious professional laboratories.

The second problem, that of the target response, is intimately tied up with the question of measurement method. Should a headphone be measured on a flat panel, adjusted to a flat frequency response, then the combination of the headphone and test jig as a flat response be adopted as the frequency response target? Unfortunately, the acoustic cavity formed between the flat plate and the headphone driver bears no resemblance to that of the cavity with an ear in it, so that flat response will not guarantee a natural tonal balance in use.

To try to work around this conundrum, Sean Olive's research group at Harman International developed their own target curve by taking a standard fixture (in their case, a full Head-and-Torso Simulator) and measuring the free-field response from loudspeakers known to be reasonably flat in frequency response. Matching headphone frequency responses taken with the simulator to this curve, they had trained listeners adjust an equalizer until the sound was, to their judgment, "right." As it turned out, most of their adjustment was at the bass end of the spectrum, with several extra decibels of bass boost sounding most natural to them.

There is a popular expression in audio, "Circle of Confusion," which in this case is far worse than usual—I'd call it a "Hypersphere of Confusion." We may end up with a single standard, but in my opinion, we'll never end up with something that works as universally and with the ease of interpretation as that for loudspeaker measurements.

The miniDSP Test Jig

With that bit of pessimism stated, let's consider how individual users and hobbyists might perform headphone measurements on their own. Professional test fixtures compliant with existing standards (e.g., the Audio Precision AECM206 that was recently reviewed in *audioXpress*) are necessary for serious headphone research, but even the lowest-cost stereo models cost more than $10,000, which is a bit much for a small lab or a hobbyist.

Enter miniDSP, best known for its line of low-cost high-performance crossovers, equalizers, streamers, and signal processors. miniDSP's starting point was recognizing that, irrespective of the sophistication, all headphone test fixtures boil down to one or more miniature microphones embedded in a mechanical interface—most commonly, pinna replicas. The company's product offering, the EARS Headphone Measurement Jig (see **Photo 1**), is intended to fill the needs of headphone enthusiasts who want to perform their own measurements without taking out a second mortgage. I should note that, although its website and all the documentation calls this unit "EARS," the silk-screened name on the jig is "HEARS." For the purpose of clarity, I'll use the former name in this review.

miniDSP offers the following suggested uses for this device:

1. Check that your headphones are operating correctly (e.g., the balance of left/right frequency response).
2. Take measurements that you can use as a basis for equalizing your headphones.
3. Observe the effect on frequency response of different ear-pads and ear-tips.
4. Measure the effect of modifications to the headphones.
5. Compare your headphone measurements with other EARS users.

Based on its popular UMIK-1 room measurement microphone technology, the EARS basically consists of a pair of molded ears, two electret microphone

Photo 1: The miniDSP EARS headphone jig is a basic and inexpensive alternative to laboratory-grade measurement systems. (Photo courtesy of Cynthia Wenslow)

TEST & MEASUREMENT FOCUS

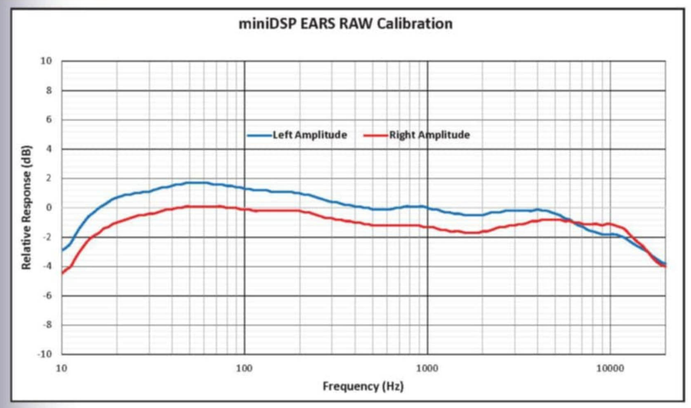


Figure 1: The RAW EQ files provided with the miniDSP EARS compensate for the frequency response of the embedded electret microphone capsules.

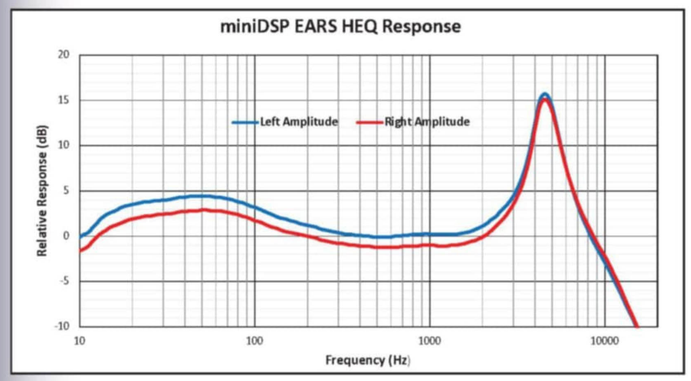


Figure 2: The HEQ equalization curves account for the acoustic effects of the simulated pinnae and ear canal, and are a starting point for users to equalize to a flat frequency response.

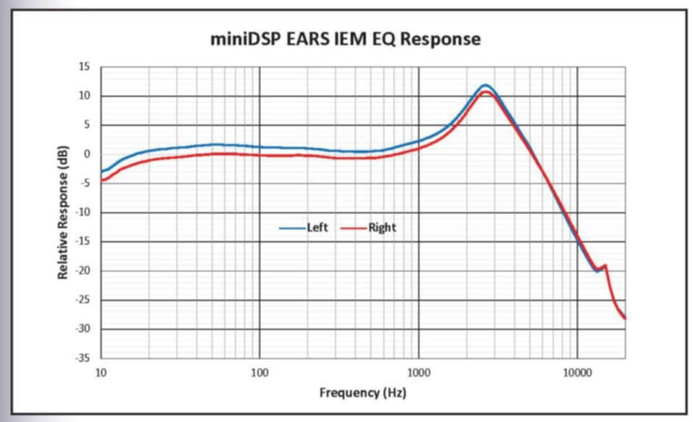


Figure 3: miniDSP also provides IEM equalization curves for users of in-ear headphones where the HRTF contribution of the pinnae are minimized.

capsules, and a USB interface, all mounted on a stamped metal stand with a curved upper piece to allow the headband of the headphones under test to be adjusted for an optimal fit. The vertical distance between the top of the jig and the ears is adjustable to accommodate a range of headphone sizes—the distance between the ears is fixed at about 14 cm (5.5″).

Unlike more expensive fixtures that use elaborate and expensive couplers, there's no attempt to simulate the acoustic effects of the ear entrance or canals—the passageway between the molded ears and the microphones is a simple cylindrical hole.

Because of the stamped sheet metal construction, the unit is relatively lightweight, with none of the acoustic isolation that metal mass brings to expensive fixtures. What that means is that you can't test the acoustic isolation of over-ear headphones, nor the effects of active noise cancellation circuitry.

This horizontal distance is small and not adjustable. Smaller than my head, smaller than my wife's head. Small enough that getting a lot of headphones to sit properly on the jig is an issue (with concomitant variability in bass response measurement because of reduced tension from the headphones' headbands). Hoping that I could modify this by putting spacers between the molded rubber ears and the metal stand, I removed the ears and found that the microphone capsules were glued to the metal and couldn't be easily repositioned on a new spacer. Adding the spacers would greatly change the acoustic properties because of the extra ear canal length. Several people on the Internet have suggested using rubber bands to hold the headphones more tightly against the ears, but the variability of that fix is unacceptable. I'd call this a basic design flaw, but it's one that's easy to fix, and I hope that miniDSP does that for the next version of this jig.

The interface electronics are connected to the user's computer via a standard USB printer cable. No special driver is needed, and the USB connection also provides power to the EARS. A set of DIP switches allows the gain of the microphone preamps to be adjusted to accommodate a wide dynamic range of test signals. The factory default setting is 18 dB, which allows a maximum SPL of about 120 to 125 dB SPL for a 0 dB FS output, but gain can be reset in 6 dB increments between 0 and 36 dB. Sample rate and bit depth should be fixed at 48 kHz and 24 bits for the jig to properly function.

Which brings us to a potential snag: calibration to obtain SPL. Because it's a USB device, the EARS jig will give a response curve referenced to full scale—but what's full scale in actual SPL? miniDSP provides a calibration procedure that works with Room Equalization Wizard (REW) software and (in theory) gives actual decibel sound pressure level (dB SPL). With any other measurement software (e.g., ARTA, Virtins MI, and RightMark), you're on your own. You can at least get into the ballpark by noting that at the default 18 dB gain setting, 0 dBFS equates to about 123 dBSPL.

And to be fair, REW is an excellent software package, free to use, and with a large user base for support. Nonetheless, you should be aware of this limitation.

The EARS jig is serialized—entering the serial number on the support page at miniDSP's website will get you access to a set of calibration equalization (EQ) files in text format. There are separate RAW files for left and right, which compensate for the non-flat response of the electret mic capsules, but not the fixture itself. miniDSP also provides files called HEQ, which provide frequency response compensation and for over-ear headphones, and IDF, which provide the same function but for in-ear monitors (IEMs). With the HEQ and IEM equalizations, the intent is to make the target frequency response curves for headphone equalization be flat, but you can certainly experiment with modifying the provided EQ files to suit your taste—and (this time, literally) your ears.

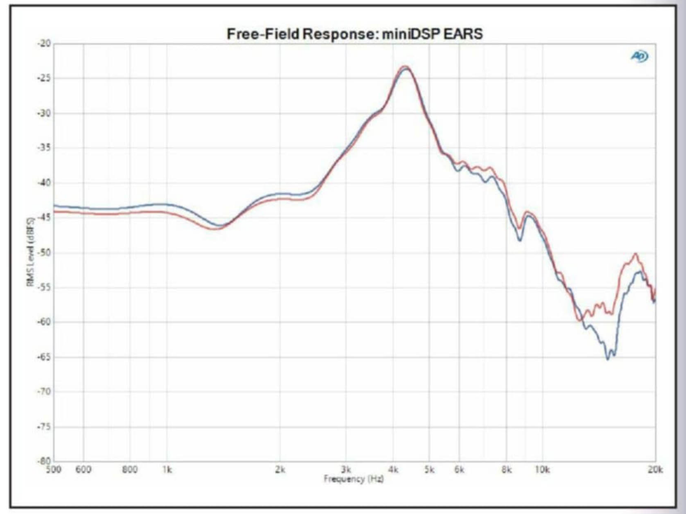
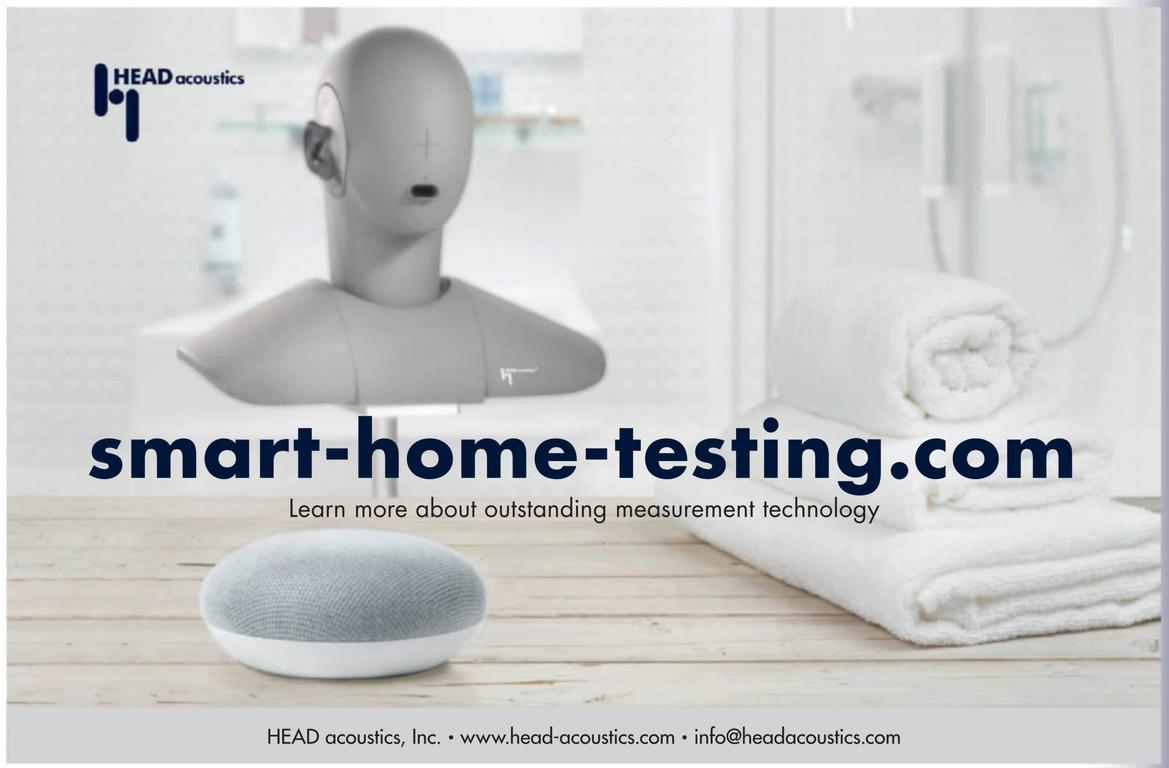


Figure 4: The free-field response of the EARS test jig was taken using a 1 meter distance from the source and gating out reflections.

Figure 1 shows a graph of the RAW EQ files provided for my particular EARS jig. **Figure 2** shows a graph of the HEQ files. There's a couple of decibels difference between the channels and the upper midrange and treble don't track well, but that's the price one pays for using inexpensive mic capsules. However, this is the point of providing



these curves—those differences can be removed by implementing the RAW EQ files.

You can see the effect of the jig on the acoustic cavity between headphones and jig due to the pinna and the simple hole used for the ear canal in the HEQ curves shown in Figure 2—the latter forms an almost undamped Helmholtz resonator, which causes the large peak at about 4.5 kHz. This peak dominates any waterfall plots and spectrograms, so the jig really should only be used for frequency response and not for spotting non-minimum phase phenomena. It's also evident that the channel matching issue is accounted for in these curves. The relatively steep drop in the treble is characteristic of this ear geometry as well.

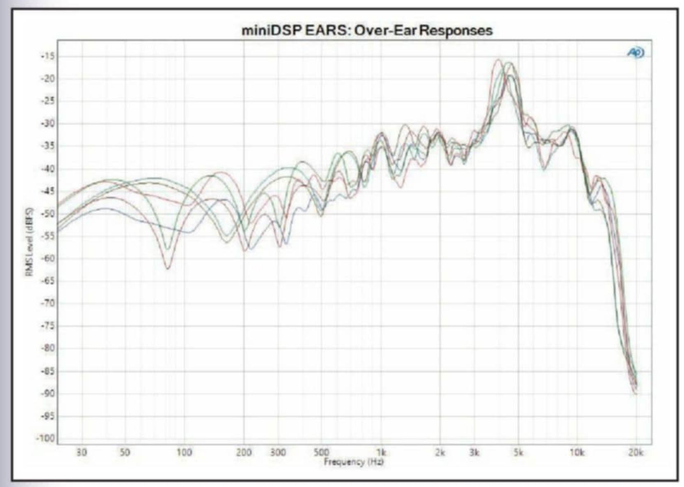


Figure 5: The variation in frequency response caused by removing and replacing the headsets between repeated measurements demonstrates the need to average over multiple repetitions.

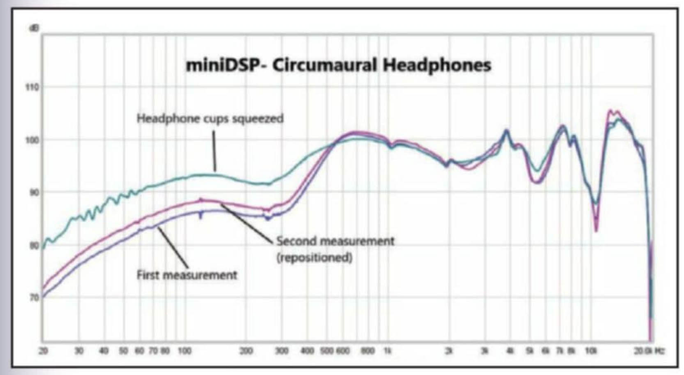


Figure 6: The measured bass response is greatly affected by headphone positioning and tension. Blue curve = first measurement. Magenta curve = measurement after repositioning the headphones. Green curve = measurement with pressure applied to the headphone cups.

The IEM EQ curves are shown in **Figure 3**. The lack of pinna/ear-cup cavity interaction removes a lot of the frequency response ripple. The peak from the simulated ear canal is still present, but lowered in frequency to about 2.5 kHz because of the IEM stopping up the end of the canal. There's also the same relatively steep treble roll-off seen in the HEQ curves. Of course, different insertion depths or tip geometries will change these features greatly, which adds another layer of measurement uncertainty.

Measurements and Use

As usual, my measurement setup consists of an Audio Precision APx525 with an APx1701 acoustic interface. The reference microphone was a PCB Piezotronics 376A33 phantom power 1/2" condenser mic. For REW measurements, the headphones were driven by a Scarlett 2i2 interface.

First, I measured the free-field response by positioning the EARS one meter from a mini-monitor, running log chirps, and gating the impulse response to remove echoes. The results were normalized to the mini-monitor's response measured by the reference mic at the same position. The measured response for the EARS jig is shown in **Figure 4**. We would expect some differences from the HEQ calibration curves because of the geometry of the EARS jig, but the major feature of the 4.5 kHz canal resonance is still evident. You can also see the same steep treble roll-off that appears in the HEQ and IEM EQ curves and that the bass response is relatively even.

Next, I checked the measurement repeatability with headphones placed and removed between each repeated measurement. This is shown in **Figure 5**, and emphasizes the need for multiple runs and power averaging when using this (or any other) headphone test jig to determine a frequency response. I would recommend at least six averaged measurements.

I then tried to measure some circumaural headphones that I've been using a lot recently. The frequency response acquired by REW is shown in the bottom (blue) curve in **Figure 6**. The response here has a shelf below 500 Hz, which suggests a very lightweight bass. This does not accord at all with my subjective perception, which is that of a deep and well-defined low-frequency end. Suspecting that this could be caused by a sealing issue, I repositioned the headphones and re-ran the frequency response, resulting in the second (magenta) curve, which was slightly better, but not much. Remembering the small width of the fixture, and hence, the low compression of the ear-pads to form a seal, I then tried re-running with extra compression. I hated the idea of rubber bands, so after hitting the "measure" button in REW, I quickly reached over and manually squeezed the cups more tightly against the jig using my hands. This produced considerably improvement in the bass response (the green curve), but still indicating inadequate bass. And I think I know the reason.

Photo 2 shows a close-up of the mechanical interface between the EARS and the headphone pads. The screws miniDSP uses to assemble its pinnae onto the stand are round head and not flush with the surface. So there's an inevitable leakage path for headphones (like these) that have relatively large diameter pads. It's an easy design flaw to correct, and replacing those screws with the same size screws having countersunk heads is an easy modification for EARS users to make without having to countersink the pinnae moldings; the rubber used is soft enough to deform under the screw head pressure to give a much flatter surface.

Figure 7 shows distortion vs. frequency for the circumaural headphones measured above. It's difficult to attribute the high bass distortion to the headphones alone given the seal issues previously noted. But the distortion in the mid-band was quite low (other than a peak at the 4.5 kHz resonance, almost certainly an artifact), so this jig should be able to give a user some insight into headphone distortion performance.

Photo 2: The screws used to fasten the rubber pinnae to the test fixture can compromise the seal of circumaural headphones. These should be changed to countersunk screws. (Photo courtesy of Cynthia Wenslow)

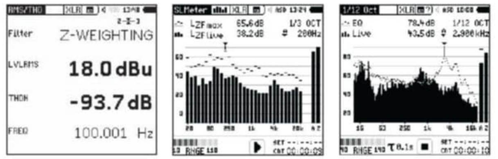
About the Author

Stuart Yaniger has been designing and building audio equipment for over half a century, and currently works as a technical director for a consumer products 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.

Ready for any Challenge!

Discover an Audio Analyzer you can trust for your application:

- Sound Level Meter class 1
- Level RMS / THD+N
- 1/1 or 1/3 octave RTA (up to 1/12 optional)



Distortion Analyzer | RealTime Analyzer | 1/12 octave Analyzer

NTi AUDIO | NTi Audio Inc.

americas@nti-audio.com | (503) 684 7050 | www.nti-audio.com/XL2

Fresh From The Bench

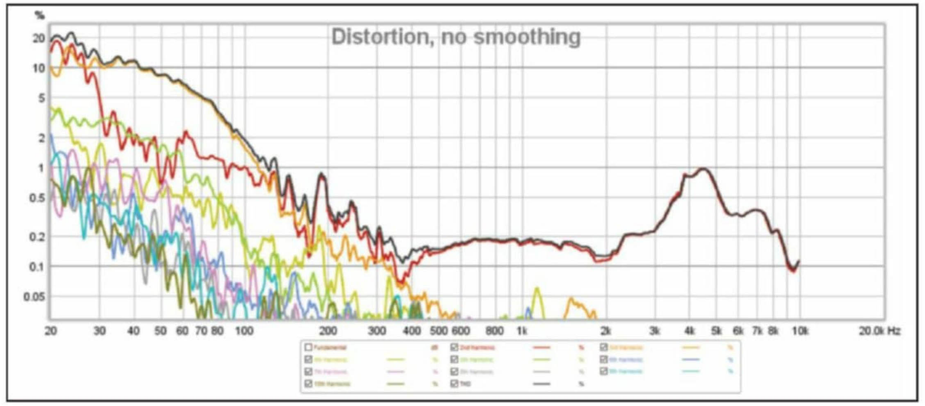


Figure 7: The distortion vs. frequency curves for the circumaural headphones used in testing indicate that the EARS's distortion floor is reasonably low, at least at mid frequencies.

In another typical user scenario, I investigated some gaming headphones I had on hand that never sounded quite right to me. **Figure 8** shows the frequency response using the RAW equalization, and the performance issue stands out starkly. One channel has a severe bass droop and the matching in the 1 to 2 kHz octave is off by several decibels. As a check, reversing the headphones caused the inter-channel anomalies to reverse as well, confirming that the headphones were the source of the problem.

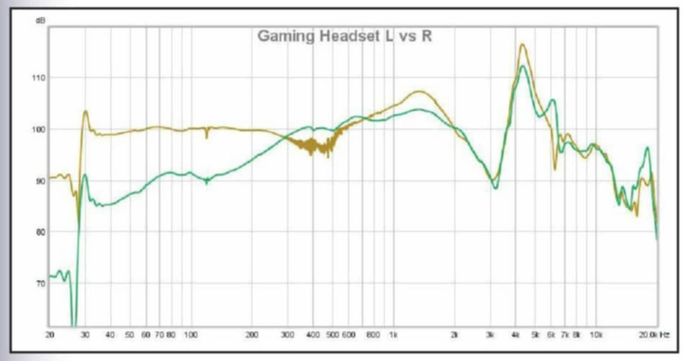


Figure 8: Using the EARS to test some gaming headphones that sounded bad immediately showed the problems.

Wrap-Up

A product like this ought to be approached with the right attitude. It's not a $15,000 laboratory fixture—there's a lot of interesting measurements that just can't be done (e.g., acoustic isolation, noise cancelling, waterfall response, spectrograms, etc.). The EARS frequency response does not correspond to current standards, though admittedly all standards suffer from the quid est veritas problem, since all of them will be guaranteed to be inaccurate for any actual individual user. If an ambitious user wants to seriously examine head-related transfer function (HRTF) issues and develop more customized EQ curves, a better option might be Ron Tipton's DIY HRTF binaural microphone head (see Resources).

For less sophisticated, but still useful calibrated measurements, the miniDSP EARS user is tied to one piece of software (albeit, a very good one). The mechanics and packaging, particularly the width between left and right pinnae, cause unnecessary variability and uncertainty. But $200 is almost impulse purchase money, and with a large installed user base, the EARS response could arguably be said to have the same validity, and perhaps even better portability, as the IEC or Harman curves. And of course, the user can modify the EQ files in any manner desired. The driverless installation and operation is a major user convenience. With a couple of minor design changes, miniDSP could make this even better without increasing the cost.

The bottom line is that the miniDSP EARS is not a research tool, but is meant to be an inexpensive source of fun and interest for headphone enthusiasts, giving them insight into the performance of their headphones and efficacy of modifications and EQ curves that alter frequency response and perhaps distortion. And that fun is delivered in spades.

Resources

M. Klasco and J. Martins, "Earphone and Headphone Test Instrumentation: A Comprehensive Roundup," *audioXpress*, April 2018.

M. Klasco, "Testing Headphones and Earphones: The GRAS Low Noise Ear Simulator," *audioXpress*, March 2016.

S. Olive and T. Welti, "Factors That Influence Listeners' Preferred Bass and Treble Balance in Headphones," Audio Engineering Society (AES) Convention Paper 9382, New York, 2015, www.aes.org/e-lib/browse.cfm?elib=17940.

R. Tipton, "A DIY HRTF Binaural Microphone Head," *audioXpress*, April 2018.

S. Yaniger, "Fresh From The Bench: The Audio Precision AECM206 Headphone Test Fixture," *audioXpress*, April 2018, www.audioxpress.com/article/fresh-from-the-bench-audio-precision-aecm206-headphone-test-fixture.

CRYSOUND

Focusing on acoustic test for 20 years, the industry leader in China

The CRY6100 series testers are the most widely used electroacoustic testers in the world.

Electroacoustic Measurement:

Speaker Headphone Microphone Amplifier Telephone

CRY6151B Electroacoustic analyzer:

- Ideal test solution for **Bluetooth, ANC** and **Lightning** headphone and microphone
- All in one design – soundcard, amplifier and mic power supply in one box
- Local service and distribution in North America
- Test your headphone and microphone simultaneously, no extra setup needed

CRYSOUND Measurement Microphones:

- Perfect for headphone and speaker test
- Use laser-welded titanium membranes
- Highly stable in harsh environments
- Manufactured in our ISO class 3 clean room
- Every capsule conditioned for 6 months to guarantee stability

Web: www.crysound.com

Add : No.31-1, Xianxing Rd, Xianlin Ave, Yuhang Dist. Hangzhou, Zhejiang 311122 CN

Tel: +86-571-88225198

E-mail: cry@crysound.com

Fresh From the Bench

Measuring Speech Intelligibility in Audio Systems

While the majority of consumer-focused advertising stresses the high audio quality of the latest tech gadgets, speech intelligibility is a critical performance measure for a wide array of systems from smart speakers to first responder radios. Joe Begin, of Audio Precision, examines the measurement methods currently in use for the objective—and repeatable—evaluation of speech intelligibility.

By
Joe Begin
(Audio Precision)

Speech intelligibility, the degree to which an average person can understand speech transmitted from a talker to a listener (listeners) via an audio system or channel, is different from speech quality. More specifically, speech intelligibility can be described as the percentage of words and other sounds uttered by the speaker, and sent via the transmission channel, that can be recognized by the listener.

For example, a Voice Over Internet Protocol (VOIP) system could have high bandwidth—and thus, high quality—but a tendency to drop entire words, making intelligibility poor; whereas, a Plain Old Telephone Service (POTS) has a lower bandwidth but no dropouts, thus providing lower quality but higher intelligibility.

Key Methods for Measuring Speech

One technique developed to measure speech intelligibility is the Speech Transmission Index (STI). Normal fluctuations in speech signals from the acoustic separation of sentences, words, and phonemes carry the most relevant information related to speech intelligibility. Modulation depth and modulation rate form a modulation spectrum. For clear speech, modulation rates range from 0.5 Hz to 16 Hz, with maximum modulation at around 3 Hz.

Another measurement technique is known as the Modified Rhyme Tests (MRT). In an MRT, trained listeners in a controlled laboratory environment are presented with sets of rhyming single-syllable words in a random order and asked to select the word that they heard from a list. Conclusions are drawn, based on the number of correct responses.

A speech algorithm, Articulation-Band Correlation Modified Rhyme Test (ABC-MRT), uses a form of Automatic Speech Recognition (ASR) to conduct an automated MRT that does not involve human subjects, thereby reducing test time, complexity, and expense. Published in 2013, the algorithm was initially limited to narrowband communication systems (250 Hz to 3.85 kHz bandwidth). In 2016, bandwidth was extended to cover all bandwidths up to full-band (20 Hz to 20 kHz), and an attention model was added. The enhanced method is called ABC-MRT16.

STI Methodology

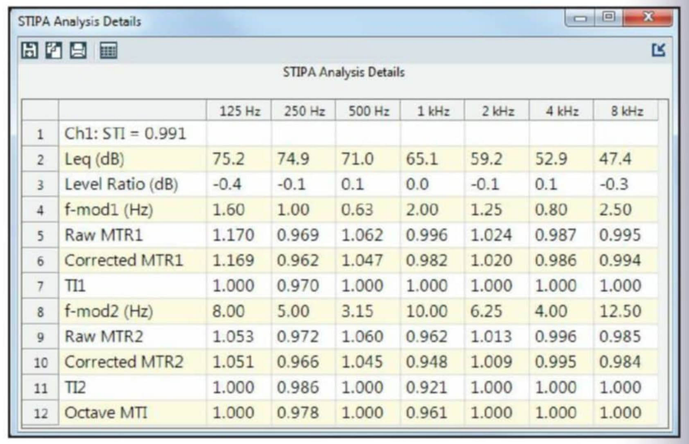
Generally, an STI signal is formed from seven octave-band noise signals, with the overall spectral shape matching the long-term average spectrum of speech. Each noise carrier is modulated with one or more of 14 low-frequency sine waves at modulation rates from 0.63 Hz to 12.5 Hz, spaced at 1/3-octave intervals. The STI signal passes through the transmission channel with the output signal captured at a listener position. From this output signal, the modulation transfer ratio (the reduction in modulation depth) is determined and used to calculate a single-number STI value.

The original implementation, known as full STI, requires 98 test signals to be applied sequentially; each measurement takes about 10 seconds for a total measurement time of approximately 15 minutes. A subsequent refinement known as STI for Public Address (STIPA) uses simplified, faster and more practical methods, making it popular.

In STIPA, each of the seven octave bands contains a pre-defined set of two modulation rates. These 14 combinations are generated simultaneously in one signal and processed in parallel, to reduce measurement time to between 10 and 20 seconds (see **Figure 1**).

Although the STI approach might seem similar to audio measurements such as Perceptual Evaluation of Speech Quality (PESQ) and Perceptual Objective Listening Quality Analysis (POLQA), both PESQ and POLQA were developed to measure speech quality over telephone transmission networks in ideal listening conditions with very little background noise. STI, however, is a metric for speech intelligibility more applicable to high-noise environments (e.g., a busy train station or an industrial complex) where intelligibility is primary, and quality is a secondary concern.

The STI uses the modulation transfer function (MTF), which is the ratio of modulation depth of the received signal to the modulation depth of the transmitted signal as a function of modulation rate. Distortions in the transmission channel, such as noise, reverberation, echoes and digital codecs, reduce the modulation depth, and distort speech intelligibility (see **Table 1** for more information).



		125 Hz	250 Hz	500 Hz	1 kHz	2 kHz	4 kHz	8 kHz
1	Ch1: STI = 0.991							
2	Leq (dB)	75.2	74.9	71.0	65.1	59.2	52.9	47.4
3	Level Ratio (dB)	-0.4	-0.1	0.1	0.0	-0.1	0.1	-0.3
4	f-mod1 (Hz)	1.60	1.00	0.63	2.00	1.25	0.80	2.50
5	Raw MTR1	1.170	0.969	1.062	0.996	1.024	0.987	0.995
6	Corrected MTR1	1.169	0.962	1.047	0.982	1.020	0.986	0.994
7	TI1	1.000	0.970	1.000	1.000	1.000	1.000	1.000
8	f-mod2 (Hz)	8.00	5.00	3.15	10.00	6.25	4.00	12.50
9	Raw MTR2	1.053	0.972	1.060	0.962	1.013	0.996	0.985
10	Corrected MTR2	1.051	0.966	1.045	0.948	1.009	0.995	0.984
11	TI2	1.000	0.986	1.000	0.921	1.000	1.000	1.000
12	Octave MTI	1.000	0.978	1.000	0.961	1.000	1.000	1.000

Figure 1: The STIPA analysis details results and definitions

ABC-MRT Evolution

For ABC-MRT measurement, a specialized ASR algorithm emulates the MRT methodology by recognizing keywords transmitted through a communication system. It also uses frequency bands, called Articulation Index (AI) bands. There are 17 AI bands for narrowband speech and 21 for full-band speech.

The algorithm uses time-frequency (T-F) representations of keywords in the MRT. The T-F pattern of an impaired speech signal is correlated with corresponding patterns of the six unimpaired options from a list of six rhyming words in multiple AI bands (see **Figure 2**).

Figure 2a is a spectrogram of the sentence "Please select the word went," as spoken by Female 1 in the Institute for Telecommunication Sciences (ITS) ABC-MRT speech database. Frequency is from 0 to 10 kHz and the time span is approximately two seconds. The signal level at a particular frequency and instant in time is represented by color ranging from blue (low) to red (high). The portion of the spectrogram containing the keyword "went" is highlighted with a red rectangle. Figure 2b shows the spectrograms of the six keywords from the list, plotted on the same time and frequency scales.

For each trial, a sentence containing the carrier phrase and keyword is generated and passed through the system, or Device Under Test (DUT). The signal is acquired from the output of the DUT and transformed to a T-F pattern using the same technique applied to the 1,200 isolated keyword recordings in the database.

The keyword is located within the T-F pattern by cross-correlating it in two of the AI bands with the T-F pattern of the original keyword in the template file. The portion of the DUT's T-F pattern containing the keyword is extracted and matched with one of the six rhyming words in the AI bands of interest. Eliminating negative results leaves 17 narrowband and 21 full-band AI band correlations for each candidate keyword. Keywords in the 16 AI bands with the highest correlation values are selected and compared to the known correct keyword. The average number of correct keyword selections over the 16 AI bands provides the success rate. The average success rate across all the trials is corrected to account for the effect of guessing in MRT tests, and provides the ABC-MRT Intelligibility Score.

Item	Description
Leq (dB)	The overall RMS level for each octave band (referred to as the equivalent sound level for acoustic inputs)
Level Ratio (dB)	Indicates the spectral shape of the received signal relative to the signal source, normalized to the 1 kHz octave band
f-mod* (Hz)	The modulation frequency
Raw MTR*	The calculated raw modulation transfer ratio
Corrected MTR*	The MTR after correction for auditory masking and reception threshold
TI*	The transmission index
Octave MTI	The average of TI1 and TI2
*These items have two values for each octave band.	

Table 1: The STIPA definitions

TEST & MEASUREMENT FOCUS

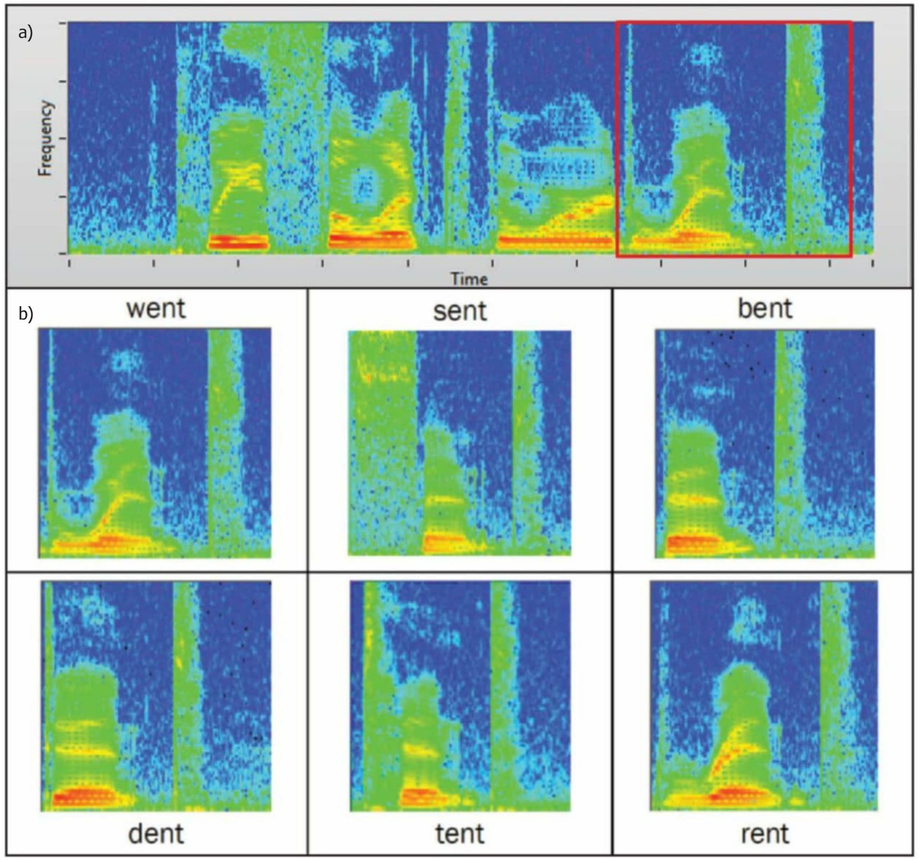


Figure 2: The test sentence (a) and spectrograms of the six key words on the MRT list (b)

An ITS evaluation of ABC-MRT and ABC-MRT16 found extremely high correlation with subjective MRT and low estimation error. ABC-MRT16 performed significantly better than ABC-MRT for the 139 narrowband conditions, indicating the importance of the attention model.

Target Applications

STI is used in high-noise environments where intelligibility is paramount. Examples include Public Address (PA) systems, aircraft Voice Announce (VA) and emergency communication systems, in-vehicle communication systems, auditorium systems, and assistive hearing systems.

It should not be used for measuring systems with transmission channels that contain vocoders (i.e., codecs that operate only on speech elements), but may be used for digital codecs that operate on the entire signal. STI is also less likely to work for systems with aggressive noise-suppression algorithms, because the STI signal is essentially noise, and it would likely be suppressed by the algorithm.

The ABC-MRT algorithm works with voice codecs and noise suppression systems, and can be used for any of the standard speech bandwidths.

Having ABC-MRT measurement capability in an audio analyzer complements its traditional audio test functionality. Depending upon the analyzer, it can also provide access to multiple audio interfaces (analog, acoustic, AES3-SP/DIF, digital serial, Bluetooth, PDM, HDMI and ASIO), a built-in test sequencer and reporting engine, test limits, multiple channels, and a wide variety of specialized audio measurements.

About the Author

Joe Begin, PE (Acoustical Engineering), a graduate of McGill University (B.Sc.) and the University of Canterbury (M.Sc.), more than 35 years' experience in test and measurement. As Director of Applications and Technical Support at Audio Precision, he is involved in product management, audio and electroacoustic test applications engineering, and technical support.

NFPA Test Methods

The US National Fire Protection Association (NFPA) 1981 standard covers self-contained breathing apparatuses (SCBA) for emergency services. It requires a minimum STI of 0.55 for non-electronic systems, and 0.60 for supplementary systems. The test methods for both non-electronic and supplementary systems are similar. Both require a hemi-anechoic chamber and a Head and Torso Simulator (HATS) with mouth simulator (see **Figure 1**).

NFPA test methods require measuring STI at a test microphone that is 1.5 m in front of the artificial mouth, while simultaneously generating pink noise from a separate speaker located below the test microphone. (Pink noise power per hertz decreases as the frequency increases; white noise has an equal power per hertz through all frequencies.)

NFPA 1981 requires that the mouth simulator is equalized to a specific 1/3-octave spectral shape at the mouth reference point of the mouth simulator. For convenience when adjusting the EQ by trial and error, the output EQ curve in the audio analyzer software can be specified at standard 1/3-octave center frequencies.

ABC-MRT and STI are available as software plug-ins for Audio Precision APx500 Series audio analyzers. The plug-ins can be downloaded and installed to implement the ABC-MRT16 algorithm and STI capabilities, respectively.

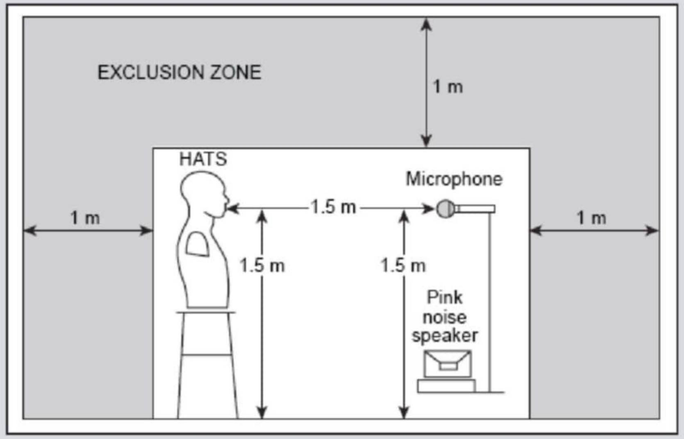


Figure 1: Test setup with anechoic chamber

Attenuators for Measurement

In this article, our author discusses external attenuators and compares the impact different attenuators' thermal noise will have on measurements of distortion. At low test voltages, the added thermal noise of the attenuator is of primary concern. At high test voltages, the heat dissipation in the external attenuator is of primary concern. For this reason, it can be handy to have multiple attenuators of similar attenuation but with different impedances on hand.

By
Jonathan Novick
(United States)

External attenuators are an effective and cost effective way to measure signals above the maximum input voltage of an audio analyzer. However, commercial off-the-shelf attenuators are usually not a good choice in this application as they are typically designed for low-voltage use only. There are several trade-offs to consider when designing attenuators for higher voltages, including topology, power handling and voltage ratings, resistor choices, and added measurement noise.

Attenuation values are typically given as a ratio expressed in decibels (dB). **Table 1** represents the ratio of the input voltage to the output voltage, which can be determined using Equation 1:

$$Attenuation = -20Log\left(\frac{V_{output}}{V_{input}}\right)$$

Attenuator Topology

Attenuators can be constructed using many topologies, each offering different attributes to suit different needs. For audio noise measurements, an attenuator with a low output impedance is desirable. This makes the L-pad and U-pad topologies ideal choices for power amplifier measurements. These attenuators are asymmetrical in nature. In other words, they have a specific input and output direction. Symmetric topologies, such as T-pads and Pi-pads and their variants are primarily used in RF communications, where impedance matching and bi-direction signal flow are of greater importance than handling high voltages and minimizing added noise.

Attenuation	6 dB	10 dB	12 dB	18 dB	20 dB	24 dB	30 dB	40 dB
Voltage Ratio	2:1	3.2:1	4:1	8:1	10:1	16:1	32:1	100:1

Table 1: Attenuation and voltage ratios

An L-pad is a simple resistive divider network consisting of a series resistor attached between the signal source and the audio analyzer and a shunt resistor across the audio analyzer input. The output voltage of an attenuator can be calculated using Ohm's law:

$$\text{L-pad output voltage} = Vin \times \frac{Rshunt}{Rshunt + Rseries}$$

The U-pad is simply two identical L-pads placed back-to-back in a mirrored configuration; each side attenuating one side of the balanced signal (see **Figure 1**). The two shunt resistors of a U-pad can be combined into a single resistor but keeping them separate provides a ground reference that can be useful to control common mode issues with floating test systems. The ground point also enables the U-pad to be used in both balanced and unbalanced configurations. The U-pad resistor values are shown here as half the L-pad values. This is so the same calculations used here can be applied to either case.

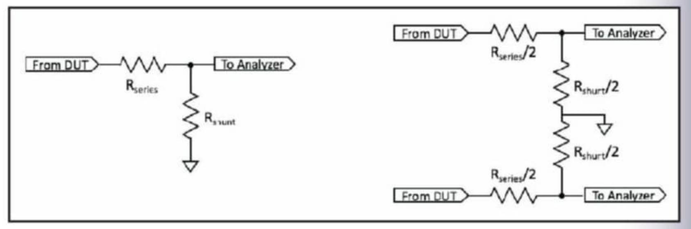


Figure 1: The U-pad is simply two identical L-pads placed back-to-back in a mirrored configuration—each side attenuating one side of the balanced signal.

The Test Load as an Attenuator

Power amplifiers are often tested with resistive loads. These loads can also be configured as L-pads or U-pads using multiple resistors. This is the recommended way to attenuate an amplifier output for measurement.

An 8-Ω load can be made with eight 1 Ω resistors in series. Each node between the resistors provides a differing attenuation value. The center node (between the fourth and the fifth resistors) provides 6 dB attenuation. The node between the sixth and the seventh resistors provides 12 dB attenuation and the node between the seventh and the eighth resistors is attenuated 18 dB. Amplifiers with push-pull outputs should be tested using a U-pad with each side having eight 0.5 Ω resistors. The total resistance is 8 Ω and there is still access to 6 dB, 12 dB, and 18 dB attenuation points (see **Figure 2**).

Testing Without a Load

Testing amplifiers without a load requires more careful consideration of the attenuator's design. Most commercially available attenuators have an impedance of just a few hundred ohms and are comprised of 1/8 W or 1/4 W resistors. At high voltages, the power rating of these resistors could be exceeded, causing them to overheat. The heat dissipation can be reduced by using higher resistance values. However, higher resistance values add excess noise that may impact measurement accuracy. The remainder of this article discusses the design of higher impedance attenuators.

Choosing a Resistor Type

There is no such thing as a perfect resistor. There are many trade-offs to be considered for both electrical and physical properties as well as cost.

Thermal (Johnson) noise is the voltage noise produced by a resistor that is proportional to the square root of the resistance value. The higher the resistance, the higher the noise voltage. The RMS level of this noise is defined as:

$$V = \sqrt{4 \times R \times k \times T \times BW}$$

where, R = Resistance value
k = Boltzman's constant, 1.38 x 10^{-23}
T = Operating temperature in degrees Kelvin
BW = Bandwidth in Hertz (Hz)

Current noise, sometimes referred to as 1/f noise, is caused by the electron flow through the resistive material and varies with the voltage across the resistor. Carbon-film, thick-film and carbon-composition resistors have a higher noise index than other types and should be avoided for measurement purposes.

The temperature coefficient indicates the sensitivity of a resistor's value to temperature changes. Heating differences between the series and shunt resistors in an L-pad can produce changes in attenuation that can throw off measurements. This can be mitigated by using resistors with lower temperature coefficients (≤50 ppm), using higher wattage resistors in the series element, or constructing the series element from multiple resistors. Small surface-mount resistors with low thermal mass can exhibit nontrivial changes in temperature within a single cycle of a low-frequency signal and are not recommended unless used in large arrays to minimize heating.

Inductance in attenuators should generally be minimized. Wire-wound resistors can handle tremendous heat loads, but they also exhibit higher inductance than other resistor types. The actual inductance is a function of the construction method but vary from 0.1 to 400 µH depending on wattage and value. The maximum acceptable inductance should be considered for the specific device being tested.

Magnetic susceptibility occurs because wire wound resistors can both emit and pick up magnet fields. If used, they should be placed away from sensitive circuits or strong magnetic sources such as power transformers.

Power ratings are an indication of how hot resistors can operate without failing. However, operating the resistor well below its maximum rated power insures the operating temperature stays closer to the ambient temperature.

Cost and availability can vary significantly with resistor type. Metal-foil resistors approach

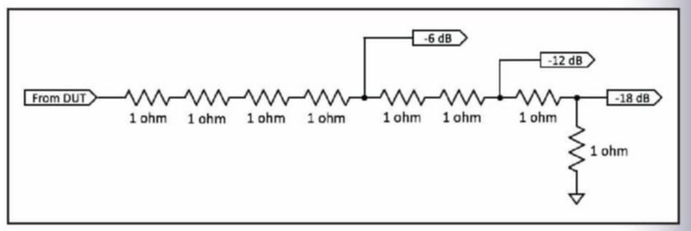


Figure 2: Amplifiers with push-pull outputs should be tested using a U-pad with each side having eight 0.5 Ω resistors. The total resistance is still 8 Ω and there is still access to 6 dB, 12 dB, and 18 dB attenuation points.

	Resistor Choices		Actual Attenuation	Thevenin Resistance	Thermal Noise
	Series	Shunt			
Case 1	100 Ω	11 Ω	-20.08 dB	9.1 Ω	0.057 µV/-145 dBv
Case 2	1,000 Ω	110 Ω	-20.08 dB	99 Ω	0.18 µV/-135 dBv
Case 3	5,000 Ω	560 Ω	-19.94 dB	504 Ω	0.41 µV/-128 dBv
Case 4	10,000 Ω	1,100 Ω	-20.08 dB	991 Ω	0.57 µV/-125 dBv
Case 5	20,000 Ω	2,200 Ω	-20.08 dB	1,982 Ω	0.81 µV/-122 dBv
Case 6	47,000 Ω	5,100 Ω	-20.19 dB	4,601 Ω	1.23 µV/-118 dBv

Table 2: L-pad resistor combinations for 20 dB

the performance of an ideal resistor in many ways. However, they can cost up to 100 times more than other resistor types. Availability can also be a problem. Leaded thin-film resistors offer an inexpensive, readily available compromise for constructing attenuators. Thin-film resistors rated at 0.6 W, 1% tolerance and a 50 ppm/°C temperature coefficient cost approximately 15 to 30 cents each in individual quantities. These can be wired in series and parallel combinations to handle higher power dissipation at a reasonable cost.

Impact of Noise on a Measurement

The thermal noise from an attenuator can impact the readings of certain measurements. This will be most evident in noise-related measurements such as signal-to-noise ratio (SNR) or total harmonic distortion + noise (THD+N), particularly at lower signal levels.

The thermal noise of an L-pad is determined as a function of its Thevenin equivalent resistance. Assuming the amplifier under test has a negligible output impedance, the Thevenin equivalent circuit is simply the parallel combination of the series and shunt resistors of the attenuator. **Table 2** shows some common resistor combinations that can produce ~20 dB attenuation as well as the thermal noise produced by that combination over a 20 kHz bandwidth at 27°C.

The significance of this thermal noise depends on the needed measurement accuracy, the signal level of the test and the residual noise and distortion of the product being tested. Suppose the desired measurement is distortion. This is defined by the equation for THD+N:

$$\text{THD+N (in dB)} = 20 \text{ Log}\left(\frac{\text{residual noise and distortion}}{\text{total signal}}\right)$$

Putting This to Work

The following two examples demonstrate the types of trade-offs needed when designing an attenuator. I used AverLAB, the new audio analyzer from Avermetrics, to measure high-powered amplifiers. To keep AverLAB small, its maximum input was limited to 15.5 Vrms (balanced) and 7.7 Vrms (unbalanced). Without an attenuator, it can test up to 30 W into an 8 Ω load at the balanced inputs or 7.5 W at the unbalanced inputs. For larger amplifiers, 20 dB of external attenuation increases those limits to 3000 W for balanced and 750 W for unbalanced connections. AverLAB has provisions to offset measurements for external attenuation which makes making these measurements easy.

Measuring a 200 W Amplifier

This hypothetical amplifier is rated at 200 W into 8 Ω with a THD+N of -100 dB (0.001%). The amplifier output is 40 V at the rated power and solving Equation 4 shows that there is 400 µV of residual noise and distortion. A 20 dB L-pad will reduce the output signal to 4 V, which is well within AverLAB's input range, and will also reduce the residual to just 40 µV. The analyzer will still show a THD+N reading of -100 dB unless the attenuator has added too much noise of its own. The question is how much of this added attenuator noise can be tolerated.

If the maximum acceptable measurement error due to added noise is 0.1 dB, the distortion reading must be ≤-99.9 dB. Equation 2 is solved for -99.9 dB to show the total residual seen by the analyzer cannot exceed 40.46 µV. Given that the amplifier is producing 40 µV of this residual, the RMS allowable noise from the attenuator can be determined. Since noise sources are uncorrelated RMS summation, the equation:

$$\text{Total noise}^2 = \text{amplifier residual}^2 + \text{attenuator noise}^2$$

is used to determine the allowable attenuator noise. This shows that the maximum allowable attenuator noise for 0.1 dB accuracy is 6.1 µV. Any of the attenuators shown in Table 2 are thus sufficient to measure the amplifier. However, this is only true for measurements at 40 V.

Amplifiers are often tested by sweeping from a low amplitude to maximum power. An audio analyzer would normally auto-range its input as the signal level changed to ensure maximum measurement accuracy. However, a fixed external attenuator cannot be similarly switched in and out as needed during an automated level sweep. Therefore, the noise of the attenuator must be considered at the lowest level being tested.

At 1 W (2.83 V into 8 Ω) from the amplifier, the analyzer sees only 0.283 V with the 20 dB attenuator in place. The same equations as above are used again to find that allowable attenuator noise for 0.1 dB accuracy is now just 0.43 µV. Table 2 shows that Case 3 offers the highest impedance resistor combination that would satisfy that accuracy target.

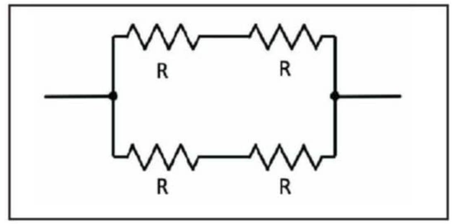


Figure 3: Any resistor of value R may also be constructed from multiple elements to further reduce heating.

Measuring a 1,500 Watt Amplifier

The previous example was only for a 200 W amplifier, such as those found in home sound systems. Professional live-sound amplifiers frequently have outputs in the kilowatt range. A 1500 W amplifier produces approximately 141 V into an 8 Ω load. A 20-dB attenuator would reduce this voltage to ~14 Vs (within AverLAB's balanced input range) but heat dissipation within the attenuator becomes a significant factor to consider. **Table 3** shows the same resistor combinations as Table 1 along with the power dissipated by each resistor at both 200 W and 1,500 W.

It is easy to see that significant power can be dissipated in an attenuator at 141 V. Case 6 is the only attenuator that could be made using 0.6 W

	Resistor Choices		Dissipation at 40 V		Dissipation at 141 V	
	Series	Shunt	Series	Shunt	Series	Shunt
Case 1	100 Ω	10 Ω	13.22 W	1.322 W	165 W	16.5 W
Case 2	1,000 Ω	110 Ω	1.30 W	0.143 W	16.2 W	1.8 W
Case 3	5,000 Ω	560 Ω	0.259 W	0.029 W	3.23 W	0.36 W
Case 4	10,000 Ω	1,100 Ω	0.130 W	0.014 W	1.62 W	0.18 W
Case 5	20,000 Ω	2,200 Ω	0.065 W	0.007 W	0.81 W	0.09 W
Case 6	47,000 Ω	5,100 Ω	0.028 W	0.003 W	0.35 W	0.04 W

Table 3: L-pad power dissipation at 40 V and 141 V

More Helpful Information

The following calculations are helpful in the design and use of external attenuators:

- Equations for design: Determine the ideal resistor values based on test level and desired accuracy
- Equations for use: Determine the measurement accuracy based on actual resistor values, test level and distortion reading

These equations are simply variations of equations shown earlier solved for different variables. These can be conveniently entered into a spreadsheet as a handy reference. Such a spreadsheet is available for download at www.avermetrics.com.

Project Files

To download a spreadsheet with all the equations to calculate the attenuators, visit www.audioxpress.com/page/audioXpress-Supplementary-Material.html

Variable	Equations for Design
Vn_{DUT}	$Vn_{DUT} = Level \times 10^{Dist/20}$
Vn_{Att}	$Vn_{Att} = Vn_{DUT} \times 10^{Atten/20}$
Vn_{meas}	$Vn_{meas} = Vn_{Att} \times 10^{Acc/20}$
Vn_{Atten}	$Vn_{Atten} = \sqrt{Vn_{meas}^{\ 2} - Vn_{Att}^{\ 2}}$
R_{Thev}	$R_{Thev} = Vn_{Atten}^{\ 2} / \left(4 \times Temp \times BW \times 1.38E^{-23}\right)$
R_{series}	$R_{series} = R_{Thev} / 10^{Atten/20}$
R_{shunt}	$R_{shunt} = R_{Thev} / \left(1 - 10^{Atten/20}\right)$
P_{series}	$P_{series} = \left(\frac{Level}{R_{series} + R_{shunt}}\right)^2 \times R_{series}$
P_{shunt}	$P_{shunt} = \left(\frac{Level}{R_{series} + R_{shunt}}\right)^2 \times R_{shunt}$
Equations for Use	
$Atten_{Actual}$	$Atten_{Actual} = 20 \times Log\left(R_{shActual} / R_{shActual} + R_{seActual}\right)$
$R_{ThevActual}$	$R_{ThevActual} = \left(\frac{1}{1/R_{seActual} + 1/R_{shActual}}\right)$
Vn_{Actual}	$Vn_{Actual} = \sqrt{4 \times Temp \times BW \times 1.38E^{-23} \times R_{ThevActual}}$
Vn_{Meas}	$Vn_{Meas} = LevelActual \times 10^{Dist_{Meas}/20}$
$Vn_{DUTActual}$	$Vn_{DUTActual} = \sqrt{Vn_{Meas}^{\ 2} - Vn_{Actual}^{\ 2}}$
$Dist_{Actual}$	$Dist_{Actual} = 20 \times Log^{\left(Vn_{DUTActual} / Level_{Actual}\right)}$
Acc_{Actual}	$Acc_{Actual} = Dist_{Meas} - Dist_{Actual}$

Variable	Description	Units
Level	The output level of the device under test (DUT)	volts (V)
Distortion	The rated distortion level of the DUT	decibels (dB)
Temperature	Temperature in degrees Kelvin	K
Bandwidth	Bandwidth of measurement	Hertz (Hz)
Vn_{DUT}	The residual noise and distortion of the DUT at Level and Distortion	volts (V)
Attenuation	The value of the external attenuation	decibels (dB)
k	Voltage divider ratio of a given attenuator	k:1
Vn_{post}	Value of N_{DUT} following the external attenuator	volts (V)
Accuracy	Tolerable measurement of accuracy	decibels (dB)
Vn_{meas}	Residual seen by the analyzer at limit of accuracy specification	volts (V)
Vn_{Atten}	Noise contribution of the attenuator at limit of accuracy	volts (V)
R_{Thev}	Highest Thevenin equivalent resistance for an L-pad attenuator that will meet the accuracy specification at the rated distortion and level (at 27°C and 20 kHz BW)	ohms (Ω)
R_{series}	Highest value of a series resistor for an L-pad that meets accuracy specification	ohms (Ω)
R_{shunt}	Highest value of a shunt resistor for an L-pad that meets accuracy specification	ohms (Ω)
P_{series}	Power dissipated in the series resistor at Level	ohms (Ω)
P_{shunt}	Power dissipated in the shunt resistor at Level	ohms (Ω)
$R_{shActual}$	Chosen shunt resistor	ohms (Ω)
$R_{seActual}$	Chosen series resistor	ohms (Ω)
$Atten_{Actual}$	Precise attenuation given the chosen resistors	decibels (dB)
$R_{ThevActual}$	Thevenin equivalent resistance of the actual attenuator	ohms (Ω)
Vn_{Actual}	Noise produced by the actual attenuator	volts (V)
$Dist_{Meas}$	Measured distortion reported by the audio analyzer	decibels (dB)
Vn_{Meas}	Residual noise and distortion measured by the analyzer	volts (V)
$Level_{Actual}$	Signal level seen by the audio analyzer	volts (V)
$Vn_{DUTActuual}$	Residual noise from DUT (without attenuator noise)	volts (V)
$Dist_{Actual}$	Actual distortion of the DUT	decibels (dB
Acc_{Actual}	Accuracy of the measured distortion based on the test level and attenuator used	decibels (dB)

resistors. However, Table 2 shows that the noise will be 1.23 µV.

At full power, the analyzer will be seeing a 14.1 V signal and the amplifier's residual would be 0.14 µV for a distortion of -100 dB. The 1.23 µV of additional noise from the attenuator would not cause the reading to change significantly. In fact, it would take 21.6 µV of noise to change the measurement by 0.1 dB. However, at 1 W, it takes only 0.43 µV of noise to change the measurement by 0.1 dB. With 1.23 µV of attenuator noise the new THD+N reading would be -99.25 dB, a difference of 0.75 dB.

The series resistor in Case 3 could be built using multiple 0.6 W resistors to handle the heat load. For instance, eight 10 kΩ resistors could be wired as four series-pairs in parallel to produce the 5000 Ω series resistance. This brings the dissipation in each resistor to ~0.4 W. However, at 0.4 W, these resistors will see almost a 70°C temperature rise, which may not be tolerable.

The resistor combination in Case 4 changes the measurement reading by less than a 0.2 dB and Case 5 would cause less than a 0.4 dB. These other solutions make it easier to handle the heat load in the attenuator.

About the Author

Jonathan Novick is an electrical engineer with more than 25 years of test and measurement experience, the last 15 exclusively in audio. He most recently served as VP of Sales and Marketing for Avermetrics, LLC. Prior to that he was a Director of Sales for Audio Precision and a Senior Product Manager at Agilent Technologies (now Keysight). He is also a former vice president and governor of the Audio Engineering Society. Jonathan co-chairs the R3WG2 audio standards committee of the Consumer Technology Associations and contributes on other working groups as well.

The shunt resistor dominates the noise produced by the L-pad. In Case 3, the temperature of a 0.6 W thin-film shunt resistor will rise almost 60°C whereas in Case 4 it will rise less than 20°C. Any resistor of value R may also be constructed from multiple elements to further reduce heating (see **Figure 3**).

Conclusions

This article discusses the impact an attenuator's thermal noise will have on distortion measurements. At low test voltages, the added thermal noise of the attenuator is of primary concern. At high test voltages, the heat dissipation in the external attenuator is of primary concern. For this reason, it would be handy to have multiple attenuators of similar attenuation but with different impedances on hand. The use of 0.6 W 1% thin-film resistors makes this a very practical and cost-effective solution.

Loudspeaker Efficiency

The Truth Behind the Numbers

Image courtesy of AER Loudspeakers

By
Geoff Hill
(United Kingdom)

Having high-efficiency loudspeakers is a good idea. So, why do we not see more high-efficiency designs in common use? This article explores the problem and reminds us that, quite simply, we have forgotten that it is the total system efficiency that matters.

As most of us know, one of the major driving forces of our society today is efficiently or effectively using resources, whether this be the number of miles or kilometers traveled for a given fuel/charge or the amount of energy used for everyday tasks. There are massive efforts put into reducing the power wasted throughout society. This is especially so in the design of amplifiers and electronics of all types, when so many of our devices are very low powered. However, common sense seems to have been thrown out when it comes to the efficiency of loudspeakers! I would ask: Does it really make sense to match 98% efficient amplifiers with 0.007% efficient loudspeakers? I would like to give you an outline of the scale of the problem, the first recognized steam engine by Thomas Newcomen achieved 0.5% efficiency, while James Watts' improvements increased this 2.5%. A modern steam plant is capable of 10% plus. While turbines regularly achieve 40% or above. Let us contrast this with an "average" domestic loudspeaker and most are below 0.5% efficiency!

It is my contention that this is intolerable and as an industry we should be ashamed of ourselves. Also, we have forgotten that it is the total system efficiency that matters! There is absolutely no point in having 98% efficient Class D amplifiers if the loudspeaker is 0.007% efficient as was pointed out by Wolfgang Klippel at the 2018 Audio Engineering Society (AES) convention in Milan, Italy. Unfortunately, this is typical in a modern mobile phone!

Worse when you talk to someone in that industry many of the words discuss doing more with less or smaller (inefficient) loudspeakers and increasing the power consumption. With a 0.007% efficiency that inevitably means that 99.993% of it is wasted energy! I was curious as to how efficient people thought loudspeakers were. So after a recent AES meeting in London, I asked a couple of people how efficient a phone's loudspeaker was. I was surprised when mostly they said it was more than 60% to 70%! These were not stupid people, and they work with speaker setups all the time! So, either the truth has been very carefully covered up or many people really do not understand what is going on. It is my contention that high-sensitivity or high-efficiency loudspeakers are a good idea. And not just because they are more efficient. Other reasons include:

- High-efficiency loudspeakers sound better because they need less power to reach the same loudness as less efficient loudspeakers. Quite simply neither the loudspeakers nor the amplifiers have to work so hard to produce sound.
- The lower power requirements mean that high-efficiency loudspeakers do not heat up as much as other lower-efficiency speakers.
- Low efficiency tends to be a consequence of high mass, small diaphragm area, low shove (Bl^2/Re) and or poor coupling.
- Higher efficiency means that loudspeakers do not move as much when reproducing music.
- The lower movement of high-efficiency loudspeakers means that the distortion is often dramatically reduced.
- High-efficiency loudspeakers can be used with lower power amplifiers, even a headphone amplifier may be suitable.

- High-efficiency loudspeakers are inherently more dynamic and responsive.
- High-efficiency loudspeakers can produce higher sound pressure levels.

In summary, high efficiency tends to lend itself to an ease or flow that sounds natural and unforced.

Conventional Designs

Okay, I believe we have established at least some of the reasons why high-efficiency loudspeakers are a good idea. This being the case why do we not see more high-efficiency designs in common use? Notice I say common use—as in some applications this is not the case.

Fundamentally, almost all conventional loudspeakers (99.5%) are moving coil transducers (see **Photo 1**), and they are mostly operated under the constant gain bandwidth product outlined by Edward Kellogg and Chester Rice in their 1925 patent. Under this patent, a loudspeaker's low-frequency performance is directly tied to the area of the loudspeaker's diaphragm area, the volume velocity, and the loudspeaker's efficiency or sensitivity, such that over a broad range of frequencies a loudspeaker tends to produce a nominally flat output level vs. frequency.

The result has tended to be a general lowering of efficiency in the pursuit of greater low-frequency bandwidth—given that historically amplifier power was initially scarce but increasingly became readily more available, lowering of efficiency has not been a big problem. But I ask, should we be happy with the 0.1% to 0.5% efficiency that most loudspeakers achieve? I would say no. Our paradigms need serious attention.

The questions are: What do we need to do to significantly increase efficiency and why? Let us look at the raw numbers as we know the efficiency of a loudspeaker does strongly depend on the loading conditions whether, corner loading, desk or half space/baffle or full spherical changes the numbers. However, let us assume the following:

112 dB = 100%
102 dB = 10%
92 dB = 1%
82 dB = 0.1%
72 dB = 0.01%
70 dB = 0.007%

The question is what are we trying to achieve? A better question is how much acoustic power is required? Do we need to produce 112 dB at 10 m at 20 Hz for an outside music concert? Or do we need to use 10 W for a movie audience as was

Photo 1: A typical 10 mm transducer

demonstrated by the 1934 Western Electric Horn System by Silbatone (see **Photo 2**) at the High-End Show in Munich in 2017?

If so, then we will need a completely different solution if we need to produce 80 dB at 0.5 m on a desk at 50 Hz for a home-based computer sound system. Let us bear in mind that if an orchestra produces a peak of say 10 or 20 acoustic watts, the answer is: the power required is not as might be thought as reproducing an orchestra in a domestic setting would be overwhelming, if it were even physically possible. And it is also true that 92 dB is very loud, but does that mean that we should deliberately waste energy?

The big problem that I see is that there has been a gradual concentration of loudspeaker transducer designs from relatively few companies, who now produce most of the loudspeaker drive units, and those drive units tend to be built to a standard recipe. So, these companies tend to produce transducers that meet the average requirement of the market. Thus, the nominal specification has changed over the years to mean that most Hi-Fi loudspeakers are

Photo 2: This is the Western Electric 1934 System shown by Silbatone at the Munich High End Show in 2017.

between 82 dB and 92d B or between 0.1% and 1% efficient, while mobile phone loudspeakers are often lower than 0.01% efficient.

So what is the point in chasing small gains in amplifiers when we are wasting so much efficiency in the loudspeakers themselves and why is this such a problem? Well, if we want to increase the efficiency of loudspeakers, which I think we should, then we need to fundamentally change how we go about the design of the underlying transducers. We know we can gain efficiency by increasing area or shove, or decreasing mass, or increasing coupling, which is what horns do. Everything else is window dressing! Smoke and mirrors, if you prefer. Let us remember that fundamentally there are three parts to the efficiency equation:

$$SPL\ (dB) = 112.1 + 10 * \log^{10} *(p0/ (2 * \pi * c) * ((B * l)^2)/Re *Sd^2/Mms^2$$

$$\text{The air part} = p0/ (2 * \pi * c)$$

We cannot do much about the first part, so we will have to concentrate on the air load and the second and third parts:

$$\text{The Electromagnetic} = ((B * l)^2)/Re \text{ The Mechanical} = Sd^2/Mms^2$$

Also, please note that many of these are "Squared" terms and dramatically affect the sensitivity or efficiency. Altering any of these parameters inevitably also changes the response shape especially at lower frequencies. Neville Thiele and Richard Small taught us this nearly 60 years ago. But are the Thiele-Small alignments the only way to design today's loudspeakers? Do we have to design for flat maximal bandwidth? And do we have our design parameters right? We have more amplifier power available than ever. The 1, 2, 3, or 4 kW amplifiers are common, but this tends to lead toward increased power dissipation and lower efficiency, which in my opinion is a bad idea. DSP is readily available, but we are still stuck in the old paradigms. The question I believe we should answer is: Why do we have our current paradigms?

Changing the Paradigm

Back in 1936, F.H. Brittain who was very influential early on, provided a ranking of relative importance of various factors vs. sound quality for loudspeakers. They included: frequency response (amplitude response), harmonic distortion, spurious noises and intermodulation distortion, frequency shift (fm distortion), dynamic range compression (power compression), transient distortion, phase distortion, group delay, electroacoustic efficiency, power handling capacity, and constancy of performance.

I believe it is this list, albeit with a few minor changes, that has constrained the design choices. Unfortunately, we can see this has not changed much in 80 years. Now many would argue that this proves that it set the right direction for loudspeaker development. While I would argue that electroacoustic efficiency needs to be much higher up this list certainly in the top three and not the bottom three. However, is this necessarily so? Would Brittain, yet alone Rice and Kellogg, stick with the technique used to extend and control bandwidth given our current technologies if they were working today? Maybe but maybe not. The important point is that we have different techniques and technologies available now that they did not.

Making High-Efficiency Designs

So, I would propose a different solution, we can produce high-efficiency transducers, and advances in materials are allowing lighter and stronger diaphragms. However, we need to scrap the flat response or gain vs. bandwidth trade-off and allow DSP or equalization and amplification to take its rightful place. A little reflection shows us that just lowering mass will not suffice to meet all requirements. One of the quandaries and conundrums of loudspeakers is undoubtedly the terrible variation in fidelity between different speakers. Poor control of frequency response and resonances or coloration has certainly played a part in this.

Supposedly passive crossovers are inherently wasteful and tend to suck and dissipate energy, but is this true? While active or DSP technology readily enables us to produce conventional filters, and to a more limited extent, correct response variations—the main problem is one of stability of response variation. It is easy to boost or cut a signal; however, if the "Problem" is unstable to any degree (e.g., a peak or dip shifting in frequency or in Q or bandwidth) then it is easy to over- or to under- correct a response. This can easily end up generating unrelated artifacts to the desired signal and these are extremely easy to perceive. This would indicate that we should concentrate on under correction and evenness of response over perfection, while reserving large corrections for highly stable conditions that can be completely and solidly predicted.

My thought is to reserve the most correction and power for the most stable region or the bass end of the spectrum, to match a relatively high-power amplifier with smaller, perhaps even much smaller, amplifiers for midrange and high frequencies, focusing as ever upon high sensitivity drive units but applying sensitive delicate response corrections to ensure flat responses over the critical bandwidth(s) with a maximum 3 dB boost in midrange and high frequencies with appropriate cut to be determined according to stability criterion. Bass, on the other hand, can be much more readily controlled so a larger boost can potentially be used here—bearing in mind that such results will inevitably rely heavily on room and individual setup anyway.

One of core concepts in setting up passive crossovers is that of adjusting and matching levels. It is here that the greatest losses tend to occur. However passive crossovers are not inherently lossy. In fact, you can gain or tweak extra energy in some regions. So, should we use passive crossovers to do much of the frequency splitting, but leave level shifting to the active circuits or to the DSP?

Klippel has demonstrated many techniques that are useful in improving the performance of a loudspeaker drive unit, but what if such techniques were to be applied to excellent designs in the first place... First and foremost being the relationships between various large signal parameters and distortion. Secondary would be the relationship between modal responses and the final response of a diaphragm. We can and should use all the tools and techniques from simulation through modeling and measurement.

So what would make an excellent design? First, minimize any distortion(s). Let us consider larger diaphragms that tend to beam and lobe at medium to high frequencies. However, they do not need excessive movement to produce enough SPL especially at the lowest frequencies. As Richard Honeycutt pointed out in a recent *audioXpress* article (see Resources), larger diaphragms tend to produce lower modulation distortion. If we combine elliptical or rectangular diaphragms, we can maximize the area with elliptical voice coils. As demonstrated by the recent Scan-Speak drivers (see **Photo 3**), we can drive the diaphragms more evenly with lower breakup and subsequent distortion, taller and narrower diaphragms will also beam less in the horizontal plane at least, and as our ears are more or less symmetrically attached to our heads, this will tend to produce a more even sound field.

Photo 3: Scan-Speak 18WE/4542T00 driver's front; and side view.

Also consider one of our more recent principles, that of ensuring sufficient Xmax or linear movement, which is essential and it will reduce distortion. But in reality, it is or should only really be a problem at the lowest frequencies where the excursion is at a maximum, and this becomes more pronounced with vented and ported or bandpass designs that have become today's "de facto" standard designs. So maybe we should break away from the single diaphragm to cover the entire frequency range. However, none of the changes I have discussed in this article are truly new.

Conclusion

What has prevented loudspeaker design from moving forward is not ideas or technology but rather the concepts of how we can solve these problems. I'm up for the challenge and it is one that I believe we must tackle head on to overcome. Marketing, over decades, has promised smaller and smaller loudspeakers, which successfully address part of the problem but carefully ignore the rest. This can be seen in the multiplicity of choices available with ported loudspeakers. The trouble is that at the lowest frequencies, the impedance mismatch is such that little power is effectively used to produce the desired sound. It is my contention that we need to take a fresh look at how we design our loudspeakers, considering many factors, not the least being the increase in mobile electronics and self-powered systems where vast amounts of power are wasted. Will you join me and my fellow engineers who are not happy with the appalling low efficiency of most loudspeakers and do something about it?

About the Author

Geoff Hill is a passionate engineer and can get frustrated when there is a better way to do something, but nobody else can see it. After qualifying in electronics (City and Guilds FTC in Telecommunications), he co-founded Impulse, a company that made high-efficiency horn loudspeakers. Later, Geoff worked with many companies and along the way he has designed everything to do with sound, including automotive, mobile phones, PA, hi-fi systems, and loudspeaker drivers from 25 mm tweeters up to 18" sub-bass drivers. His latest book, *Loudspeaker Modelling and Design*, was recently published by Routledge. An active member of the Audio Engineering Society, he is currently working within AES SC-04-03 on the subject of loudspeaker measurements and is a British member of IEC TC100. Geoff can be reached via www.geoff-hill.com or through his business, www.hillacoustics.com.

Resources

G. Hill, *Loudspeaker Modelling and Design: A Practical Guide*, Routledge, 2018.

R. Honeycutt, "Modulation Distortion in Loudspeakers and Its Dectability," *audioXpress*, September, 2018.

T. Perazella, "Beyond Thiele/Small DUMAX and Klippel Driver Measurement Systems," *audioXpress*, March, 2003

Build a Discrete High Output Current Buffer with JFET Input

Our audio expert explores ways to provide better isolation between input and output terminals and lower input noise current using complementary JFETs from Linear Integrated Systems.

By
Dimitri Danyuk
(United States)

A unity-gain buffer has many applications: input output isolation (make preceding circuit less sensitive to reactive loading), higher output current (increase the output capability of a preceding circuit, e.g., an IC op-amp), reduction of the thermal feedback induced errors in previous circuit (IC op-amp). There are many open-loop IC buffers on the market (e.g., the BUF634, the LMH6321, the LT1010, and the HA-5003). These integrated buffers are based on bipolar technology. They are essentially complementary emitter followers connected in series (except the LT1010).

Obsolete LH0033 and LH0063 buffers were fabricated with hybrid assembly technology and built with field-effect and bipolar transistors. These buffers comprise of an input junction gate field-effect transistor (JFET) source follower and an output bipolar complementary emitter follower. Due to the JFET input stage, these buffers may provide better isolation between input and output terminals and lower input noise current than buffers built with bipolar devices only. With the appearance of complementary JFETs from Linear Integrated Systems, it is possible to construct a discrete buffer with JFET input. The n-channel LSK189 (or the LSK489, which is dual version of the LSK189) and the p-channel LSJ289 (or the LSJ689, which is a dual version of LSJ289) are used in this design.

Circuit Description

The schematic of the discrete buffer based on the LH0033/LH0063 is shown in **Figure 1**. The first input source follower U1A (LSK489) and the first emitter follower U3B, U5A, U5B (LS3250) provide current gain for the positive signal while the second source follower U2B (LSJ689) and second emitter follower U4B, U6A, U6B (LS3550) provide current gain for the negative signal. The output emitter followers are biased with bipolar transistors U3A and U4A in diode connection. These forward conducting diodes also set the gate-to-source voltage in the input source followers. With gate-to-source voltage being equal to 0.6 V, drain current of U1A and U2B is about 1.5 mA. LSK189 (LSK489) and LSJ289 (LSJ689) are particularly suited to this circuit because of large gate-source pinch-off voltage, low capacitance, and low noise.

JFETs U1B and U2A are configured as current sources. They are adding DC current to the output current of associated input source follower. Quiescent current of the output devices is set by the current through diodes U3A, U4A. Without current sources

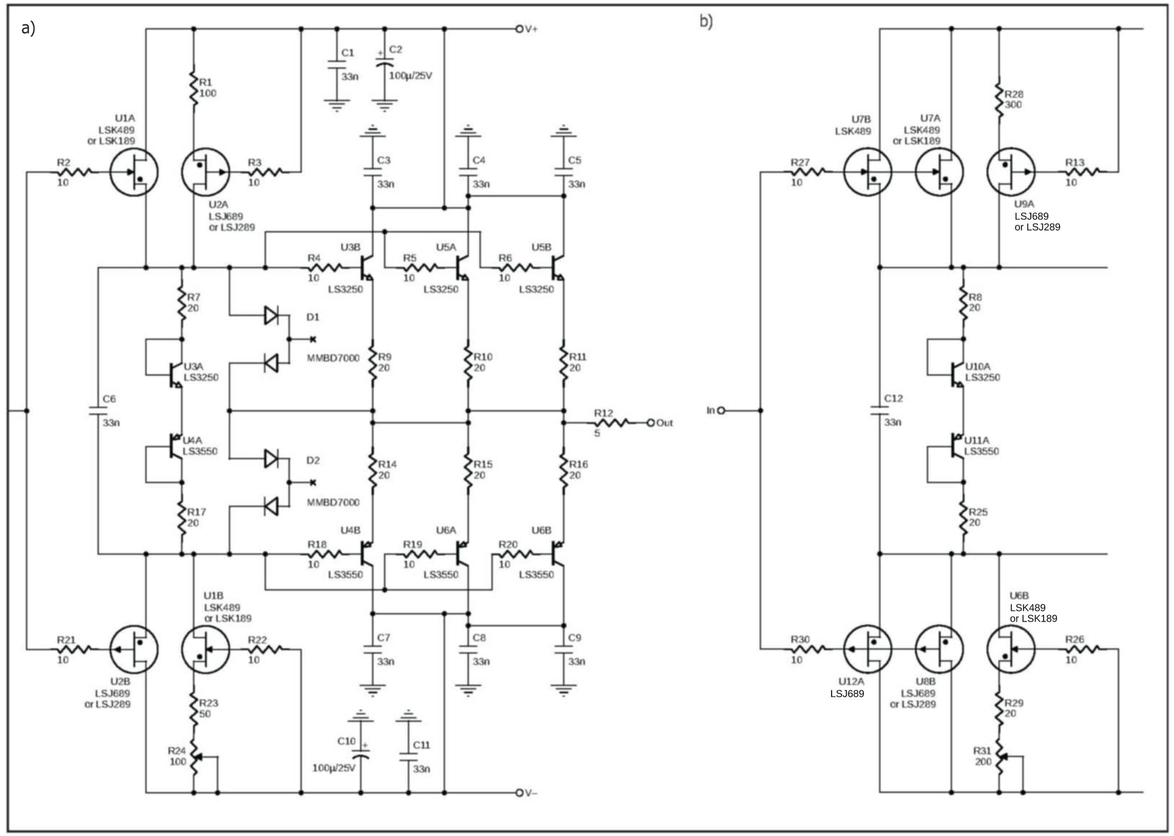


Figure 1: Here is a schematic of a discrete buffer with JFET input.

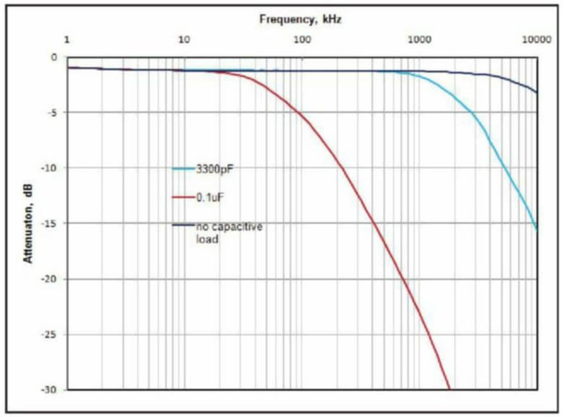


Figure 2: Frequency response of the discrete buffer with JFET input with different load capacitance. Output 2 V_{PP}, load 100 Ω.

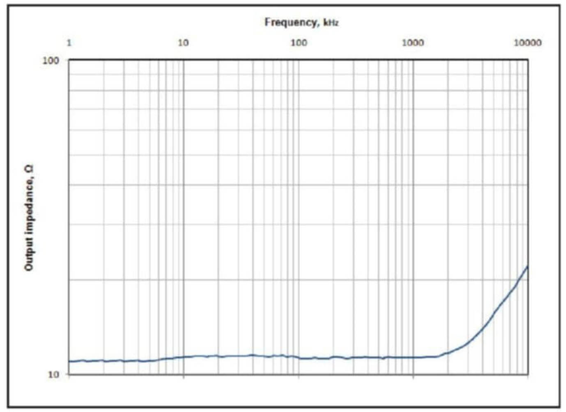


Figure 3: The output impedance of the discrete buffer with JFET input, measured at 10 mA peak output current.

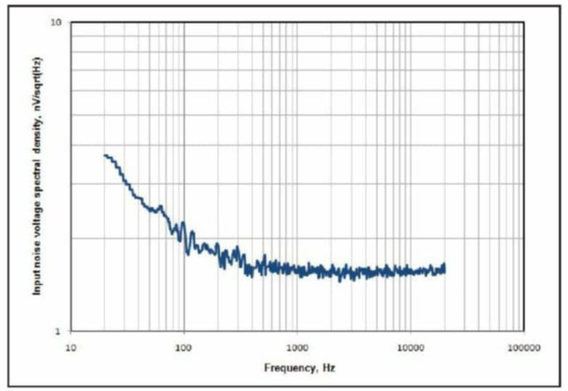


Figure 4: The input noise voltage spectral density of the discrete buffer with JFET input, the input is shorted to ground.

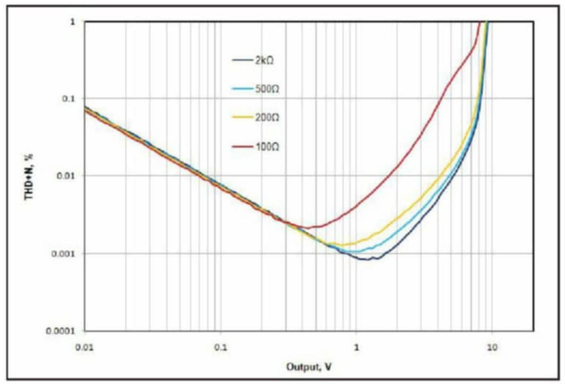


Figure 5: The total harmonic distortion plus noise (THD+N) of the discrete buffer with the JFET input is shown as a function of the output voltage for different loads (100 Ω, 200 Ω, 500 Ω, and 2 kΩ). Power supply voltage is ±24 V, signal frequency is 1 kHz.

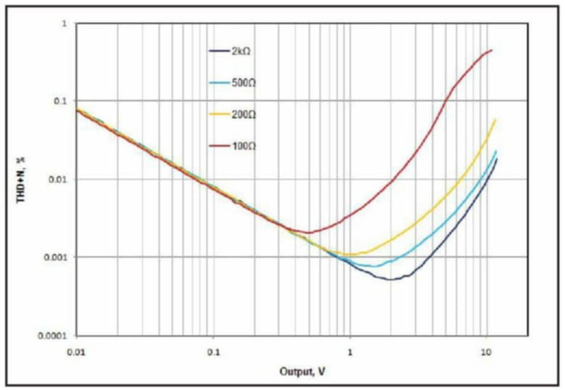


Figure 6: The THD+N of the discrete buffer with the JFET input is shown as a function of the output voltage for different loads (100 Ω, 200 Ω, 500 Ω, and 2 kΩ). Power supply voltage ±15 V, signal frequency is 1 kHz.

Figure 8: The distortion spectrum is measured for a 5 Vrms output voltage at 500 Ω, fundamental frequency is 1 kHz.

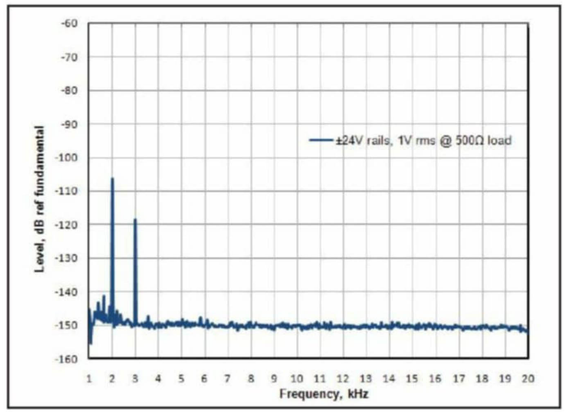


Figure 7: The distortion spectrum is measured for 1 Vrms output voltage at a 500 Ω load, fundamental frequency is 1 kHz.

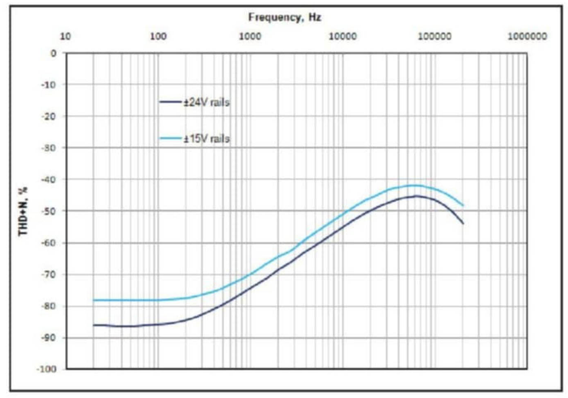


Figure 9: The THD+N is shown as a function of frequency for different power supply voltages (±15 V and ±24 V). Source resistance is 100 kΩ, output voltage is 5 Vrms at a 500 Ω load.

U1B and U2A quiescent current will be 1.5 mA and this value may be too low. With the value of source resistors (R1, R23+R24) about 100 Ω, U1B and U2A supply additional 2.5 to 3 mA, and the current through bias diodes U3A, U4A is equal to 4.5 to 5 mA.

Quiescent current of the output devices can be chosen by proper selection of resistor R1, R23+R24. Variable resistor R24 can be used to adjust the current of the U1B current source. This adjustment trim DC offset between input and output and the symmetry of the output sine wave (second order distortion.) The optimal setting for zero DC offset is different from the optimal

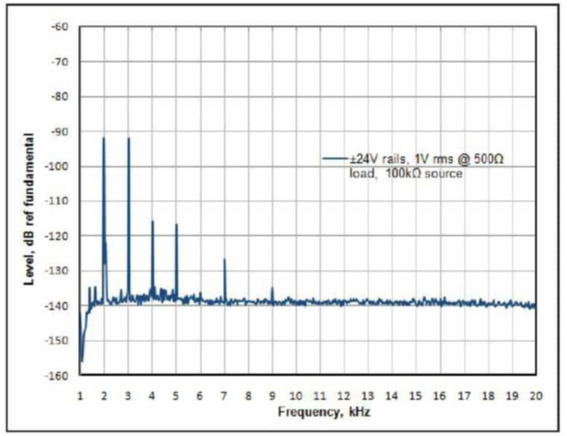


Figure 10: The distortion spectrum is measured for 1 Vrms output voltage at a 500 Ω load, source resistance is 100 kΩ, fundamental frequency is 1 kHz.

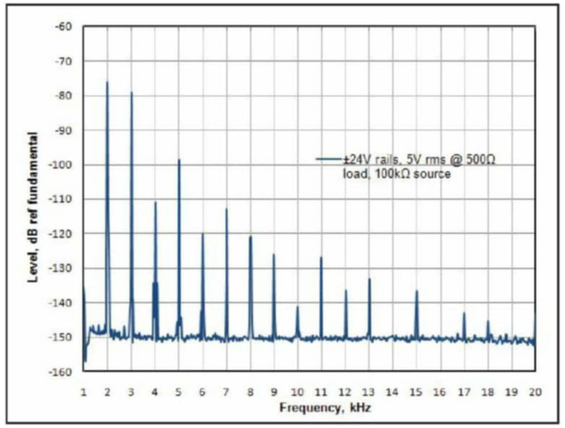


Figure 11: The distortion spectrum is measured for 5 Vrms output voltage at a 500 Ω load, source resistance is 100 kΩ, fundamental frequency is 1 kHz.

Spiral Air tm Interconnects
Coax Is Cool Again!
Cardas Golden Strand
Air Filled Cotton Dielectric
Unique Spiral Litz Shield
ETI Tellurium Copper Connectors
Affordable Litz Interconnects and Bulk Wire
Tonearm - Phono - Line - Loudspeaker
KAB
Preserving The Sounds Of A Lifetime
www.kabusa.com

DO IT YOUR WAY
Cables/Panels/Boxes
WITH REDCO AUDIO'S
"Build Your Own" web pages
Full control for design, editing and more.
AND WE'LL BUILD IT JUST THE WAY YOU WANT!
REDCO AUDIO
www.redco.com 1-800-572-7280

About the Author

Dimitri Danyuk has been the principal hardware engineer at Harman luxury audio since 2013. He graduated with honors from Kyiv Polytehnic in 1985. His passion is to help audiophiles in all aspects of audio design. His articles have appeared in a number of magazines including *Journal of the Audio Engineering Society* (JAES), *Institute of Electrical and Electronics Engineers* (IEEE) *Transactions*, *EDN*, *Electronic Design*, and more. Dimitri also provides consulting services for various businesses.

setting for minimum of the second harmonic. P-channel device (the LSJ289 or the LSJ689) has higher transconductance than the n-channel counterpart (the LSK189 or the LSK489) at the same operating conditions.

Output followers U3B, U5A, U5B, and U4B, U6A, U6B may sink or source a substantial amount of current and the voltage drop between their bases and emitters will decrease as they heat up. The bias diode U3A and the output follower transistor U3B are matched components and they share the same package. Temperature of the bias diode is similar to the temperature of output device. This thermal tracking helps to keep nearly constant quiescent current in the output followers and prevents thermal runaway. The current through each output transistor is about 4.5 to 5 mA, making the total quiescent current of the output stage equal to 15 mA. Three devices are connected in parallel to ensure operation within the dissipation limit of 200 mW for the SOT363 package (U3-U6). Dual diodes D1 and D2 provide the current limiting, clamping the drive to the bases of output emitter followers.

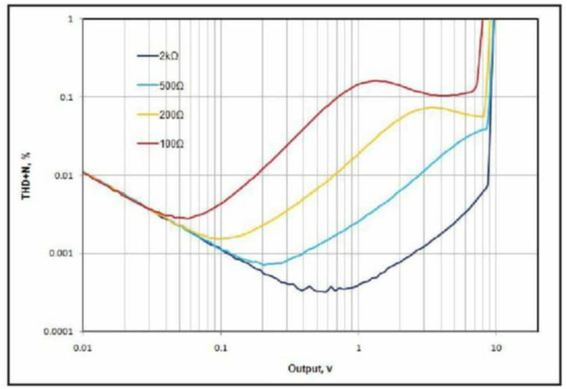


Figure 12: The BUF634's THD+N is shown as a function of the output voltage for different loads (100 Ω, 200 Ω, 500 Ω, and 2 kΩ). The power supply voltage is ±15 V, signal frequency is 1 kHz, wide bandwidth mode.

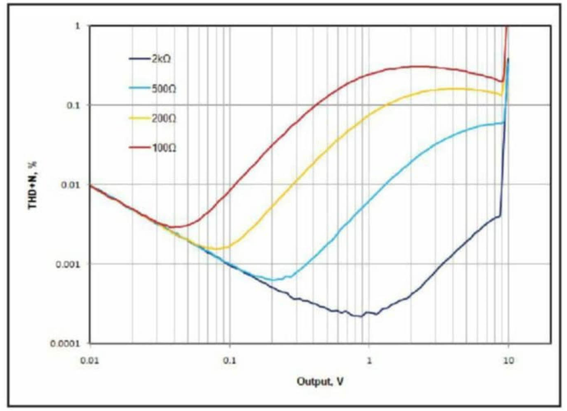


Figure 13: The HA5002's THD+N is shown as a function of the output voltage for different loads (100 Ω, 200 Ω, 500 Ω, and 2 kΩ). The power supply voltage is ±15 V, signal frequency is 1 kHz.

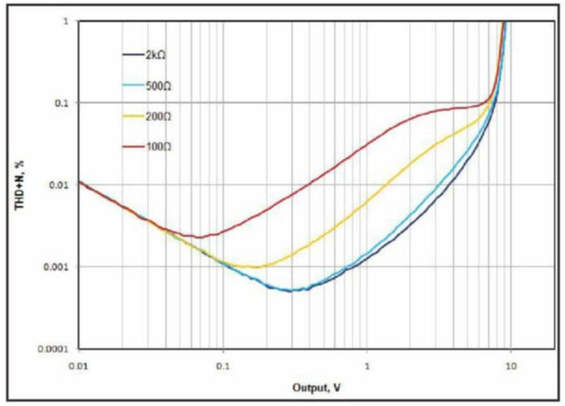


Figure 14: The LH0033's THD+N is shown as a function of the output voltage for different loads (100 Ω, 200 Ω, 500 Ω, and 2 kΩ). The power supply voltage is ±15 V, signal frequency is 1 kHz.

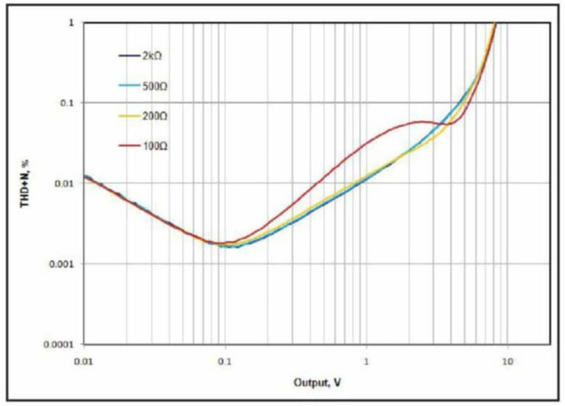


Figure 15: The LH0063's THD+N is shown as a function of the output voltage for different loads (100 Ω, 200 Ω, 500 Ω, and 2 kΩ). The power supply voltage is ±15 V, signal frequency is 1 kHz.

Test Results

The measured buffer frequency response is shown in **Figure 2**. With a resistive load, the bandwidth is wider than 10 MHz (about a 2 dB roll-off at 10 MHz, phase lag is about 30°). The bandwidth is limited by the output impedance of the JFET source followers, the collector-base capacitances of the output transistors, and the parasitic capacitance of PCB traces. The frequency response can be extended by doubling the amount of source follower devices (see Figure 1b). The penalty is correspondent rise in the input capacitance. The buffer's input capacitance (see Figure 1a) is equal to 18 pF (the gate-drain capacitance of the input JFETs and the capacitance of PCB input trace—about 3 pF).

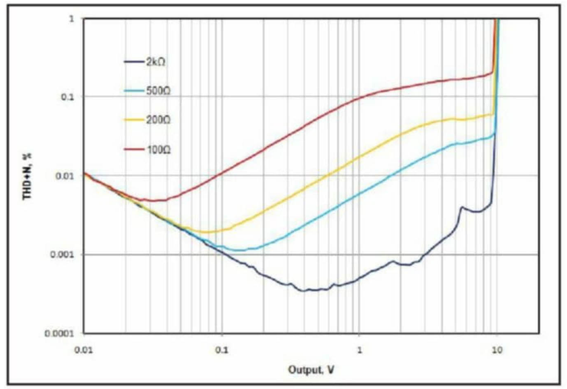


Figure 16: The LME49600's THD+N is shown as a function of the output voltage for different loads (100 Ω, 200 Ω, 500 Ω, and 2 kΩ). The power supply voltage is ±15 V, signal frequency is 1 kHz, wide bandwidth mode, BW pin connected to VEE.

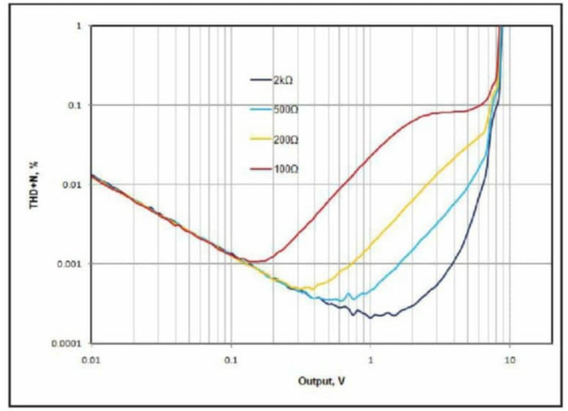
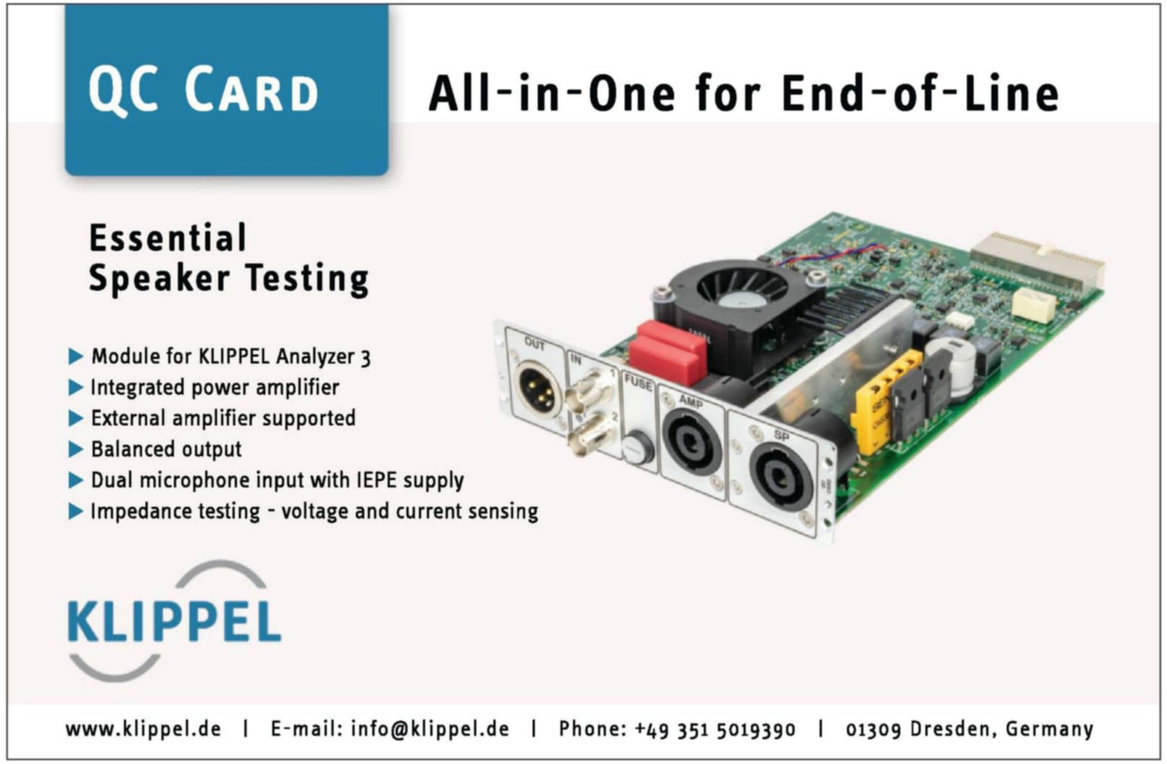


Figure 17: The LMH6321's THD+N is shown as a function of the output voltage for different loads (100 Ω, 200 Ω, 500 Ω, and 2 kΩ). The power supply voltage is ±15 V, signal frequency is 1 kHz.



The buffer output impedance is essentially resistive (10 Ω) and maintained constant up to 2 MHz, as shown in **Figure 3**. This includes the series resistor R12 (5 Ω). The series resistor may prevent parasitic oscillations with the capacitive loads. The noise voltage spectral density shown in **Figure 4** is equal to 1.7 nV/√Hz at 1 kHz.

Figure 5 and **Figure 6** illustrate the unity buffer's total harmonic distortion plus noise (THD+N) as a function of the output level for two different rail voltages. Provided distortion data were measured with low signal source resistance (50 Ω). The distortion remains constant with the frequency within the 20 kHz range. Typical distortion spectrums for different output voltage are shown in **Figure 7** and **Figure 8**.

The buffer input may come from a high impedance signal source. If the input impedance of the buffer is changing with the input voltage, the resulting distortion will be significantly higher than the values obtained for the low source impedance. Ideally, to preserve low distortion when operating with high source impedances, the input capacitance needs to be independent of the input voltage.

Measured distortion as a function of frequency is shown in **Figure 9** for the value of signal source resistance 100 kΩ. While the frequency response is flat until 100 kHz, distortion increases with the rise of the frequency of the input sine wave. Distortion spectrums for different output voltage are shown in **Figure 10** and **Figure 11**. These spectrums can be compared with the ones shown in Figure 7 and Figure 8. The input impedance nonlinearity can be improved by implementing series cascodes in the input stage (e.g., with MMBFJ310/MMBFJ176).

It is instructive to compare the performance of the proposed circuit with open-loop IC buffers. While comprehensive test results for IC buffers can be found in their corresponding datasheets (see Resources), open-loop distortion measurements are hard to locate. Open-loop distortion measurements for standalone IC buffers are presented in **Figures 12-18**. The test conditions are the same as the test conditions for measurements shown in Figure 5 and Figure 6. It is worthwhile to note that the simple circuit shown in Figure 1a produces lower open-loop distortion than many IC buffers.

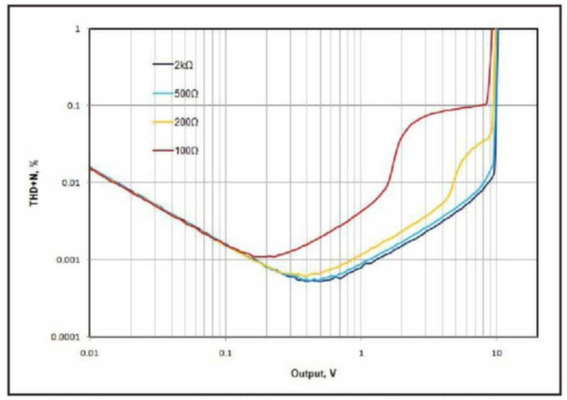


Figure 18: The LT1010's THD+N is shown as a function of the output voltage for different loads (100 Ω, 200 Ω, 500 Ω, and 2 kΩ). The power supply voltage is ±15 V, signal frequency is 1 kHz, high current mode, Rbias = 25 Ω.

Conclusion

The remarkable low distortion, wide bandwidth, low noise, and high output drive capability of the discrete buffer with JFET input allows it to be used in many audio circuits, even open loop. Among the possible applications, high-end volume control, transconductance amplifying stage isolation, line buffer, and headphone preamp come to mind. High-end volume control may consist of low impedance volume control of the designer's choice (potentiometer, resistor ladder, transformer) isolated from signal source and the following circuit by a pair of discrete buffers with JFET input. If lower load resistance is specified, the output current capability can be increased by adding extra parallel devices to the output stage. LSK489/LSJ689 in the input stage can be replaced with MMBFJ310/MMBFJ176 at the expense of increased noise. Dual-matched bipolar devices LS3250/LS3550 can be substituted with BCM847/BCM857 or higher power PHPT610035NK/ PHPT610035PK.

The proposed discrete design outperforms off-the-shelf ICs in certain features and may serve as an economical alternative. By using various types and number of semiconductor devices, the discrete buffer can be optimized for different sources and loads.

Datasheets

"LH0033/LH0063 Fast and Ultra Fast Buffers," National Semiconductor, Datasheet, pp.1-14.

"BUF634 250 mA High-Speed Buffer," Texas Instruments, Datasheet, pp.1-31.

"HA-5002 110MHz, High Slew Rate, High Output Current Buffer," Intersil, Datasheet, pp.1-12.

"LME49600 High-Performance, High-Fidelity, High-Current Headphone Buffer," Texas Instruments, Datasheet, pp.1-21.

"LMH6321 300 mA High Speed Buffer with Adjustable Current Limit," Texas Instruments, Datasheet, pp.1-31.

"LT1010 Fast ±150mA Power Buffer," Linear Technology, Datasheet, pp.1-20

Resources

"Applications for a New Ultra-High Speed Buffer," National Semiconductor, Application Note 48, pp.1-8.

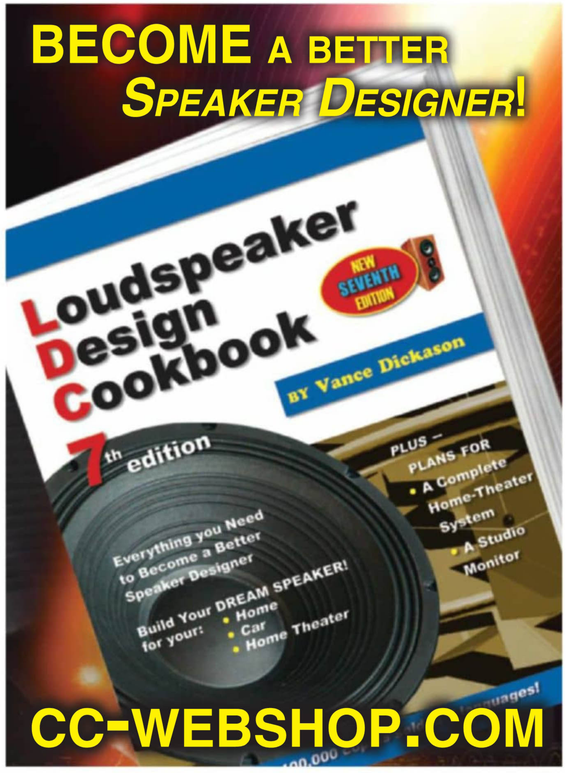
"Applications of Wide-Band Buffer Amplifiers," National Semiconductor, Application Note 227, pp.1-22.

W. Jung, "Op-Amp Audio: Realizing High Performance: Buffers: Part I," Electronic Design, September 1, 1998.

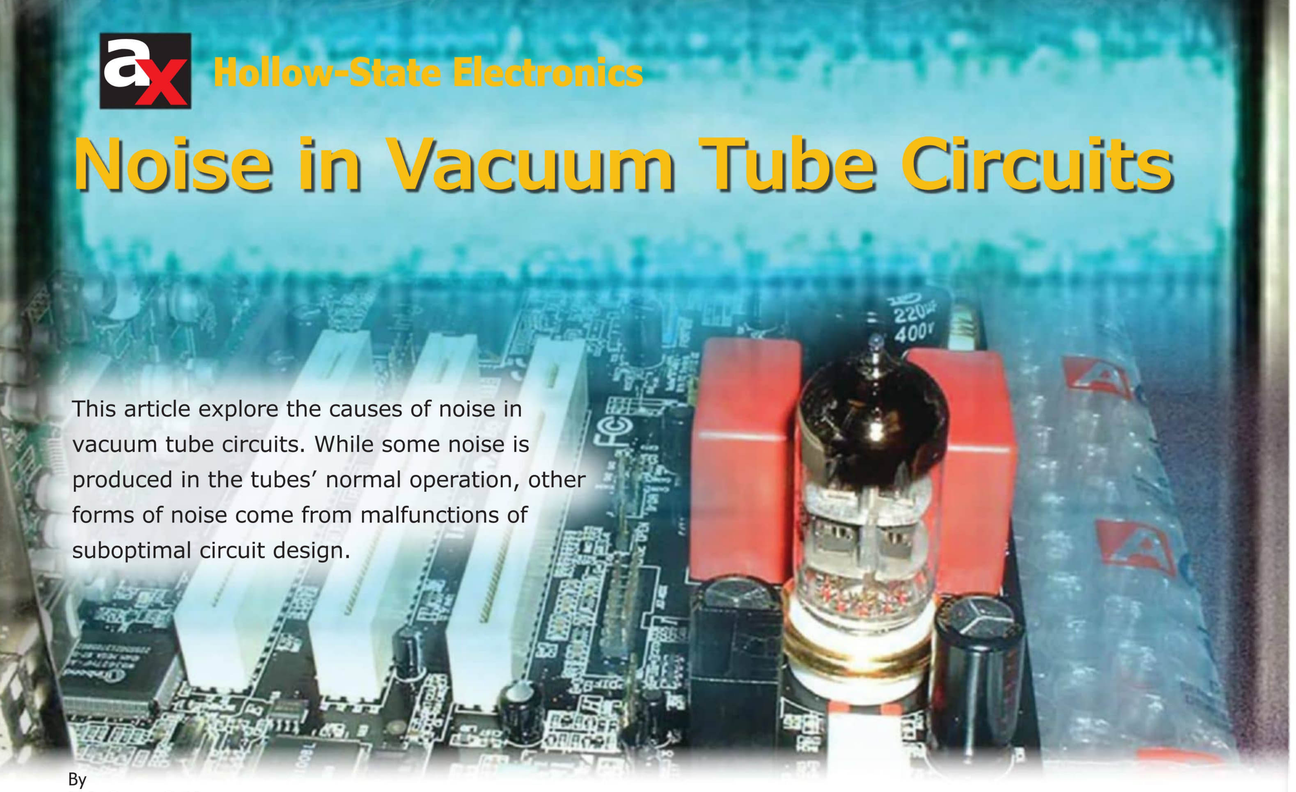
——, "Op-Amp Audio: Realizing High Performance: Buffers: Part II," *Electronic Design*, October, 1998, pp.103-104.

R. Widlar, "Circuit output stage arrangement," US Patent 4573021 (internals of LT1010 buffer).

S. Yee, "Complementary emitter follower amplifier biased for class A operation," US Patent 3418589 (first complementary emitter followers in series, dating back to 1965).



Noise in Vacuum Tube Circuits



This article explore the causes of noise in vacuum tube circuits. While some noise is produced in the tubes' normal operation, other forms of noise come from malfunctions of suboptimal circuit design.

By
Richard Honeycutt
(United States)

In the early days of hollow-state technology, vacuum tubes were used as detectors—and later, as amplifiers—in radio receivers. Subsequently, they came into use in AM radio transmitters. In neither case did noise generated internally in the tubes matter much, since atmospheric noise dominated the denominator of the signal-to-noise ratio (SNR). In the 1930s, audio applications of tubes were rapidly expanding, and several laboratories, notably the Bell Telephone Laboratories (BTL), began investigating internally generated tube noise. As early as 1926, John Bertrand Johnson of Bell Labs discovered noise due to random thermal motion of electrons in any conductor. With the help of his BTL colleague, Harry Nyquist, he described how to mathematically predict its voltage (see Resources).

Frederick Briton Llewellyn and G. L. Pearson of BTL described in 1930 and 1934, respectively, other types of noise occurring in vacuum tubes (see Resources). They discussed the three types of noise we will address in this article: shot noise, thermal ("Johnson") noise, and noise produced by ionization and secondary effects.

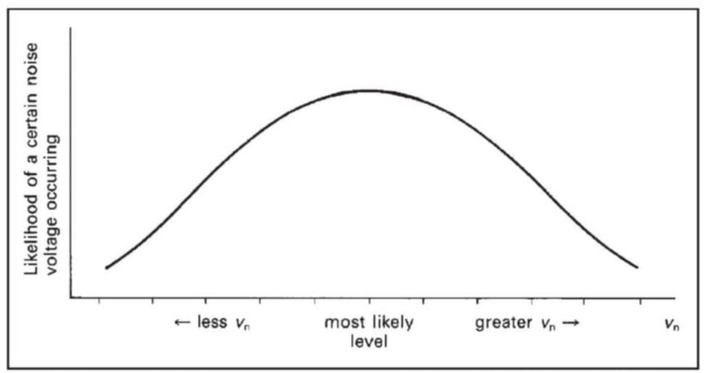


Figure 1: Gaussian distribution of noise voltages

Thermal Noise

Thermal noise is produced by the random motion of electrons caused by heat energy. Even though this noise is random by nature, we can predict the most likely noise voltage if we know the temperature and resistance. Johnson gave this equation:

$$v_n^2 = 4kTR\Delta f \text{ volts}^2 \qquad [1]$$

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

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

$$v_n = 4\text{nV}\sqrt{\frac{R}{1k\Omega}}\sqrt{\Delta f} \qquad [2]$$

This simplified equation is valid for a temperature of 298 kelvins (K), or 25°C. Often both sides of the equation are divided by $\sqrt{\Delta f}$, yielding an equation for noise in volts/$\sqrt{\text{Hz}}$ ("volts per root hertz"):

$$\frac{v_n}{\sqrt{\text{Hz}}} = \frac{4\text{nV}}{\sqrt{\text{Hz}}}\sqrt{\frac{R}{1k\Omega}} \qquad [3]$$

The likelihood of measuring a certain noise voltage in a given resistance at a given temperature, at a specific instant in time is described by a Gaussian distribution (see **Figure 1**). Thermal noise can contain literally any frequency, so by analogy with white light, which contains all frequencies, thermal noise is classed as a white noise (see **Figure 2**). Most thermal noise in well-designed high-end hollow-state amplifiers is generated in the input circuit. However, the internal plate resistance of a tube produces thermal noise just as if it were a resistor at the temperature of the cathode. The equation for internal thermal noise in a vacuum tube is then:

$$\frac{v_n}{\sqrt{\text{Hz}}} = \sqrt{4kT_0\left(r_p/\mu\right)^2\left[T_1/T_0 r_p + 1/r_0\right]} \qquad [4]$$

where T_0 and T_1, are the temperatures of the external plate resistor and the tube's cathode, respectively, and r_0, and r_p are the resistances of the external plate resistor and the tube's plate resistance. Equation [4] is the equivalent thermal noise at the input of the tube amplifier stage.

Figure 3 shows an RCA circuit for a phonograph preamplifier using the two triode stages of a 7025. In the first stage of this circuit, the external plate resistance equals the parallel combination of R3 and the sum R6 + R8, which is 92 kΩ. The plate resistance of the 7025 in this circuit is about 62 kΩ. The cathode temperature is probably about 900°C = 1174 K, and let us say that the external plate resistor has a temperature of about 35°C = 309 K. We then find that the internally generated thermal noise in that stage is 0.684 nV/$\sqrt{\text{Hz}}$. A moving-magnet phono cartridge typically has an internal resistance of about 1.5 kΩ, and the cartridge's temperature is usually about 25°C = 299 K. Thus, the input circuit (ignoring the tiny effect of R1—usually 47 kΩ—in parallel with the cartridge) creates a noise level of 6.5 nV/$\sqrt{\text{Hz}}$. As you can see, the noise in the input circuit including the cartridge dominates the total noise in the first stage.

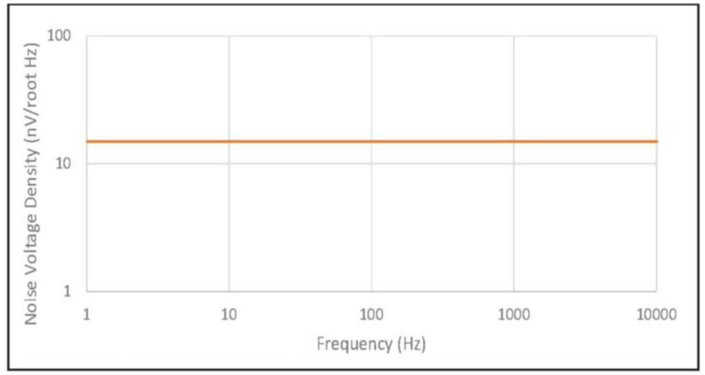


Figure 2: White noise

As you can see in **Figure 4**, in a guitar amp input circuit using a 7025, the external plate resistor often has about the same resistance as we found in the phono preamp, but the resistance of the signal source (guitar pickup) is higher (6 to 9 kΩ), increasing the input noise voltage to about 24 to 36 nV/$\sqrt{\text{Hz}}$. This increased noise is inconsequential, however, because the guitar pickup's output voltage is at least ten times that of the phono preamp, making the guitar amp's SNR higher as well.

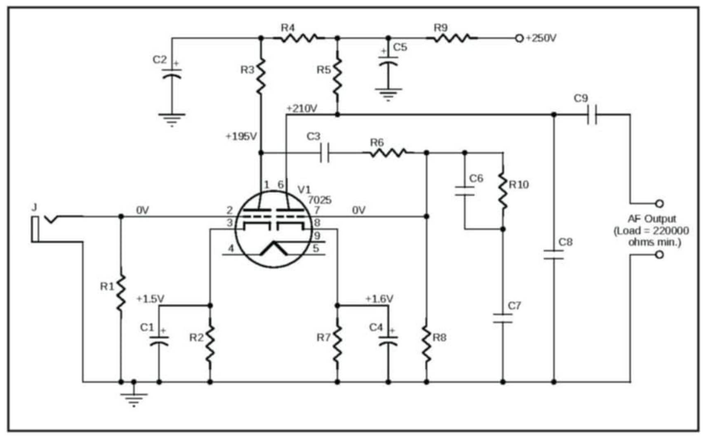


Figure 3: RCA phono preamp circuit

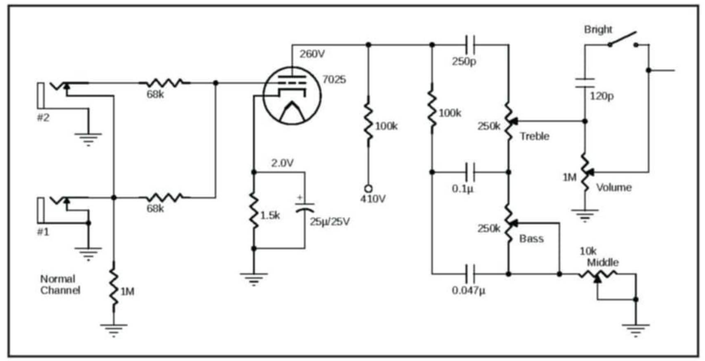


Figure 4: Input circuit of Fender Bandmaster guitar amplifier

Calculations show that as long as the designer does not severely attenuate the signal somewhere in an amplifier (as by passive tone controls), the noise in the first stage dominates the total amplifier noise. This is because the stage 1 noise is amplified by all the succeeding stages, whereas the noise of following stages experiences less amplification.

Shot Noise

Irregularities in the stream of electrons from cathode to plate cause a type of noise described by Walter Schottky (see Resources) as "sounding like lead shot hitting a tin roof." Shot noise is worse if for some reason the filament is not sufficiently heated to produce a full space charge. Under this condition, the electrons emitted from the cathode proceed from random thermal processes, and all such electrons are immediately attracted to the plate.

Figure 5 illustrates the plate current from a heated filament as a function of temperature.

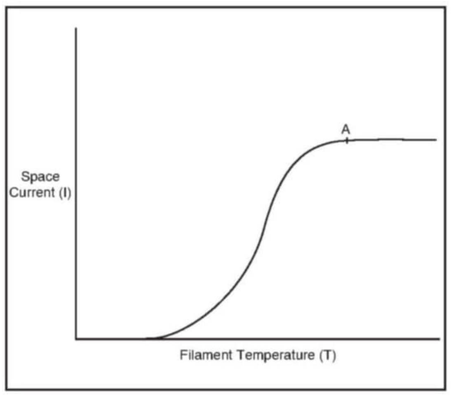


Figure 5: Variation of space current with filament temperature

You can see that the cool filament produces few electrons, limiting the plate current. But as the filament becomes hotter, it eventually emits more electrons than are attracted to the plate, and a space charge is formed (point A on the curve). This condition is called thermal saturation. With the filament at temperatures insufficient to produce thermal saturation, shot noise is high. In normal operation of a tube, shot noise is not much of a problem, since thermal saturation is usually achieved, and thus a continuous stream of electrons flows from the space cloud to the plate. Tubes whose filament emission has declined because of long use or low mains voltage can produce excess shot noise. Like thermal noise, shot noise is also Gaussian in level distribution, and it is also white in spectrum.

Noise Produced by Ionization and Secondary Effects

If the tube filaments operate in thermal saturation, ionization and secondary effects are usually responsible for most of the noise that does not proceed from thermal effects. When a tube is operating, the electrons can strike residual gas molecules remaining from imperfect evacuation of the envelope, and they will also strike the grid and plate. In either case, ions may be produced. These cause a popping noise in audio equipment, not unlike shot noise. Unless a tube is "gassy" (having a relatively large amount of residual gas), noise from negative ions is seldom much of a problem.

Positive ions are another story. They can be produced by molecules bombarded by electrons, or even by positive ions being evaporated from the filament. These positive ions are attracted to the space charge and can make substantial changes in the space-charge density.

Although not generated within tubes themselves, another type of noise, called "flicker noise," can degrade the SNR of both hollow- and solid-state amplifiers. Flicker noise occurs in the presence of direct current in semiconductors, and in carbon-composition and thick-film resistors. The noise can be somewhat controlled in the manufacturing process, but it is Gaussian in its amplitude distribution.

Unlike the other forms of noise, flicker noise has a probable amplitude that is inversely proportional to frequency, and hence is often called "1/f noise." **Figure 6** shows a graph illustrating the total output noise voltage of an amplifier, including both thermal and flicker noise. In this case, the corner frequency (where the amplitude of the 1/f noise equals the amplitude of the white broadband noise) is 60 Hz. Depending upon listening level (which affects the low-frequency response of hearing) and speaker performance, the 1/f noise of this amplifier may or may not matter.

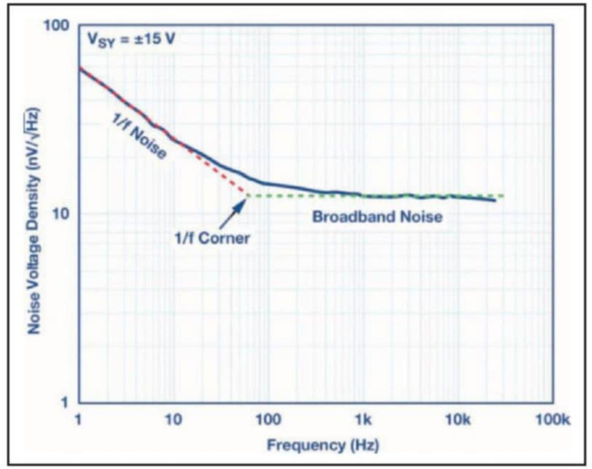


Figure 6: Combined thermal (white) and 1/f noise

Specifying Noise from All Causes

Noise voltages from different causes are not coherent (phase-synchronized). Therefore, the total noise cannot be obtained by simply adding the RMS values of the individual noise voltages. Instead, they must be added in a root-sum-square fashion:

$$v_n = \sqrt{v^2_{thermal} + v^2_{shot}} \qquad [5]$$

Noise voltages from other sources can also be squared and added under the radical. On tube datasheets, the total noise is often specified in ways that represent marketing more than data presentation. Some datasheets for the 12AX7 list the noise as 100 mV, which is presumably an output noise level, not an equivalent input noise level. In most of these datasheets, the tube's operating parameters are not given. One website, Tavish Design, in its "Audio Tube Noise Measurements" table (https://tavishdesign.com/pages/audio-tube-noise-measurments), does list the tube operating parameters and the measurement bandwidth. For example, it lists the input-referred noise from a JJ 12AX7 (ECC83S) as 0.587 µV, over a bandwidth of 30 Hz to 8 kHz, with a plate current of 1.45 mA and a gain of 60.6. The source resistance was 390 Ω.

This is useful information. We can divide the noise voltage by the square root of the frequency range of 7970 Hz to find an equivalent input noise level of 6.58 nV/$\sqrt{\text{Hz}}$. We do not know the value of the AC plate load ($r_L = R_p||R_L$), but using a value of 60 kΩ for r_p, we can calculate backward from the gain. We find that R_p was about 100 kΩ. As you will recall, the thermal noise in the output signal depends upon r_p, not only R_p. We are assuming that when this test was made, no carbon-composition or thick-film resistors were used, so the 1/f noise is negligible.

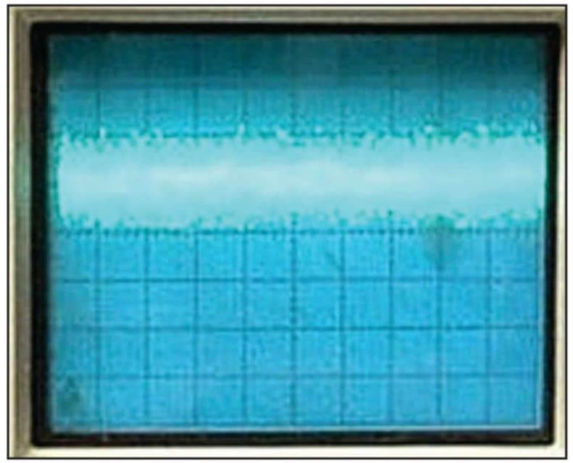


Figure 7: White noise on Channel 1

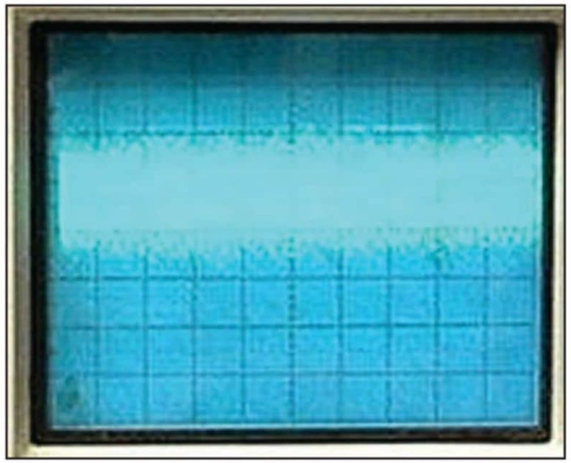


Figure 8: White noise on Channel 1 and Channel 2 merged

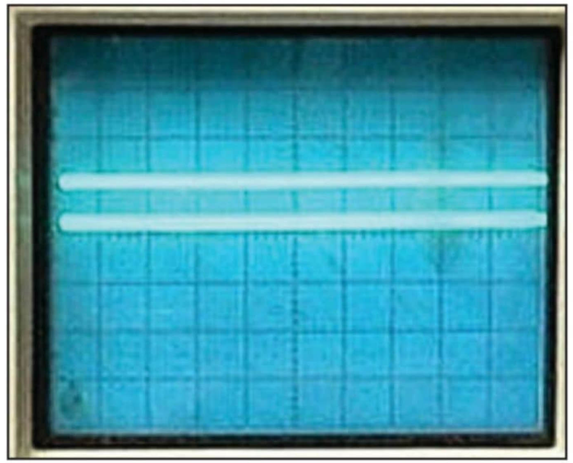


Figure 9: Channel 1 and Channel 2 inputs grounded

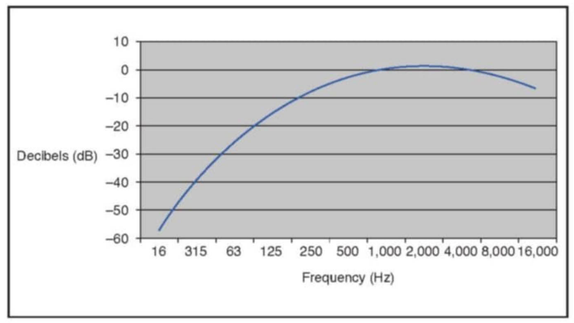


Figure 10: A-Weighting curve

Measuring Noise

Noise measurements can be made with an RMS voltmeter. Many analog voltmeters, and some digital ones as well, use a peak-responding or average-responding detector. Actually, these will give correct peak or average noise voltage readings, but the readings are scaled according to the peak/RMS or average/RMS ratio for a sine wave, so the reading would be correct if a sine wave were being measured. Noise does not have the same peak/RMS or average/RMS ratios as a sine wave, so the scaling introduces some error—typically 1.1 dB for an average-responding detector. A true RMS detector—now available on many digital voltmeters—is preferable.

However, there is another technique using a dual-trace oscilloscope that produces pretty accurate measurements. First, feed the channel 1 probe (set on H1, not H10)with the noisy signal, and adjust the channel 1 vertical volts/division switch so that the noise takes up about 1/4 or so of the screen height. Leave the continuous sensitivity adjustment locked in the calibrated position. Use the vertical position control to move the trace into the upper half of the screen. The sweep speed (time/division) should be adjusted so that the noise appears as sort of a ribbon across the screen (see **Figure 7**).

Now set the Channel 2 controls to match the Channel 1 controls and connect the Channel 2 probe to the point where the Channel 1 probe is connected. Use the Channel 2 vertical position control to adjust the position of the lower "noise ribbon" so that the dark band between the two ribbons just barely disappears (see **Figure 8**).

Now switch the probes' switches to "ground," or if they don't have switches, disconnect the probes from the circuit under test and clip the two probes together to the oscilloscope ground. You will see two horizontal lines on the screen. Using the volts/division setting, calculate how many volts are represented by the separation between these lines. In **Figure 9**, the lines are one vertical division apart. If the volts/division switches are set on 10 mV, this separation represents 10 mV, which is the RMS voltage of the noise signal.

Conclusion

We have discussed the frequency dependence of total noise: If flicker noise is significant, noise is greater at low frequencies. If not, noise voltage per root Hertz is independent of frequency, but total noise voltage is proportional to the square root of the range of frequencies to which the measuring instrument responds.

In order to provide noise measurements for audio equipment that are more-or-less aligned with human perception, noise-measurement weightings were introduced. Human hearing sensitivity varies with frequency, and the range of variation changes with level. While, at the threshold of hearing, there is a 79-dB difference between the 20-Hz sensitivity and the 4000-Hz sensitivity, at levels around 100 dB, the difference between the 20-Hz and 4000-Hz sensitivities is only about 42 dB. In order to take the ear's varying sensitivities into account, sound levels are sometimes measured with "A-weighting," a method that de-emphasizes the contribution of low frequencies to the sound pressure level. Such measurements are specified in dBA. A-weighting a noise measurement approximately compensates for the hearing response at a level of 40 dBSPL. The A-weighting curve is illustrated in **Figure 10**.

Resources

J. Johnson, "Thermal Agitation of Electricity in Conductors," *Physical Review,* 32 (97), https://journals.aps.org/pr/abstract/10.1103/PhysRev.32.97

F. B. Llewellyn, *A Study of Noise in Vacuum Tubes and Attached Circuits*, Proceedings of the Institute of Radio Engineers, Volume 18, February, 930.

H. Nyquist, "Thermal Agitation of Electric Charge in Conductors," 1928, *Physical Review*, 32 (110).

G. L. Pearson, "Fluctuation Noise in Vacuum Tubes," *Physics*, September, 1934.

W. Schottky, "Über spontane Stromschwankungen in verschiedenen Elektrizitätsleitern," *INK*, 1981, "https://en.wikipedia.org/wiki/Annalen_der_Physik" *Annalen der Physik* (in German).

"Thermal agitation of electricity in conductors," Proceedings of the American Physical Society, Minutes of the Philadelphia Meeting December, 1926, *Physical Review*, 29, 1927.

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 20–23, 2019
146th AES Convention
The Convention Center, Dublin, Ireland
Conference: March 20–23
Exhibits: March 21–23
www.aes.org/events/146

The Audio Engineering Society, celebrating more than 70 years of audio science and innovation, is gearing up to host the 146th International Audio Engineering Society Convention, March 20-23 at the Convention Center in Dublin, Ireland. As part of the Opening Ceremonies, Stefania Serafin, professor in sonic interaction design at Aalborg University Copenhagen, will examine the intersection of audio and technology in the address "Inclusive Sonic Interaction." This presentation, part of the AES Dublin Special Events series, is open to all attendees and will set the stage for four full days of immersion in audio technology and its application.

The theme of the Convention, "Excite Your Ears," will be supported by a host of Special Events, exhibition floor event stages and a myriad of opportunities to learn from—and network and interact with—audio pros from around the world.

The four day Technical Program will feature a variety of ways to listen, learn and connect with the leading minds in audio through Technical Papers, eBriefs, Workshops, Tutorials, and other presentations.

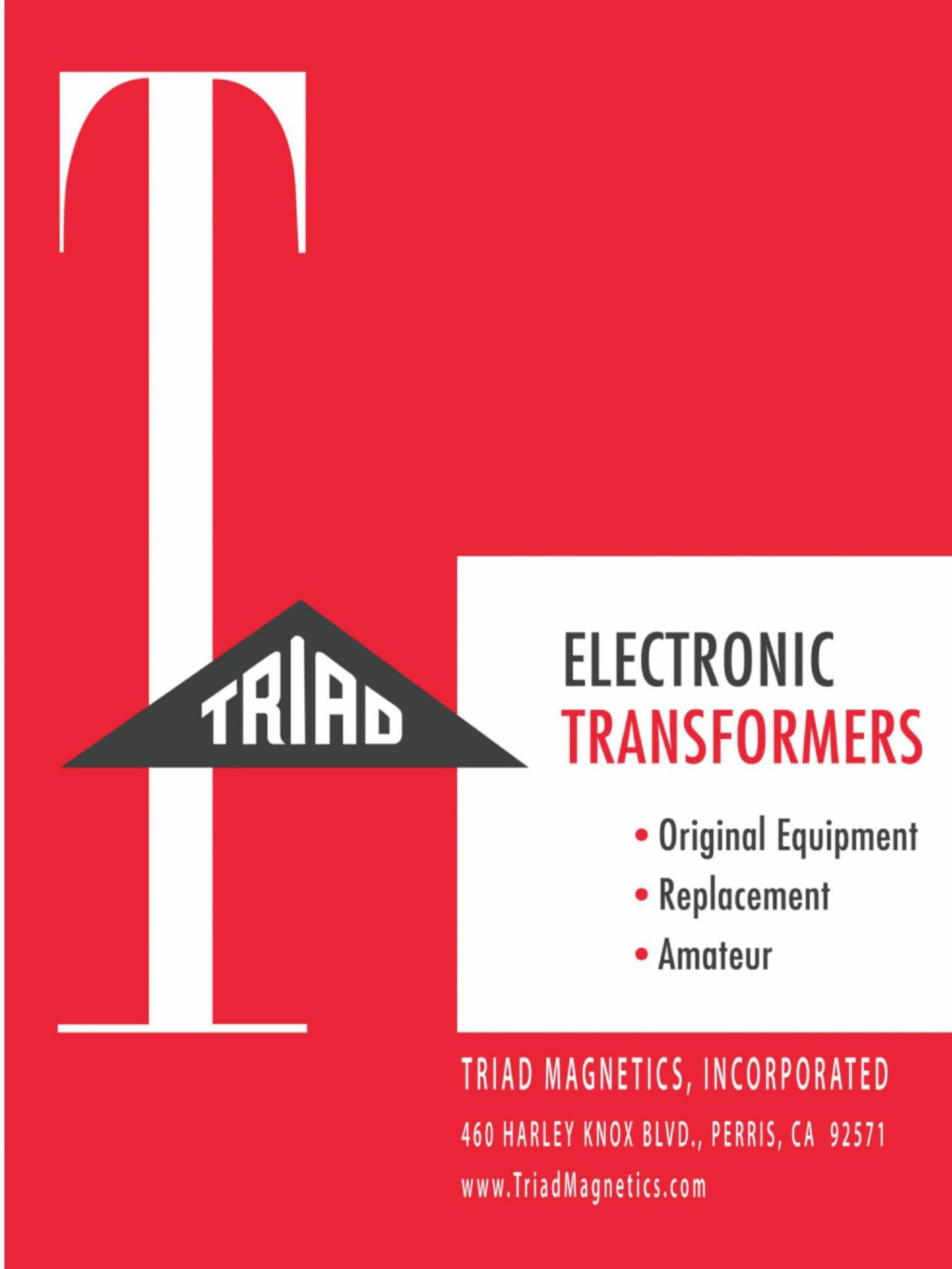
March 22-24, 2019
Salon Audio Montréal/Montréal Audio Fest
Hotel Bonaventure, Montréal, Canada
montrealaudiofest.org

The Montreal Audio Fest will be celebrating the 50th anniversary of the iconic Woodstock Fest during its 2019 edition from March 22-24, at the Bonaventure Hotel, which has just received an incredible rejuvenation update to celebrate its 50th too. A "playlist" among the best albums of 1969 will be made available to exhibitors, in order to revive this period of Peace and Love that has marked multiple generations worldwide. In short, expect to see elephant-legged trousers, flowers in the hair, round glasses, and signs of peace and love all over the place during this memorable event. The Montréal Audio Fest brings in more than 5,000 visitors who want to hear, see and touch the more than 300 brands of audio products available under this one roof. It's the biggest get-together of audio-video industry manufacturers, distributors, and retailers in Canada.

April 2-5, 2019
Prolight+Sound Frankfurt
Musikmesse
Frankfurt Messe Exhibition Center,
Frankfurt Germany
www.musikmesse.com
www.prolight-sound.com
www.messefrankfurt.com

Musikmesse and Prolight+Sound 2019: Stronger together! This is the slogan used by Frankfurt Messe to promote the 2019 edition of those two prestigious shows, which could risk being the last ones if the new adopted model doesn't work. After many years of forced separation, the Frankfurt show complex will finally open the new Hall 12 (two levels), enabling Prolight+Sound to grow into a modern space, while Musikmesse returns to the much appreciated Hall 3. More important, both shows will take place again concurrently!

With a new sequence of days—from Tuesday to Friday—the shows will strengthen their profile as a platform for the professional exchange of ideas and information between music, audio and entertainment-industry professionals.



PARTS EXPRESS
AUDIO COMPONENTS & SOLUTIONS

MAKING THE WORLD SOUND BETTER SINCE 1986

OUR NEWEST ADDITIONS!

Over 17,000 Audio Components in stock!

Advanced **transducer technology**

Wavecor has distinguished itself by crafting high-quality speaker drivers and components – which is why Parts Express is proud to be their sole U.S. distributor for speaker drivers. They take all the standard scientific measurements and pay close attention to the results, while optimizing every parameter. They are crafting the best sounding speaker for you and your project.

Innovators **by tradition**

LaVoce is a dynamic, fast-growing speaker manufacturer specializing in low and high frequency transducer design for the pro audio market. From its humble beginnings in 2008, LaVoce has become the supplier of ODM and OEM loudspeakers for many top audio brands in the industry. The true allure of LaVoce is Italian designed speakers at a price anyone can afford.

Sound made **Powerful!**

Through groundbreaking innovation and worldclass knowledge, ICEpower develops and delivers superior sound solutions for professional audio industries and end users all over the world.

They're everything you need for a bass or guitar amplifier, a subwoofer, an active loudspeaker or a high-end amplifier. In short, get ready to rock'n roll.

SAVE $10 off $100 order!
Code: AX12TEN18 Expires 4/30/2019

parts-express.com
1-800-338-0531
*Custom OEM inquiries welcome