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ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Test & Measurement Focus

Market Update

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Finding the Right Solution.**

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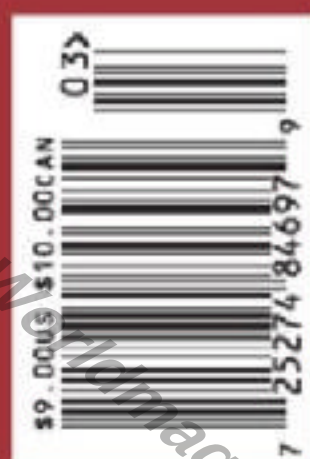
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Driving Audio ADCs

and Active Filter

Design Options

By Michael Steffes





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to your
SIGNAL CHAIN

Analog Gain Stage

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Wanted! Affordable Lab-Grade Audio Measurements

Among the audio test and measurement solutions featured in this month's Market Update, the new Audio Precision APx500 Flex audio analyzer stands out, as it was designed to support the use of APx audio measurement software with ASIO-capable third-party audio interfaces or sound cards. Since Audio Precision (AP) made the announcement, it seems as if the entire audio industry has started to rethink the way it deals with these critical applications, which means that the company was successful in launching the concept. And this certainly wasn't an easy decision for the recognized market leader in audio analyzers and audio testing equipment.

First and foremost the new APx500 Flex allows manufacturers to cost-effectively deploy measurement capabilities in their production lines at a much lower price. And by authorizing Audio Precision's APx500 measurement software to operate independently of an AP hardware analyzer, the solution—which is still far from being the most affordable there is—closes some gaps with even some of the more affordable dedicated hardware/software analyzers, highlighting the importance of professional software in this domain.

But it also revives the entire debate about what we can and cannot do with more affordable tools. As Stuart Yaniger frequently comments in his articles (*audioXpress* readers will remember his series of articles about the basics of using sound cards in audio measurement systems—all of which are now available online at audioXpress.com), there are many circumstances when the cost of a lab-grade system is not justified. And increasingly, there are audio enthusiasts, developers, and a growing community of DIYers, devoted to measurements and simply using free measurement software and very affordable hardware—which is also on the rise.

In fact, we explored many of those more affordable hardware options to include in this month's Market Update feature, but ultimately decided that more time was needed for evaluation. Some of the systems are not even commercially available, others are not yet fully operational according to claims, or simply because we've decided that we would rather investigate those solutions with our own dedicated reviews.

Stuart Yaniger and Jan Didden will have a word to say about these trends throughout this year, and hopefully that discussion will begin in our next issue.

The most important thing is that—as more and more people seriously consider all the options available for audio measurements, with all the related challenges and limitations, and the need for reliable calibration—the more attractive the fully featured solutions seem to be. And no doubt, the bridging alternative of the APx500 Flex audio analyzer becomes even more appealing as our work requirements increase and budget restrictions apply.

As one works on serious research, it becomes painfully obvious how irrelevant sometimes that discussion can be, when no compromise is acceptable and the greatest challenge is discovering ways to measure what was never before considered.

That's what I discovered when I had the opportunity to meet and interview Bruno Putzeys, Lars Risbo, and the Purifi Audio founding team at the new Danish audio company's first presentation in Munich last year.

Faced with the company's bold claims, always complemented with measurements, I couldn't help asking how is it that they were able to reach so many ground-breaking improvements in their research in both the signal and acoustical domains. Not surprisingly, they explained their work consisted of perfecting the process of deriving theoretical hypothesis through simulation, followed by measurements, and completing the loop of importing back the measurements into their simulation models until the research was considered conclusive. And for that, they had to build their own measurement systems...

And I truly believe that this is one of the crucial frontiers in research: closing the loop between theoretical models, measurements, and simulation.

As Bruno Putzeys enthusiastically describes: "...once that level of detailed knowledge was in a mathematical model, we knew where to start looking to fix the problem." And Lars Risbo reinforces that idea, "...once you understand the problem, the solution almost presents itself... We are driven from an understanding in closing the loop between our model and measurements."

J. Martins
Editor-in-Chief

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COLUMNISTS

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

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

Mike Klasco is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is a graduate from New York University, with post graduate work in signal processing, and he holds multiple patents licensed or assigned. For the past 35 years, Mike has worked on countless R&D projects for large and small companies. Mike specializes in materials and fabrication techniques to enhance audio performance.

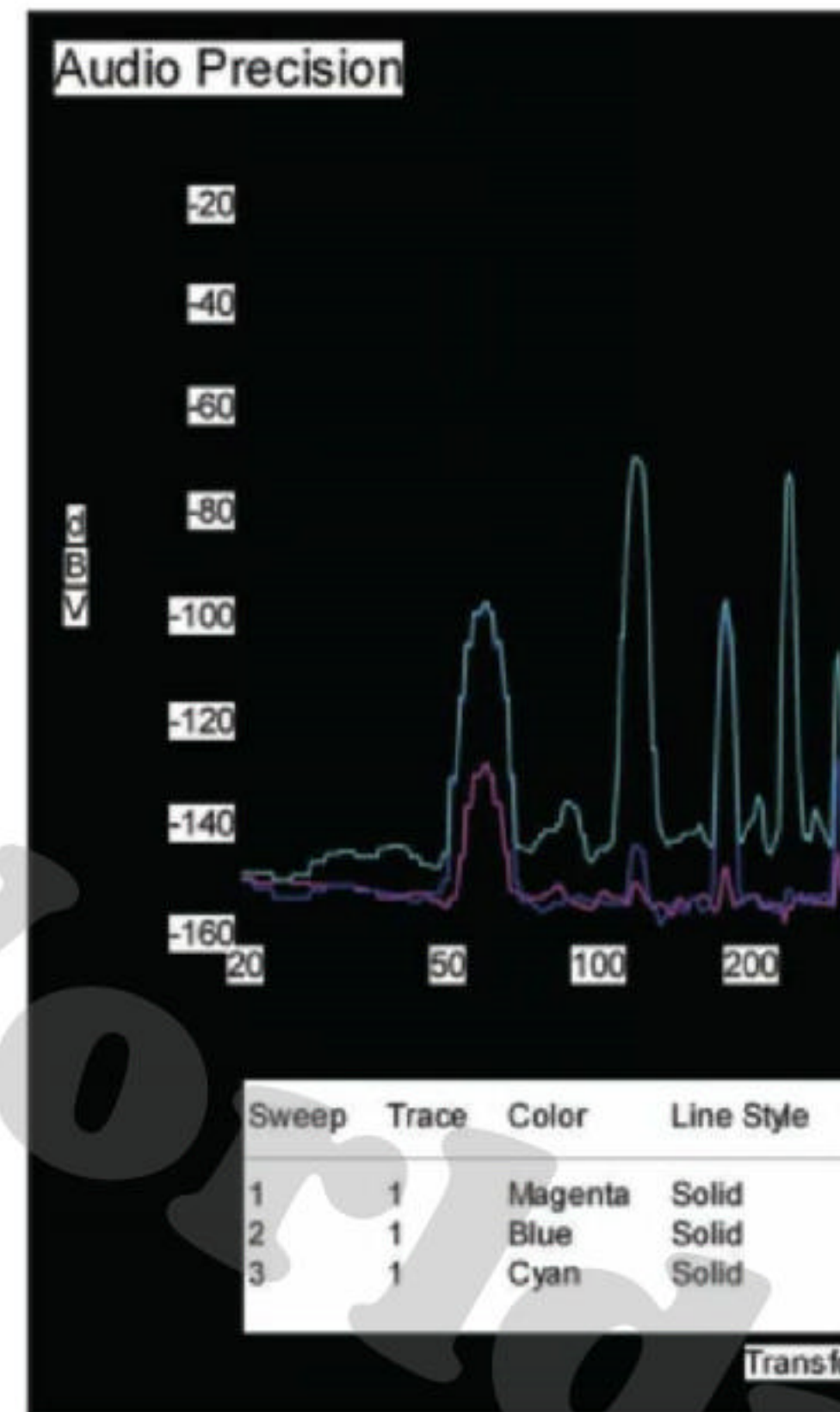
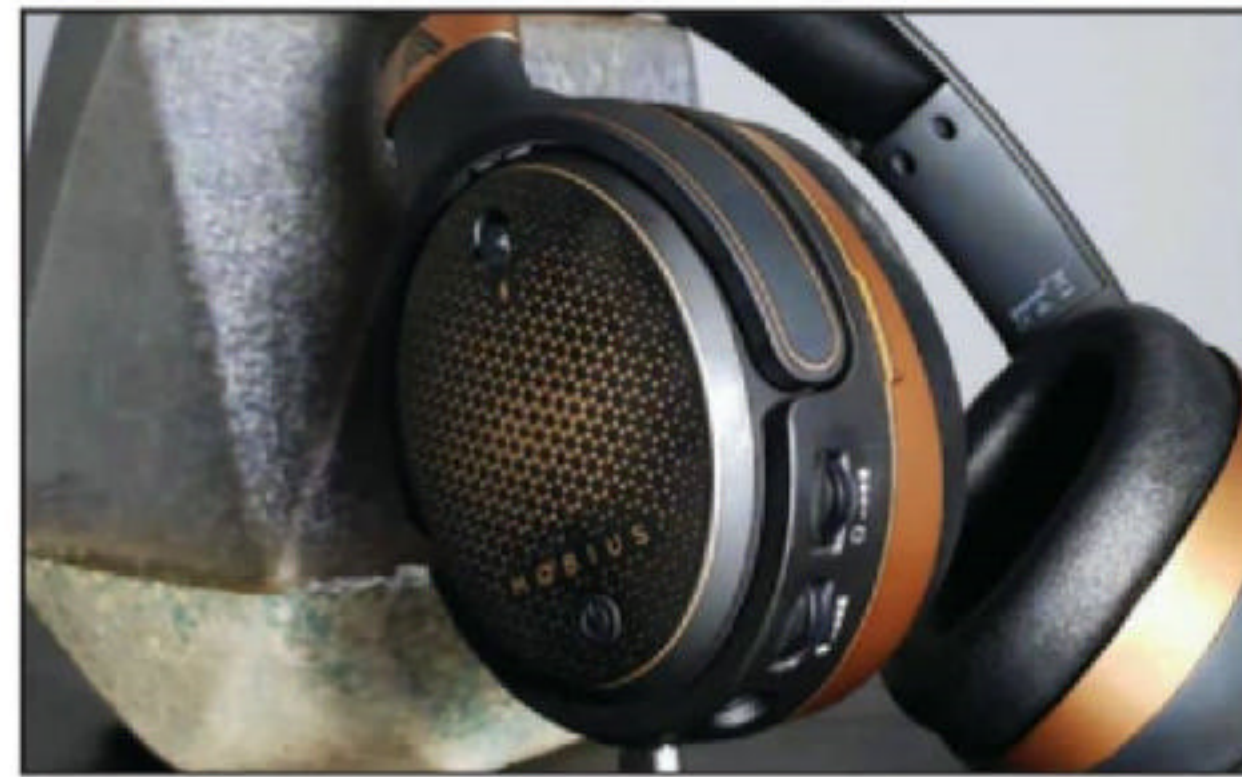
Michael Steffes has worked for more than three decades in high-speed signal path IC development, spanning IC design roles for the BurrBrown high-speed amplifier products, and product launch/application manager roles through Texas Instruments/Intersil. He continues to focus on high-performance ADC interface solutions, DSL/PLC line driver design and optimization, and signal path design optimization tools. In a career spanning five different major IC vendors, he has participated in the introduction of 150 industry leading amplifier products (setting many of the now standard practices in high speed amplifier characterization curves), publishing more than 100 contributed articles and application notes from 1989 onward.

The Audio Voice

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● Market Update—Test & Measurement

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By J. Martins

In this annual Market Update on Test and Measurement, we decided to highlight the latest solutions that are available to the audio industry, including updates to some of the most established measurement toolkits.

● Edition Focus—Test & Measurement

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Rise of the Ultra-Portable Woofer Tester

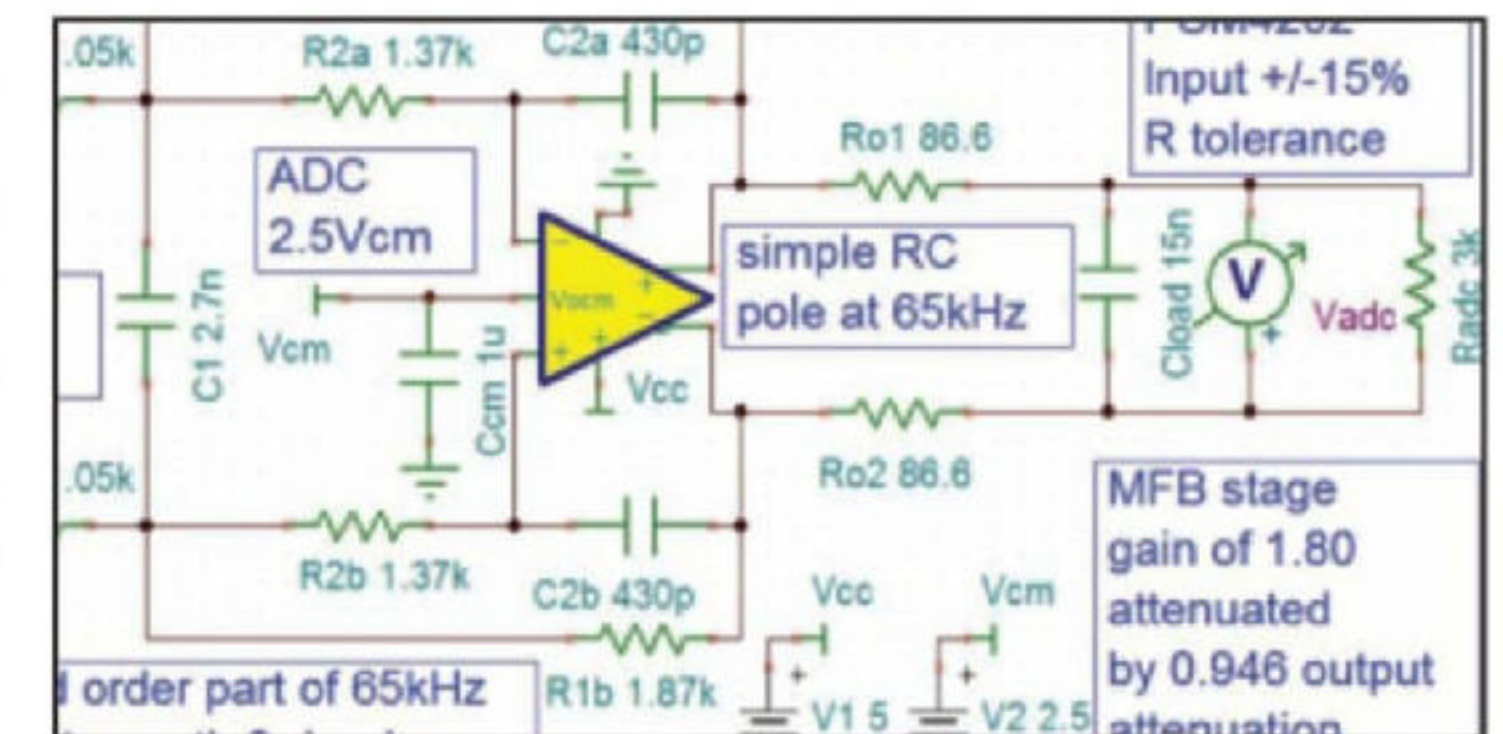
By David Logvin

After a decade of widespread use, the Dayton Audio Test System (DATS) has acquired an impressive reputation for fast and accurate testing. Now, Dayton Audio has taken the audio test system and improved it in almost every way. David Logvin reviews the latest iteration for audioXpress.

28 Strategies for Optimizing Audio Production Tests

By Dan Knighten

This article focuses on several key challenges found in the production and manufacturing test of audio products, and shares strategies for optimizing the measurement system to address said challenges.



● Features

42 Audeze Mobius Headphones Reviewed

Almost Essential, Almost Wonderful

By Luke McCready

High-end planar magnetic headphone manufacturer Audeze breaks out of the mold with the Mobius: a wireless Bluetooth headphone capable of LDAC high-resolution playback, virtualized surround sound, and a number of surprises Audeze continues to invent. We explore what this headphone can do, the use-cases for its unique features, and the limitations of this boldly conceived product.

56 Powering Your Circuits (Part 2)

AC Power Supplies

By Ed Simon

Ed Simon has provided audioXpress with a six-part article series that details ways to power circuits. After reviewing battery power in Part 1, Ed now tackles all aspects in AC power supplies, starting with transformers.

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Finding the Right Solution



For this special edition of *audioXpress*, focusing on Test and Measurement, we decided to highlight the latest solutions that are available to the audio industry, including updates to some of the most established measurement toolkits.

By

J. Martins

(Editor-in-Chief)

All the selected solutions for this market update 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.

There's no denying that the use of audio measurement solutions continues to expand, as audio engineers increasingly rely on affordable kits to continuously monitor their system's audio signal response and acoustic environments. While reference test and measurement systems gain new features and lab-grade solutions become increasingly more sophisticated, a growing number of music enthusiasts, DIY audio builders, and new generations of audio developers have started to rely on a combination of affordable hardware and software created by those who recognize the importance of measurements and have developed an interest for objective evaluation. And not surprisingly, even the most reputed manufacturers of scientific test equipment and reference measurement solutions understand the need to reach out to this expanded market and are creating more affordable solutions that allow anyone to benefit from their existing tools and know-how.

All those innovations are needed, make sense, and are good news for the audio industry. Because it is important to measure what we hear.

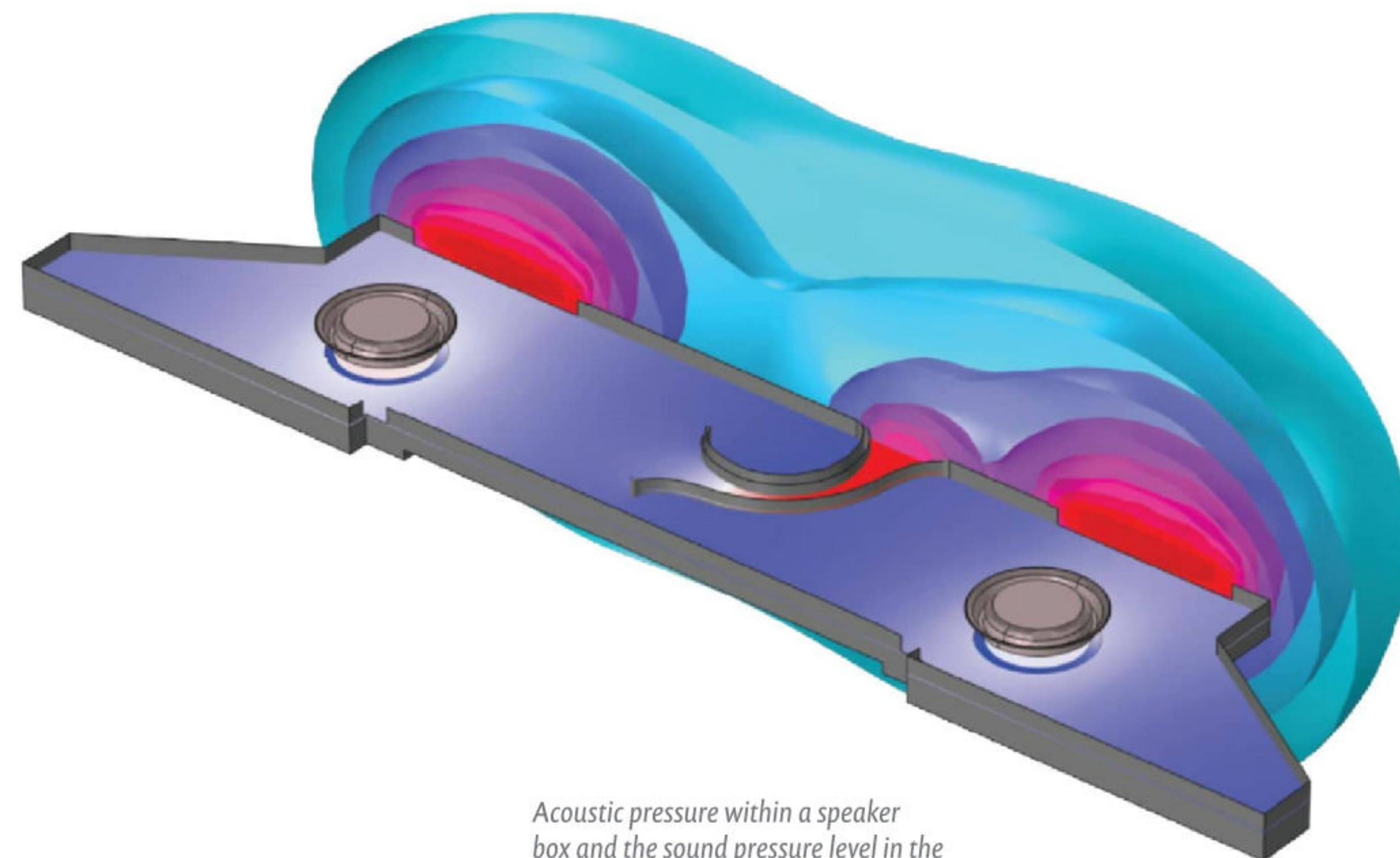
At the time of writing of this article, the consolidated audio industry in general is consistently

growing and it is forecasted to keep expanding at a compounded average of 5% for the next five years. This is an exceptional index. One that provides enough confidence in the manufacturing sector to justify serious investments in research and development activities as well as industrial capability. And with the recent trend of expanding production capabilities in and outside China, in the largest markets for audio products—consumer and professional grade—the rate of investments is much greater for the foreseeable period.

A direct consequence of this growth and expansion is the investment in audio test and measurement equipment. From simulation and design systems, to analyzers and test systems for development, to quality control and production testing. But there's another reason to expect more spending in audio test and measurement equipment, which is directly related with technology evolution.

Precisely a year ago, Audio Precision (AP) announced the introduction of its APx555 B Series two-channel audio analyzer, the successor to AP's original APx platform. This second-generation APx555 introduced the highest possible performance for audio analyzers in the industry. As Stuart Yaniger described in his *audioXpress* review of the B Series, "Part of being the gold standard is recognizing that markets and needs for test and measurement expand and shift, and effectively respond to those changes." And the industry did introduce new challenges in a new generation of much-improved electronics, such as amplification; analog-to-digital converters (ADCs); digital-to-analog converters (DACs); and

*Simulation + testing =
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Acoustic pressure within a speaker box and the sound pressure level in the surrounding domain.

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signal processors, “which can have distortion so low that existing instruments have difficulty resolving it,” as Yaniger illustrates. Effectively, the APx555 B Series of instruments sets a new standard with the lowest distortion floor available, but it is also significant that its previous standards are now almost matched by much more affordable solutions launched in the past year or so, some of which are almost at the “USB sound card and PC software” level.

Of course, Audio Precision APx and the Stanford Research SR1 analyzers continue to offer a golden standard for doing the heavy lifting and providing reliable measurements, and any lab still needs a combination of reliable instruments, from signal generators and oscilloscopes to simple multimeters. But everywhere we look, from measurement microphones to dedicated measurement instruments for headphones

and earphones, to signal conditioning and data acquisition front-ends, there are significant, verifiable improvements that continue to respond to the evolving needs of the industry.

As I previously wrote on this topic, the people who daily provide the research on which the audio industry evolves, depend on test and measurement tools to generate the data to measure and demonstrate their areas of work. Meanwhile, the test and measurement companies must follow those cutting-edge research challenges very closely to develop and provide the tools upon which that research depends. The diversity of the field work needed to understand continuously improving areas of research and convert that knowledge into practical solutions that can verify and certify all the generated data, is precisely what makes test and measurement instrumentation so interesting. 📺

Listen SoundCheck 17

In 2019, Listen announced the release of SoundCheck 17, the flagship Windows/Mac software, which now includes many new features to simplify multi-channel and voice-controlled testing, such as a new level and cross-correlation trigger, average curve/waveform post-processing functionality, the ability to easily read and work with multichannel WAV files, multiple DC Connect control, and enhanced database options—useful when processing data from a large number of channels.

SoundCheck 17 offers increased flexibility in color palettes, a save to MATLAB option, and the ability to recall CSV-formatted text files. The improved color palettes now offer ultimate flexibility in defining colors for backgrounds, grids, cursors, and graph lines. New default color palettes are included, and user-defined palettes can be saved as pre-set files, which will be applied to any new display created. Multiple palettes can be saved, allowing different color sets for different applications.

An important request from many users has also been added—all data including memory list curves and Soundmap (time frequency analysis) data can now be saved to MATLAB for additional processing. Data is saved as a standard MAT file and can be manually or automatically saved in this format.

The new SoundCheck 17 now includes a new cross-correlation “smart trigger,” which offers improved performance for open loop measurements. By using a chirp-based conditioning tone and searching for the exact log sweep frequency, it is more robust and less susceptible to false triggers than simpler level and frequency triggers. This feature requires optional module 2006 (time selective response) to be installed.

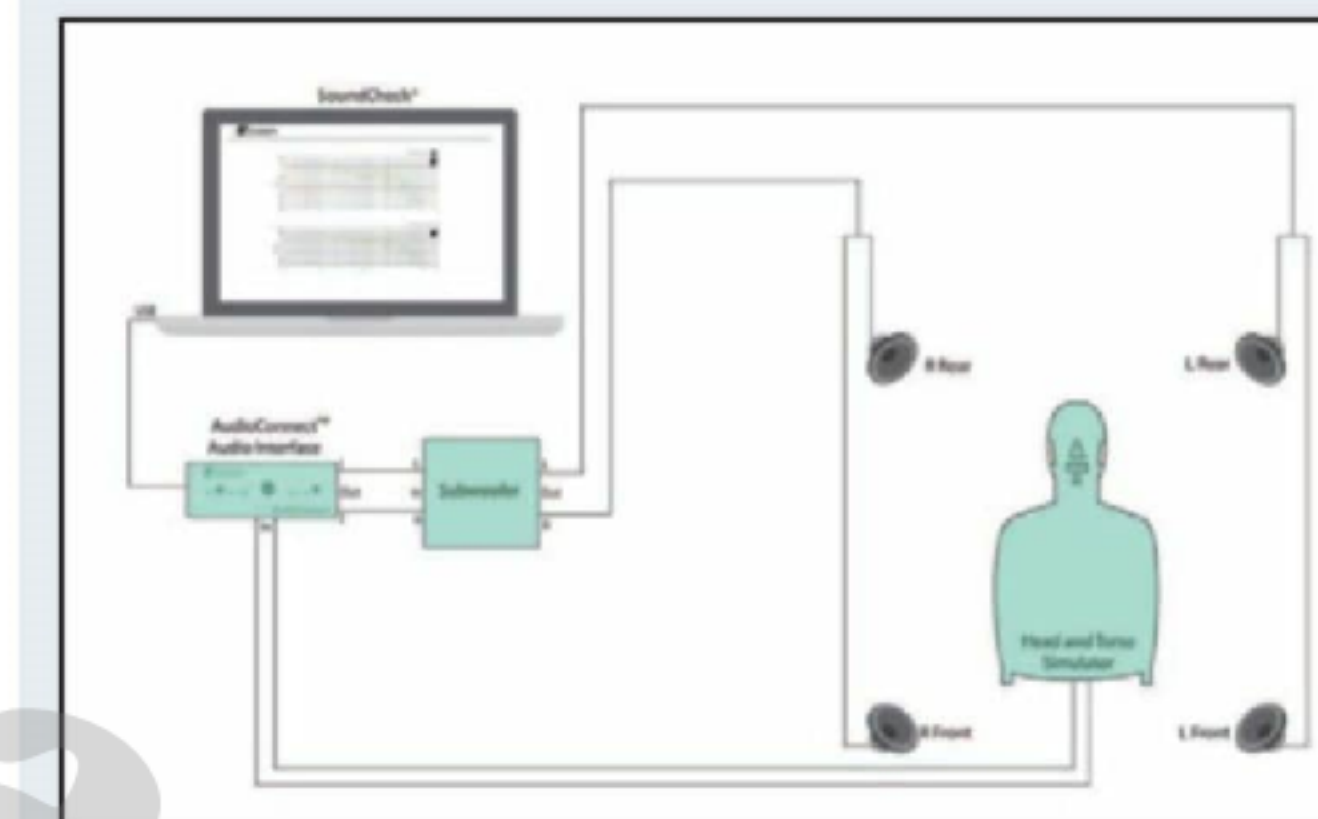
The improved database module on SoundCheck 17 is a powerful data management tool for users testing modern audio products with multiple transducers (microphone arrays, multiple speakers, etc.), as these frequently generate large volumes of data. It is also extremely valuable for production line users who

need to retain complete test data for full traceability.

Enhancements in version 17 make it four times faster and three times more space-efficient than previous versions. Such speed is achieved by saving to a binary format; SQL and Python tools are provided to facilitate data extraction so that detailed analysis and graphical representation may be implemented in any Business Intelligence (BI) system of choice (e.g., Tableau, Qlik, etc.).

Testing of multichannel devices such as microphone or speaker arrays is also much easier and faster as SoundCheck 17 now reads multichannel WAV files from the memory list, signal generator, and stimulus steps; there is no need to de-interleave the files in a separate utility before importing into SoundCheck. When opening a multichannel WAV file in the Memory list, each WAV file channel is named using the file name and channel number, and all channels are automatically grouped together.

Also for testing multi-channel devices, multiple DC Connects can now be controlled and used for data acquisition within



A new SoundCheck test sequence calibrates a 4.1 speaker array to conform with the ETSI ES 202 396-1 background noise generation standard.

SoundCheck. Separate DC Connect message steps can be used to independently configure each DC connect (output modes, signal level, polarity, voltage level, current level, etc.).

Average Curve/WFM is a new post-processing function that enables users to obtain the average curve (or waveform) of a selected group of data in the Memory List. It is particularly useful for power averaging selected curves, averaging curves from different spatial positions (e.g., microphone arrays), and complex averaging of multiple measurements with background noise.

SoundCheck 17 now supports the new aptX HD codec for high-resolution Bluetooth testing and is also supported in the newest version of the BTC Bluetooth Interface, which is fully integrated with, and controlled by, SoundCheck.

ETSI Standard Background Noise Generation System

Listen also launched the European Telecommunications Standards Institute (ETSI) standard background noise generation module, a SoundCheck test sequence that calibrates a 4.1 speaker array to conform with the ETSI ES 202 396-1 standard. This provides an equalized, calibrated playback solution to stress devices in a standardized and repeatable way. With a purchase price of just \$5,000 for the test sequence, this offers an extremely economical alternative to a conventional \$20,000+ stand-alone background noise generation system.

The test sequence has many applications, for example evaluating ANC, noise suppression, SNR optimization of microphone arrays, beamforming directionality studies, and more. It is particularly useful for voice recognition testing as the test sequence may include loops to incrementally increase the volume or change the noise and repeat the test until the voice is no longer accurately recognized.

www.listeninc.com

Updated Bluetooth Interface

The updated BTC-4149 Bluetooth measurement interface, available worldwide exclusively from Listen, Inc., together with its SoundCheck 17 software, now supports aptX HD and aptX-Low Latency Bluetooth protocols for accurate measurement of high-definition and low-latency Bluetooth devices. This enables manufacturers of Bluetooth audio devices to accurately test the audio of the latest high-performance Bluetooth sink and Bluetooth source devices, making the same measurements as on traditional audio devices.

The updated Portland Tool & Die BTC-4149 Bluetooth measurement interface now has a new, larger color screen for improved ease of operation, among other improvements. Since 2014, Portland Tool & Die has had an exclusive worldwide distribution deal with Listen, Inc., providers of the industry standard SoundCheck audio test software.

Also available is a lower cost production tool, the BQC-4149. This tests Bluetooth Sink devices only, and offers only the standard aptX, CVSD and mSBC, along with more limited control of Bluetooth

parameters. This model has been updated to use the same chipset as the BTC-4149 to ensure accurate production line replication of R&D tests for those companies who use both interfaces.

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Spectral Measurement

Prism Sound's Test & Measurement division has re-launched itself specializing exclusively in audio test and measurement solutions and trading under the new name of Spectral Measurement. Prism Media Products, Ltd. will now focus exclusively on developing new audio test and measurement solutions.

Prism Sound has developed high-quality audio test and measurement solutions since 1987, including the groundbreaking dScope Series III audio analyzer, well-established in many professional audio, automotive electronics, telecommunications, and consumer audio manufacturing operations globally. Graham Boswell, Prism Sound's founder and now managing director of Spectral Measurement, leads the new venture, together with Ian Dennis, Technical Director and co-shareholder.

The launch of Spectral Measurement coincides with the release of the new and highly competitive dScope M1 family of PC-based audio measurement solutions and the range of innovative loudspeaker test chambers developed by Hill Acoustics that replace large and expensive anechoic chambers with small and inexpensive calibrated enclosures. The new products are aimed at developers, manufacturers and maintenance providers in audio electronics and electro-acoustics.

The dScope M1 is a new addition to the range of hardware interfaces for the Spectral Measurement dScope Series III audio test and measurement software. This interface provides a comprehensive and low-cost solution for audio electronics and transducer testing.

TTC Loudspeaker Test Chambers

Meanwhile, Spectral Measurement has signed a licensing deal with UK-based Hill Acoustics, enabling it to manufacture and sell the company's Loudspeaker Test Chambers. Hill Acoustics' LTC range includes a selection of Tetrahedral Test Chambers (TTCs) as well as other geometries that provide test and measurement solutions for loudspeakers, providing an effective, low-cost solution that removes the need for large anechoic chambers.

The TTCs are compliant with the new international standard



Spectral Measurement is the new name adopted for the former Prism Sound Test & Measurement Division, led by Graham Boswell, Prism Sound founder and managing director of Spectral Measurement, and Ian Dennis, Technical Director and co-shareholder.

IEC60268-21. They are also available in various sizes, as well as non-tetrahedral custom designs that enable, for example, easy 360° polar response measurement.

Loudspeaker measurements have traditionally been difficult due to the requirement for a very large space to avoid room-related effects and the need for the device to be precisely located under test and measurement microphones. Variations of 6 dB or more in responses measured at different times are not uncommon as a result. Hill Acoustics' LTC range solves these problems inexpensively by providing a precise jig for the speaker and microphone and by calibrating the enclosure so that the equivalent free-field response can easily be calculated.

www.spectralmeasurement.com



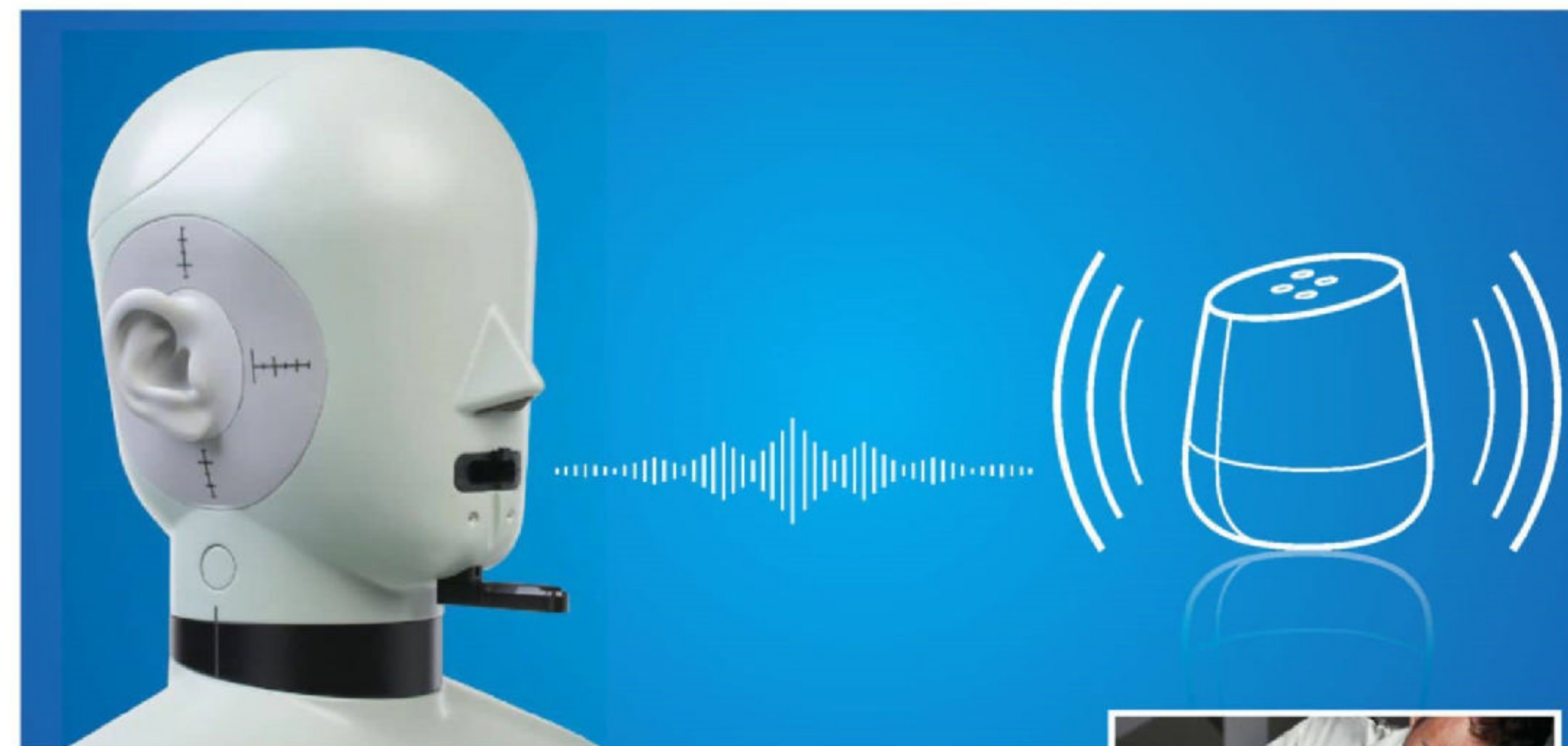
Three examples of TTC Loudspeaker Test Chambers, which deliver consistently stable, convenient, and affordable loudspeaker measurements.



The dScope M1 is the latest addition to the range of hardware interfaces from Spectral Measurement, providing a comprehensive and low-cost solution for audio electronics and transducer testing.

MATCHING THE CHARACTERISTICS OF HUMAN SPEECH AND HEARING

REALISTIC TESTING OF VOICE-OPERATED DEVICES



To improve speech intelligibility and sound quality in voice-operated devices such as smart speakers, testing methods must accurately match human characteristics for speech and hearing.

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www.bksv.com/smartspeaker

Meyer Sound M-Noise

Meyer Sound has introduced M-Noise, a new test signal to promote standardized measurement of a loudspeaker system's maximum linear output. A mathematically derived test signal that effectively emulates the dynamic characteristics of music, according to Meyer Sound, M-Noise enables a more accurate measurement of a loudspeaker system's linear peak SPL in any application requiring reproduction of musical content.

The mathematical formula for generating M-Noise was derived following extensive spectral analysis of a variety of music program material. In particular, the analysis measured the varying crest factors in music, and how these measurements compared to pink noise (crest factor is the difference between the average and instantaneous peak levels of a signal). It was discovered that the crest factors of music and pink noise were similar at

frequencies below 500 Hz, but with music exhibiting a steadily rising crest factor at higher frequencies.

The new test signal delivers an accurate and consistently repeatable way to measure a loudspeaker's linear peak SPL when reproducing music, providing a "real-world limit" for sustained, linear, and full-bandwidth music reproduction—providing the assurance that the system will have the headroom required for accurate reproduction of dynamics without compression.

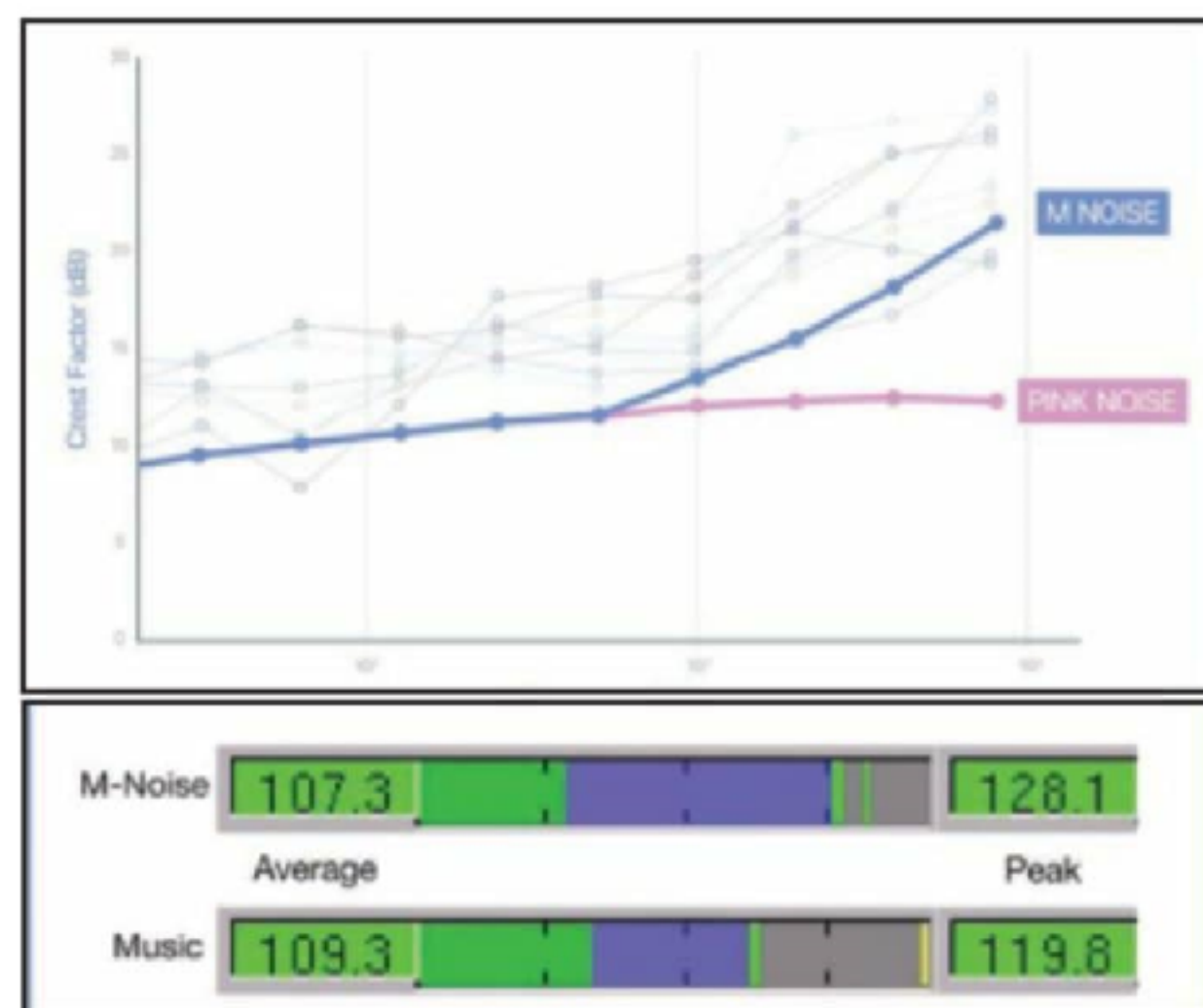
As John Meyer, the company's President and CEO explains, "At Meyer Sound we are constantly pursuing new ways of measuring and analyzing what we experience when we listen to music. By uniquely combining the consistency and repeatability of a test signal with the real-world dynamic complexity of music, M-Noise provides audio professionals a powerful new tool for system calibration."

Additional information and a free download of the M-Noise file are available on Meyer Sound's dedicated M-Noise website, m-noise.org. The file format is 96 kHz WAV with a size of 10.2 MB. Playing time is 53 seconds but playback can be looped indefinitely.

The new M-Noise site includes informative videos and a complete measurement procedure guide. A short overview summarizes the technical solution offered by M-Noise and outlines the development process. For audio professionals who want more background, "M-Noise: A New Test Signal" explains the fundamental concepts behind pink noise, demonstrates the different crest factor characteristics of music and pink noise, and reviews the methodology behind deriving M-Noise. The third video, "M-Noise: Practical Use" describes the measurement equipment (source player, analyzer and measurement microphone) required to effectively use M-Noise, and details the procedure for determining a loudspeaker system's linear peak SPL. The "Real World SPL" video serves as a prerequisite to watch before the two technical videos.

M-Noise can be used to determine the linear peak SPL of any loudspeaker system, regardless of manufacturer, size, or design. However, the accuracy and consistency of results will depend on the test equipment's type and quality, and adherence to the proper measurement procedures.

www.meyersound.com | <https://m-noise.org>



Measurements showing the differences between a measurement with pink noise and a music signal and the M-Noise test signal.

Earthworks M23R Reference Measurement Microphone

Part of Earthworks infinitely matched series of measurement microphones, the new M23R was designed by the New Hampshire-based company to be a reference measurement microphone with an affordable price. This new omnidirectional model from Earthworks is able to sustain extremely powerful SPL levels—up to 140 dB SPL maximum acoustic input—and is extremely stable with temperature variations, making it perfect for multiple acoustical measurement applications, including front-of-house measurements.

Proprietary Earthworks technologies allow for the matching of any microphone within the series, regardless of its purchase date. Each M23R Reference Measurement Microphone is individually tuned and calibrated, and shows the same flat frequency response and sensitivity, matched within ± 0.5 dB across the entire operating range of 3 Hz to 23 kHz, meeting or exceeding Type 1 specifications. This enables companies and individuals to continue to build out their arsenal of measurement microphones over an extended time period and know that each new M23R is matched within ± 0.5 dB to previously purchased M23Rs. This set of features makes the Earthworks M23R an ideal tool for companies that work with multiple teams and are looking for consistent measurements. The M23R is now available and priced at \$599 MAP.



www.earthworksaudio.com

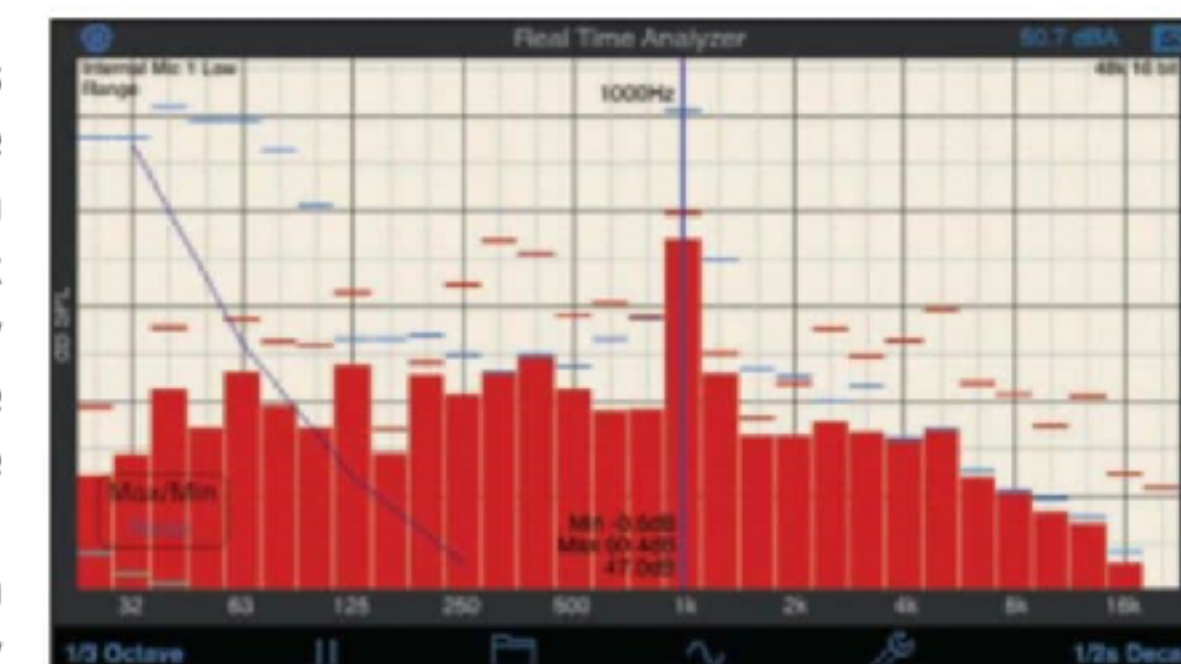
New uPrecisionMic Measurement Microphone from Studio Six Digital

Studio Six Digital develops measurement microphones and audio interfaces that complement the audio test and measurement software and apps available for the iOS platform, and which have already generated more than 1 million app downloads worldwide. Continuously updating these solutions, Studio Six Digital's popular AudioTools suite of test and measurement tools are fully compatible with the latest iOS, iPhones, and iPad models. This includes the latest iPad Pro models, which already use an USB Type-C connector, and are extremely powerful mobile computers—ideal for serious field work.

Studio Six Digital introduced the uPrecisionMic, a USB-C connected precision Class 1 microphone, which can be directly connected to the latest iPad Pro, many Android smartphones, and Mac and Windows computers. The uPrecisionMic uses a metal diaphragm capsule purposely built by ACO Pacific, while the body of the microphone holds a completely redesign of the existing Studio Six Digital precision USB connected microphone. The result meets American National Standards Institute (ANSI) and International Electrotechnical Commission (IEC) electrical and acoustic Class 1 standards and is factory calibrated to ± 0.1 dB using a National Institute of Standards and Technology (NIST) traceable pressure-compensated calibrator. The calibration factor for both high and low ranges are recorded on the factory calibration card in dBFS, so that application software can be calibrated to provide an absolute SPL level.

The useful analog output—a standard 1/8" (3.5 mm) stereo jack, line level—is available for sending audio test signals to the devices being tested. The output level can be set from the device to which it is connected (e.g., on a Mac, from the system preferences panel; or on Windows, the audio output control panel).

www.studiosixdigital.com



The detailed Real Time Analyzer screen in the AudioTools app. Great to use with a touchscreen, allowing to detail certain frequencies or freeze and capture the reading.



A new uPrecisionMic measurement microphone from Studio Six Digital connected to a new iPad Pro via USB-C. The image also shows the latest Sound Level Control, now part of the updated AudioTools app for iOS.

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HEAD acoustics Next-Generation Measurement Technology

HEAD acoustics GmbH is one of the world's leading companies for integrated acoustic solutions as well as sound and vibration analysis. Increasingly focusing on comprehensive solutions for optimizing voice and audio quality, HEAD acoustics introduced 3PASS reverb, a new option for simulating room reverberation.

Users of mobile phones and voice-operated smart home devices have to deal with different background noises and room reverberation that impair their user experiences. This requires realistic performance tests of communication devices, considering not only background noise but also reverberation. With 3PASS reverb, the audio experts of HEAD acoustics have developed a methodology that enables developers and manufacturers to reproduce different room characteristics (e.g., realistic reverberation in the laboratory). The software simulates the reverberation of the room by applying the previously recorded impulse responses to any audio signal in real time.

This new software option for HEAD acoustics' background noise simulation systems, 3PASS lab and 3PASS flex, provides a world's first acoustic environment for realistic measurements. While an artificial head measuring system plays back the direct sound signal, the background noise system covers playback of the reverberation components based on the impulse response from the original room. Examples of impulse responses from various rooms are included as standard in 3PASS reverb.

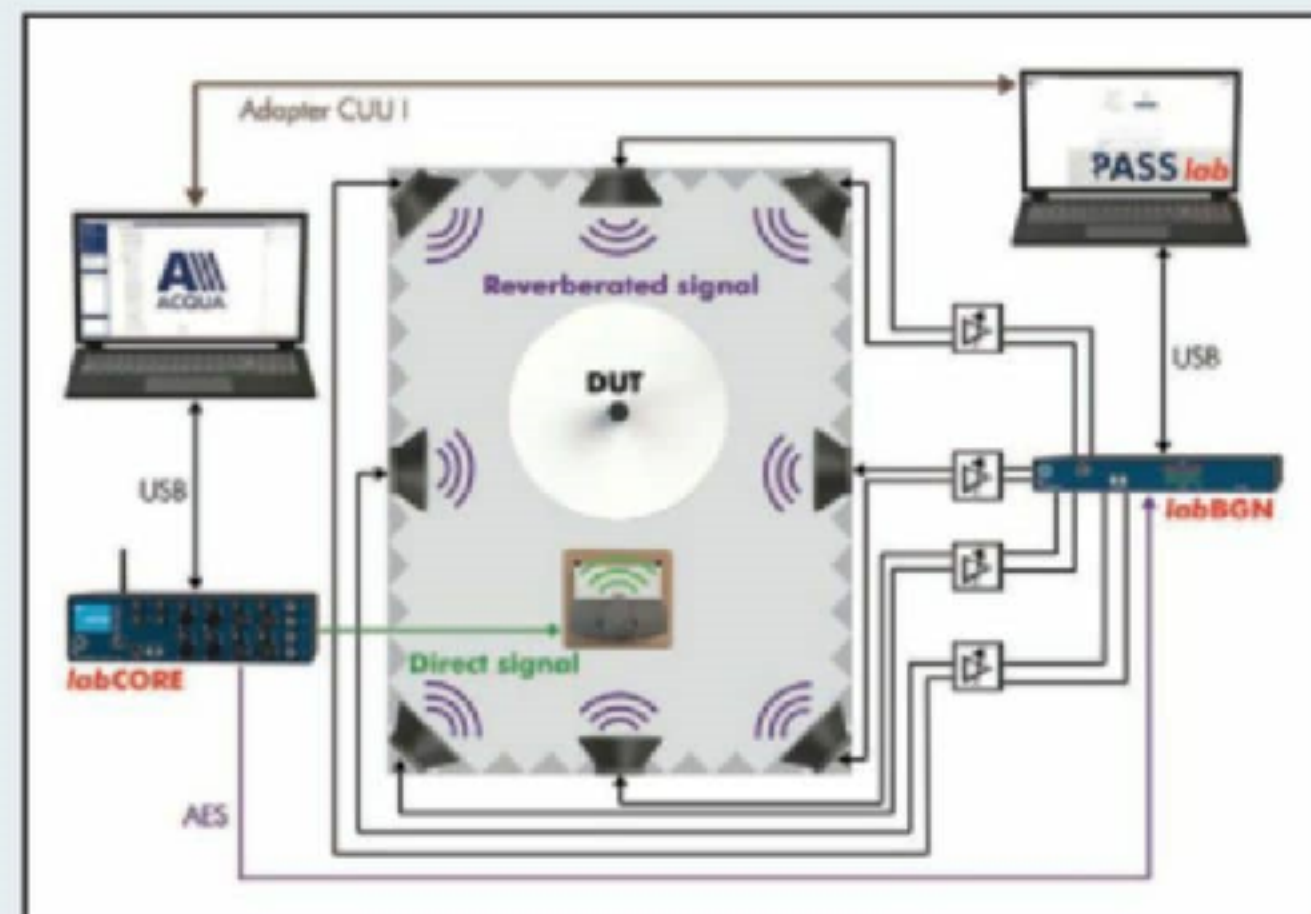
The methodology, invented by HEAD acoustics, has been successfully approved as the ETSI standard TS 103 557 in December 2018.

New Artificial Head for Low-Noise Measurements of ANC Headphones

Measuring very low signal levels such as idle noise of active noise cancellation (ANC) headphones requires highly accurate measurement equipment. For this purpose, HEAD acoustics developed a series of new tools, including a new binaural artificial head measurement system, HMS II.3-LN (Low Noise). The new artificial head is suitable for low noise measurements of close-to-the-ear sound sources, and its ear simulators feature highly sensitive microphones to assure a very low inherent noise of 16 dB SPL(A). This solution complies with the international standard ITU-T P.58 and is based on the standards ITU-T P.57 and IEC 60318-4.

Due to its maximum sound pressure level (SPL) of 149 dB SPL, HMS II.3-LN is also suitable for medium and high level measurements. The ear simulator covers the complete human hearing range, and the head is also equipped with a two-way mouth loudspeaker capable of full-band measurements. The sending transmission range goes from 50 Hz to 20 kHz and the receiving direction from 3 Hz to 20 kHz.

As with all artificial head measurement systems from HEAD acoustics, the new HMS II.3-LN can be optimally controlled via the

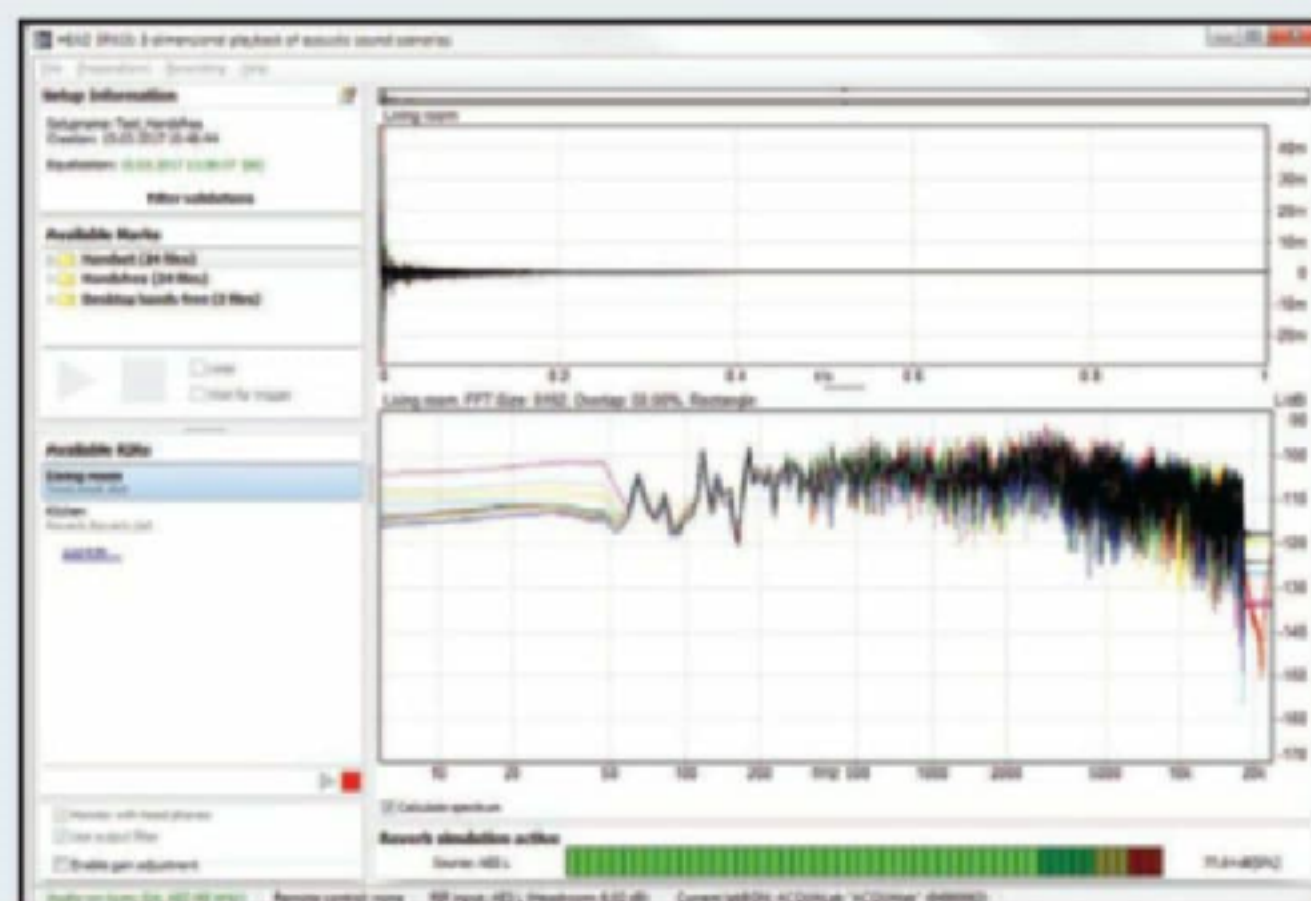


Setup diagram for 3PASS reverb, the new option for the simulation of room reverberation: A head-and-torso simulator in front of the test object plays back the direct sound signal (green waves), the 3PASS lab system plays back the room reverberation (purple waves).

measuring and analysis software ACQUA. Among electroacoustic measurements such as signal-to-noise ratio (SNR), total harmonic distortion including noise (THD+N), and various test signals (e.g., realistic speech signals and logarithmic sine sweeps), the fully automated ACQUA test sequences provide performance tests of passive noise isolation and active noise cancellation.

The HQS-Audio database, which enables headphone manufacturers to establish comprehensive and targeted tests of the audio performance of their devices, now also includes live measurements of ANC headphones.

HQS-ANC-Headset is the fully automated database for ACQUA, enabling testing and optimizing passive noise isolation and ANC headphones, measuring devices with passive (wired), USB-powered and Bluetooth connections. It also includes measurements that



The audio file of the impulse response is displayed in the upper window and the spectrum of the same impulse response in the lower window.

enable users to evaluate general audio quality, effectiveness of talk-through mode and performance in the presence of background noise. The database driven solution allows testing frequency response, SNR, intermodulation distortion (IM), THD and THD+N, as well as conducting Relative Approach analysis.

Finally, HEAD acoustics has also introduced HQS-SmartHome, the latest database for the ACQUA analysis system, which offers a series of tests to evaluate the communication and audio quality of voice-controlled connected devices like smart speakers. The implemented tests take realistic background noise and reverberation, multiple talkers and directivity measurements into account. In addition, HQS-SmartHome contains common electroacoustic tests like echo, frequency response, delay and distortion.



Ear simulators with highly sensitive microphones assure a very low inherent noise of 16 dB SPL(A) of the binaural artificial head measurement system HMS II.3-LN.

Audio Precision APx500 Flex Audio Analyzer Supports Third-Party ASIO Audio Interfaces

Audio Precision (AP) has launched a new class of test and measurement solutions designed to support the use of APx audio measurement software with ASIO-capable third-party audio interfaces or sound cards. The new APx500 Flex audio analyzer authorizes Audio Precision's APx500 measurement software to operate independently of an AP hardware analyzer, with licensing controlled by an APx500 Flex Key.

The Flex Key can be paired with ASIO-capable third-party audio interfaces to create a cost-effective solution for a variety of acoustic and electrical test applications where hardware performance requirements are secondary to test system price—such as production line test of speakers, headphones, and microphones.

But the APx500 Flex solution also enables users to leverage test development done in the R&D cycle through the ability to import project and template files developed for AP hardware analyzers, thereby reducing test development time and maintaining consistency in test methodology across an organization. And

APx500 Flex also makes for a powerful, portable audio measurement system for application engineers and QA technicians on the go.

The new APx500 Flex solution was introduced in conjunction with the release of APx500 software v5.0.2, which provides improved ASIO support, including the calibration of inputs and outputs in V, Pa, and Fs, automatic or manual delay compensation for compliant ASIO interfaces, and support for generator trigger, and acoustic and impedance measurements.

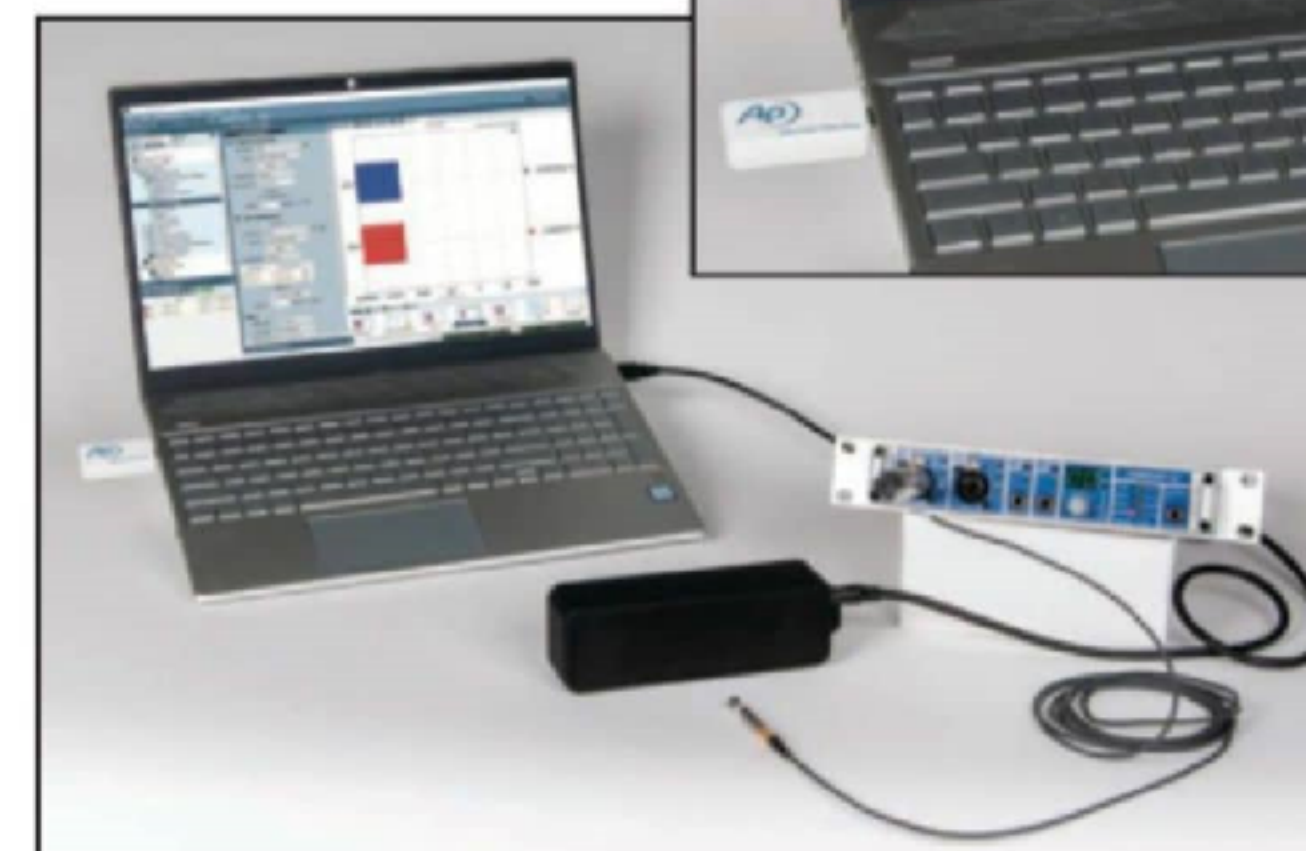
Configuration and Options

In its base configuration, Flex is configured for two channels and includes a core set of fundamental measurements and functionality:

File Analysis, Sequence Mode, Input Signal Monitors (including a Fast Fourier Transform Monitor), Level & Gain, THD+N, Loudspeaker Product Test (including Rub & Buzz), Stepped Frequency Sweep, Pass/Fail, and Signal Acquisition measurement. Configuration options for four or eight channels are also available, along with a trio of Flex Packs that each offer different groups of more advanced measurements. Perceptual audio measurements—including ABC-MRT and POLQA—can likewise be used with Flex. Finally, for the user that needs the base configuration plus just one more measurement, APx500 Flex has an à la carte menu of measurements from which to choose.

Any ASIO-enabled audio interface should suffice with APx500 Flex. However, Audio Precision has verified the compatibility of three different audio interfaces—the RME Fireface UC, Lynx Aurora (n), and Lynx E22—and configuration templates for these devices are included in APx500 software v5.0.2. Each of these verified compatible interfaces offers quality analog I/O (<90 dB THD+N, 192 kHz SR) and stable drivers with consistent delay.

The APx500 Flex audio analyzer is available now, with the two-channel base configuration retailing for \$3,000 USD. As with all other new APx analyzers, APx500 Flex ships with version 5.0 software and one year of software maintenance, effectively licensing that Flex Key for APx version 6.0 when it is released, as well as any minor releases that occur between versions 5.0 and 6.0.



The APx500 Flex audio analyzer combines the power and versatility of APx500 measurement software with third-party audio interfaces to deliver an audio test solution for cost-sensitive production test applications.

New Brüel & Kjær Type 4988-A 1/4" Pressure-Field Measurement Microphone

Brüel & Kjær has introduced a new miniature microphone ideal for high-precision acoustic measurements tests within confined spaces—the Type 4988-A. A mini, yet mighty microphone, this new 1/4" low noise, prepolarized, all titanium microphone was designed to help engineers and acousticians achieve accurate measurements in everyday tasks. The new Type 4988-A is flat from 20 Hz to 20 kHz (± 1 dB) and can be used close to hard reflective surfaces, or very close to sound ports of an audio device (near field).

The Type 4988-A design is also highly resistant to corrosion and suitable for a range of acoustic tests, even in the presence of magnetic fields. It's practically immune to temperature and humidity changes; this extreme stability, combined to a minimal spread in frequency response between individuals make it ideal for integration in a production line or for general audio measurements.

The 1/4" low-noise (29 dBA inherent noise) microphone is optimized for use in



pressure-field applications such as use with couplers, measuring close to audio device sound ports or flush-mounting. The microphone also works as a general measurement device when the operator makes use of the inclusive electronics corrections; these include response at different angles of incidence in a free field, corrections in a diffuse field or grid influence.

The microphone has a laser-welded diaphragm and is paired to a 1/4" constant current line drive (CCLD) preamplifier. This connects to CCLD input conditioning, ensuring all measurements meet the industry standard IEEEP1451.4 V1.0. Recommended applications include measurements requiring an EN/IEC 61326 EMC-compliant product, measurements in small acoustic chambers and test boxes where reflections occur, measurements in unpredictable sound field conditions, and use as a reference microphone for tuning of microphone arrays and active noise-cancelling systems.

www.bksv.com/TheMic

NTi Audio Room Acoustic Reporter Software

The Room Acoustics Reporter PC software from NTi Audio, in combination with the company's DS3 Dodecahedron Speaker and the XL2 Sound Level Meter provides the perfect solution for acoustics evaluation of a room—performing reverberation time optimization calculations, and providing comprehensive reporting according to ISO 3382 and DIN 18041 standards. Useful for anyone working on analysis and evaluation of room acoustics, the XL2 and the Room Acoustics Reporter software even enable simulations of acoustic treatment options with absorption data from manufacturers.

For standard-compliant reverberation time measurements, a sound source with omnidirectional characteristics is needed. The sound energy must be uniformly radiated in all directions, and the current building acoustics standards also require a flat-frequency spectrum in the relevant frequency range. The DS3 Dodecahedron Loudspeaker fulfills these requirements, while delivering high sound power and provides a perfect excitation signal for reverberation time measurements.

The XL2 Sound Level Meter provides reverberation time measurements in octave bands from 63 Hz to 8,000 Hz. Since many manufacturers of acoustic absorbing materials specify the material properties in octave resolution, the manufacturer datasheets and values measured by the XL2 fit well together. If a reverberation time measurement



Room Acoustics Reporter results, measured with the XL2 and showing tolerance and material parameters.

is required in one-third octave resolution, the "Extended Acoustic Pack" Option can be installed onto the XL2 at any time. The XL2 automatically triggers on an impulse sound source or a gated pink noise. When recording the reverberation times, it is possible to switch between T20 and T30—these two types of analysis differ in the level reduction required to trigger a measurement.

The RT60 measurements obtained with the Room Acoustics Reporter PC software can also be compared immediately with a tolerance band; which is specified according to the volume of the reverberant space and purpose of the room (e.g., in accordance with DIN 18041). If the measured reverberation time exceeds the tolerance, the software allows for simulation with additional absorbers. The corrected reverberation time is immediately calculated using the Sabine reverberation formula and shown in the graph. This makes it easy to verify what quantity and type of absorbers have to be brought into the room in order to meet the room specifications.

For the simulation described, the sound absorption coefficients of absorbers from various manufacturers are available for download and import into the Room Acoustics Reporter. The list of absorbers is constantly updated. Also, users can define their own materials for a simulation calculation.

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Dayton Audio Test System (DATS) V3

Rise of the Ultra-Portable Woofer Tester

After a decade of widespread use, the Dayton Audio Test System (DATS) has acquired an impressive reputation for fast and accurate testing. Now, Dayton Audio has taken the easy-to-use audio test system and improved it in almost every way when creating the DATS V3 Computer-Based Audio Component Test System (see **Photo 1**). David Logvin has been a user of the solution and reviews the latest iteration for *audioXpress*.

By
David Logvin
(United States)

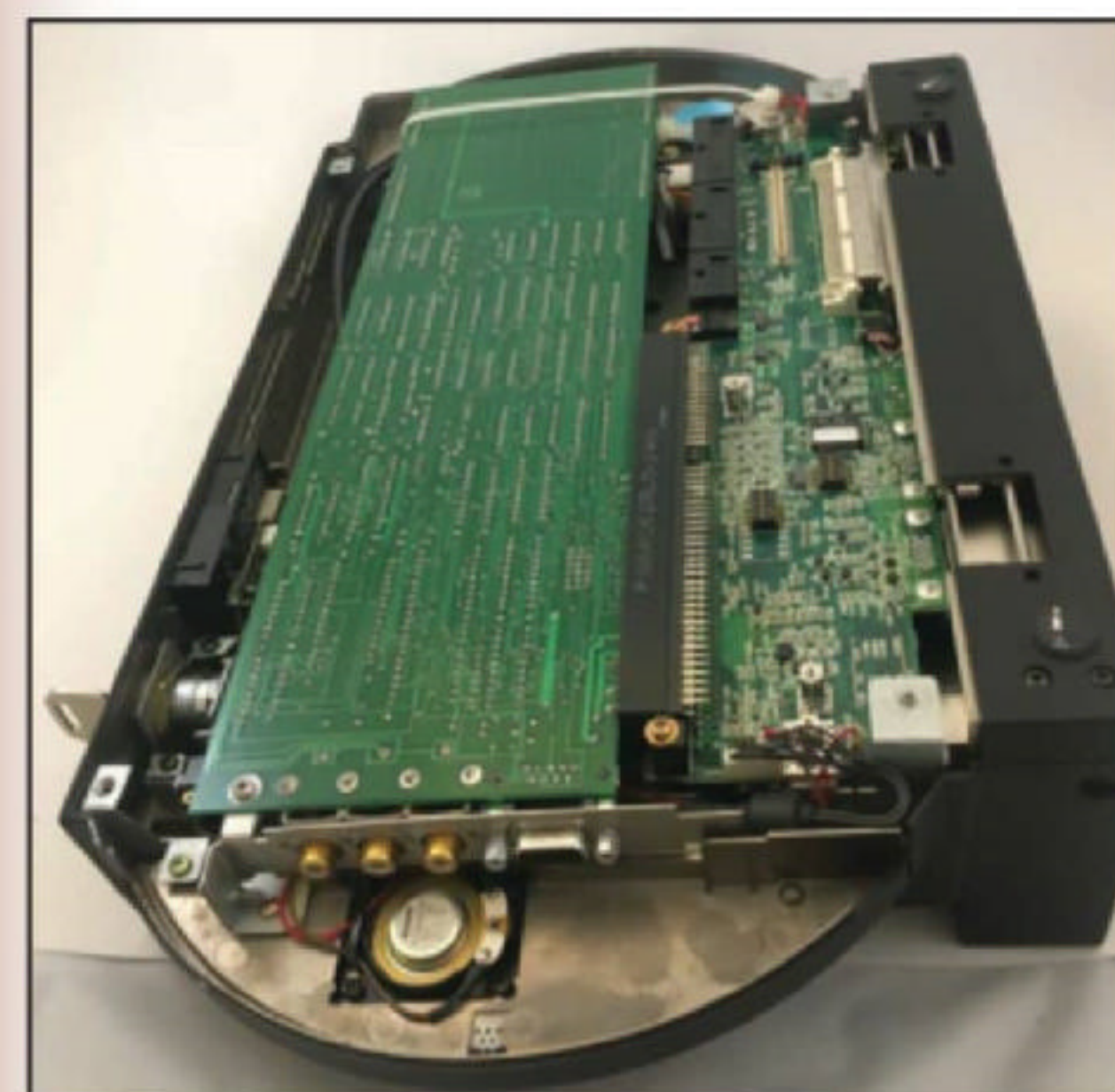


Photo 2: In the 1990s, a MLSSA measurement system was the industry standard method for measuring Thiele-Small parameters. It is shown here mounted in the author's ThinkPad Dock 1 Expansion Unit as an early example of test hardware portability.



Photo 1: DATS V3 is an ultra-portable audio test system that provides a reliable method for collection of Thiele-Small parameters and component measurements.

For as long as I have been working in audio and acoustics, portability has been at the top of the desired feature list for measurement systems. And there has been a steady and noticeable evolution in hardware: from the awkward laptop docking stations making 13" ISA-cards (see **Photo 2**) slightly portable, to the more palatable options like Liberty Audio's Praxis or Dayton Audio's OmniMic, and more recently with Audiomatix's Pocket Clio,^[1] this downsizing trend is quite clear.

While most manufacturers have streamlined and reduced the footprint of their measurement hardware, a few companies have focused on the "ultra-portable" market. These are devices about the size of a compact mobile phone, but with unique measurement capability that goes beyond the limited world of audio measurement on iOS or Android platforms.

Dayton Audio's Digital Audio Test System Version 3 (DATS V3) is one such device. On initial inspection, it appears to be a simple and compact impedance analyzer, but it actually offers a host of additional signal analysis and generation tools. Curiously, despite the "V3" suffix in the name of this device, this is actually the fourth iteration of this design.

The original DATS was dubbed "WT3," and was a simple USB-based hardware device that provided an easy method to measure Thiele-Small (T-S) parameters and impedance curves. It was useful for troubleshooting finished systems and prototypes, as well as designing

new woofer boxes, using the derived parameters for force factor, compliance, and so forth.

The original device was launched as the WT3 Woofer Tester in 2007. The hardware and software are both the design of True Audio's John L. Murphy, a veteran acoustics engineer and author.^[2] A few years later, it was replaced with an identical-looking device, but rebadged and improved as the Digital Audio Test System (DATS). This new version featured a faster start-up time, as well as new signal generator and oscilloscope modes. In 2014, the DATS v2 hardware and software was updated again. This version introduced a new rub and buzz test for advanced diagnosis of discrepant woofers.

Fast forward to 2019, and the latest generation, DATS V3, was released. This new iteration of software and hardware provides improved measurement accuracy, an interesting new linearity test, as well as dramatic improvements to the mechanicals.

Upgraded Hardware

The prior three generations of DATS all featured a simple plastic housing with non-detachable USB cable and test leads (see **Photo 3**). With the new V3 hardware, the device is significantly upgraded with an aluminum extrusion body and endcaps (both with a nice brushed and anodized finish). A USB port and banana jacks now provide for removable cable options and standard test probes for convenience and simple field replacement. Also included with each DATS V3 is a set of rubber feet for table top use, as well as brackets so it can be mounted under a benchtop.

Quick Setup

In prior DATS versions, a separate 1 kΩ calibration resistor was used to calibrate the test system. In the V3 hardware, the resistor is now permanently mounted inside the aluminum chassis and turret connectors provide a reliable test point for making the connections.

Standard procedure would be to first zero out your probes by shorting them to determine the internal probe impedance. Then, measure the reference resistor impedance and store this value into memory and are you are now ready to collect calibrated impedance and Thiele-Small (T-S) data.

With the latest generation, DATS V3 users have access to the following features: T-S parameter measurement, impedance analyzer, RLC meter, rub and buzz failure mode testing, and a unique linearity test unique to this software, as well as the signal generator and oscilloscope modes shown (see **Figure 1**).

T-S Measurement Techniques

Conveniently, DATS V3 offers four different methods to derive T-S parameters: Added (delta) mass, Test box (delta) compliance, Specified SPL, and Specified Mmd (cone mass). The choice of



Photo 3: Three generations of DATS are shown, including the WT3 (left), the DATS V2 (right), and the DATS V3 (center).

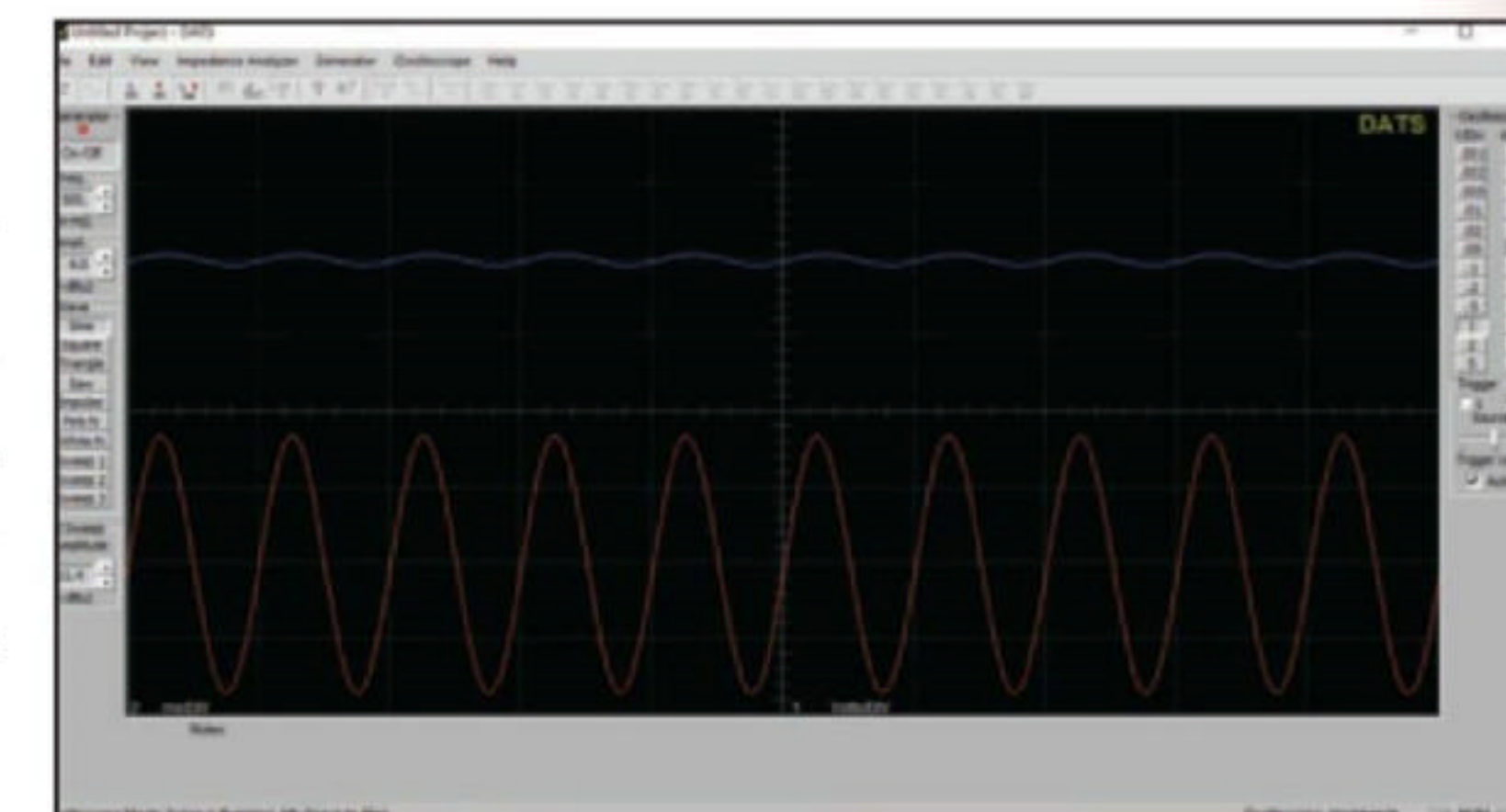


Figure 1: DATS V3 users have access to the following features: the Thiele-Small parameter measurements, an impedance analyzer, an RLC meter, rub and buzz failure mode testing, and a linearity test unique to this software, as well as the signal generator and the oscilloscope modes shown.

About the Author

David Logvin is an acoustics engineer at TDK InvenSense, working in the MEMS microphone division. With more than 20 years of experience, he has designed loudspeakers and other products for Snell Acoustics, Outlaw Audio, ClearView Audio, Cambridge Sound Management, and others. During his 14 years at Snell Acoustics, he developed and designed dozens of high-end loudspeakers and amplifiers, as well as helped to create loudspeaker standards in close collaboration with THX. David then focused his efforts on R&D, creating innovative, award-winning piezo-driven membrane loudspeakers with ClearView Audio (formerly known as Emo Labs). At Cambridge Sound Management, he put his skills in room acoustics to use in the field of sound masking for corporate applications. Most recently, he rejoined his fellow Snell Acoustics alumni, Dr. Joseph D'Appolito, to design several new loudspeakers for Outlaw Audio. In addition to his lifelong love of music and acoustics, David is an active participant in the Boston tabletop game design community. He is a member of the Audio Engineering Society (AES) and the Boston AES Section and lives in Lowell, MA with his family.



Photo 4: The DUT is tested using the Vas Added Mass method (a) and the DUT is tested using the Vas Test Box method (b).

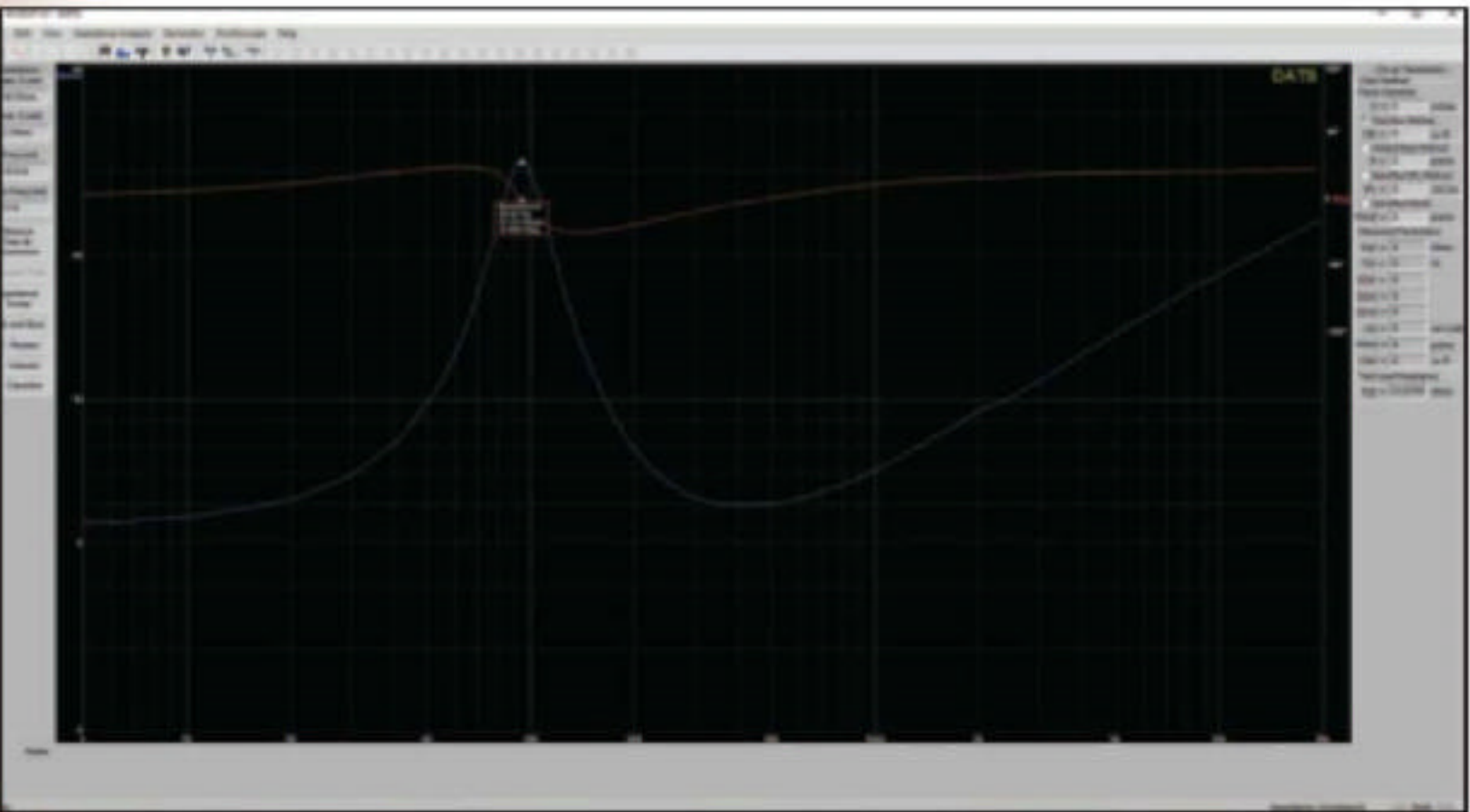


Figure 2: The main software UI is shown where all DATS V3 tests can be accessed and set up. The highly-efficient Impedance Sweep function was used to plot impedance magnitude and phase for the Peerless 850102 SDS.

method depends on auxiliary equipment, user preference, and if any manufacturer specifications are available for the device under test (DUT).

The first two listed methods are traditional and rely upon taking a free air impedance curve and then carefully shifting the resonant frequency using either a delta mass or delta compliance approach. If you have access to a precision grams scale, the delta mass method is often the easiest, although if you already have a sealed test box of a known and suitable volume, then the delta compliance (aka “Test Box” method) can also be convenient.

For delta mass testing, after collecting the free air impedance data, you will add mass (known to at least one significant figure of precision) to the cone that will result in a minimum of a 25% shift downward in resonance. The other classical method for measurement of T-S uses a delta compliance method. After the free-air resonance impedance curve is taken, the DUT is placed into a known volume that will result in an upward resonance shift of at least 50%. The delta compliance method could be particularly useful if you don’t have a known mass handy, or if the installation of the mass loading itself tends to temporarily shift the resonance and you don’t want to wait for the suspension to recover and return to equilibrium (see **Photo 4**).

To investigate the T-S methodologies, I selected a Peerless 4” mid-woofer, the 850102 SDS. The results for the delta mass measurement (and the three other methods, see **Figures 2–4**) are shown and correlate well with the published manufacturer specifications (see **Table 1**). It is worth noting that this particular driver’s F_s is over 20% higher than the datasheet, so if it was going to be used in an actual design, a driver break-in program should be performed to exercise the suspension and ensure that the parameter values have settled to their final values.

The other two methods are “quick-look” and rely upon having access to the most basic of driver parameters (either the nominal sensitivity or the cone mass). Both rely on the accuracy of the manufacturer’s supplied data for either Known Sensitivity (at 1 W/1 m or 2.83 V/1 m) or a known cone mass (Mmd). Many manufacturers will supply this basic data, even if they choose not to share their T-S parameters.

These two methods are less accurate than the traditional methods (delta mass and delta compliance), as they assume the test speaker is close in tolerance to the published parameters. While this might be true for a good working unit, it wouldn’t be useful if you were trying to troubleshoot a defective unit. These methods only require a

single measurement to be taken, and can provide meaningful data for rapid evaluation of drivers, at the cost of introducing more measurement error into the collected data.

To utilize the Specified SPL method, you need only measure the free-air impedance curve and enter the standard sensitivity in dB SPL at 1 W/1 m, as well as the piston diameter. It is standard practice to measure the piston diameter using half to one-third of the surround roll, if this dimension isn’t listed on the datasheet.

The other efficient method is the Specified Mmd method. Similarly, you need to measure the free air impedance of the DUT, and enter the piston diameter, as well as the Mmd (cone mass) from the datasheet.

Rub and Buzz and Linearity

Most traditional acoustic analyzers can be “trained” to detect rub and buzz failures in cone drivers with failure modes due to particle contaminant in the magnetic gap, or a rubbing voice coil. The basic detection scheme relies upon spectral analysis of the sixth through the 10th-order harmonics that are present in the defective units.

Driver Editor			
General Information Parameters Physical and Mounting Information			
Manufacturer: Peerless			
Model: 850102 4" SDS (Delta Mms)			
Nominal Diameter = 116 mm (4 inches)			
Resonance in Free Air	f(s) = 91.651 Hz	Reference Efficiency	n(D) = 0.26089 %
Resonance on Baffle	f(b) = 0 Hz	Voice Coil Inductance	L(e) = 0.67213 mH (1k Hz)
Total Q	Q(ts) = 0.52525	Flux Density	B = 0 Tesla
Electrical Q	Q(es) = 0.5442	Length of Wire in Gap	L = 0 meters
Mechanical Q	Q(ms) = 2.8446	BL Product	BL = 5.3903 N/Amp
Equivalent Volume	V(as) = 2.2895 liters	Effective Moving Mass	M(ms) = 5.6325 grams
	V(as) = 0.08085 cu ft	Voice Coil Diameter	D(vc) = 26 mm
Compliance	C(ms) = 0.535 mm/N	D(vc) = 1.0236 in	
Mechanical Resistance	R(ms) = 0 kg/s	Voice Coil Depth	D(jd) = 9 mm
DC Resistance	R(e) = 5.7707 Ohms	Magnetic Gap Depth	D(mg) = 6 mm
Maximum Impedance	Z(max) = 31.252 Ohms	Voice Coil Material:	
Minimum Impedance	Z(min) = 5.7707 Ohms	Voice Coil Former:	
Max Thermal Power	P(t) = 0 Watts	Voice Coil Layers:	2
Thermal Resistance	R(t) = 0 deg C/W	Voice Coil Wire Gauge:	
Max Linear Excursion	X(max) = 0 mm, peak	Voice Coil Vent:	
Max Excursion	X(peak) = 0 mm, peak	Wright Parameters:	
Piston Area	S(D) = 0.005518 sq m	K(j) = 0.29546	
Peak Volume Displ	V(D) = 0 liters	X(j) = 0.35326	
Sensitivity	SPL = 86.265 dB SPL (1W/1m)	K(i) = 0.096433	
	SPL = 87.684 dB SPL (2.83Vrms)	X(i) = 0.42839	

While DATS V3 does rely upon a human to guide and teach the algorithm the limits (using a known “Golden Sample” to verify the pass target), the DATS detection scheme is quite different than the harmonic distortion techniques utilized by other acoustic test systems.

Elegantly, the DATS rub and buzz algorithm is looking for shifts in resonance. If the resonance

Figure 3: The Driver Editor panel is shown for Vas Delta Mass testing. Derived T-S and Wright parameters are listed and user notes can be added to the records.

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dramatically shifts, the loudspeaker driver is either no longer operating in the small-signal, linear range or a failure mode has been detected such as a rocking mode causing a rubbing voice coil or a tinsel wire tapping against the underside of the cone. Highly intermittent failures (from such things as contaminants in the magnetic gap) will be more difficult to detect using the DATS methodology.

As this hardware is running off a USB 5V DC bus, it typically is capable of staying in the small signal domain for most drivers. If you are dealing with microdrivers, then you may need to exercise some judicious level setting (30 dB to 40 dB below the typical DATS sweep amplitude of +4 dBu) to keep it out of the large signal domain.

In the panel shown in **Figure 5a**, you can see the set-up panel for rub and buzz detection for signal level, resonance, and impedance maximum (Zmax) tolerance. Then you need to connect DATS V3 to a known good unit, free from any

unusual distortion and possessing a nominal impedance curve. This is the teaching method that requires some “pre-work” in both the selection of this “Golden Unit,” as well as appropriate level setting based on the power handling capabilities. Interactive prompts will guide the operator to change to a different unit in order to determine if any rub or buzz issues exist (see **Figure 5b** and **Figure 5c**).

While the rub and buzz test was first developed for the DATSv2 hardware, the linearity test is a new (but related) test developed for the V3 release.

By relying upon the principle of small signal performance and judicious setting of input drive levels, some preliminary analysis of power handling capability is available. As with the rub and buzz test, you need to properly set the drive levels. But unlike that particular test, no golden unit is required in order to take advantage of this test method. Full control over the drive range, the number of sweeps, as well as if you want to sweep from high-to-low or low-to-high amplitude levels provides maximum flexibility. This is a quite interesting use of the DATS V3 as an impedance analyzer, as linearity is typically measured using amplitude data (see **Figure 6**).

Software Integration

DATS V3 directly integrates with several other software packages for maximum flexibility. In addition to the option to export a native file *.zma or *.txt format, it is also possible to export impedance data as a text file with a CLIO compatible header. Once T-S driver parameters have been computed, they can be saved either as a native DATS file or as a WinSpeakerz driver file

Figure 4: The four different test methods for Vas are shown: Test Box, Added Mass, Specified SPL, and Specified M(md).

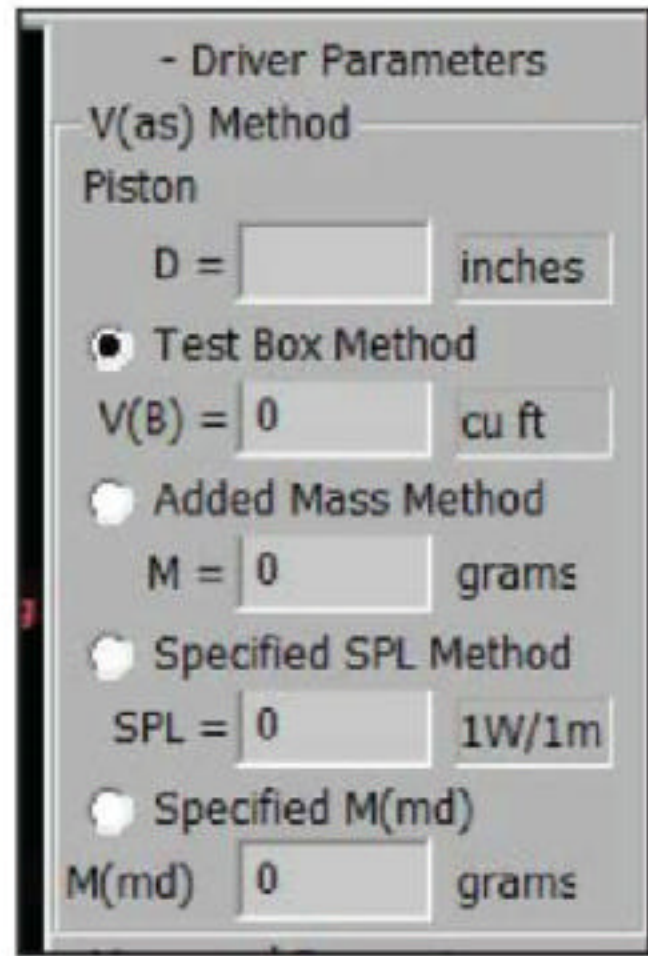


Table 1: The T-S parameters for the Peerless 4” SDS are collected from the manufacturer’s official datasheet, as well as the measured parameters using each of the four available test methods for DATS V3.

	Datasheet	Added Mass	Test Box	Specified SPL	Specified Mmd
Zmin	6.2 Ω	5.7707 Ω	5.7707 Ω	5.7707 Ω	5.7707 Ω
Zo	31.6 Ω	31.252 Ω	31.252 Ω	31.252 Ω	31.252 Ω
Re	5.6 Ω	5.7707 Ω	5.7707 Ω	5.7707 Ω	5.7707 Ω
Le	0.9 mH	0.67213 mH	0.67213 mH	0.67213 mH	0.67213 mH
Fs	74.2 Hz	91.651 Hz	91.651 Hz	91.651 Hz	91.651 Hz
Qms	2.41	2.8446	2.8446	2.8446	2.8446
Qes	0.52	0.6442	0.6442	0.6442	0.6442
Qts	0.43	0.52525	0.52525	0.52525	0.52525
Vas	3.4 ltr	2.2895 ltr	2.0052 ltr	1.83 ltr	2.1 ltr
Cms	0.74 mm/N	0.535 mm/N	0.469 mm/N	0.428 mm/N	0.486 mm/N
Bl	5.6	5.3903	5.7598	6.0268	5.6575
Mms	6.2 g	5.6325 g	6.4312 g	7.0411 g	6.2 g
Sensitivity (2.83 V/1 m)	86.7 dB SPL	87.684 dB SPL	87.108 dB SPL	86.7 dB SPL	87.264 dB SPL

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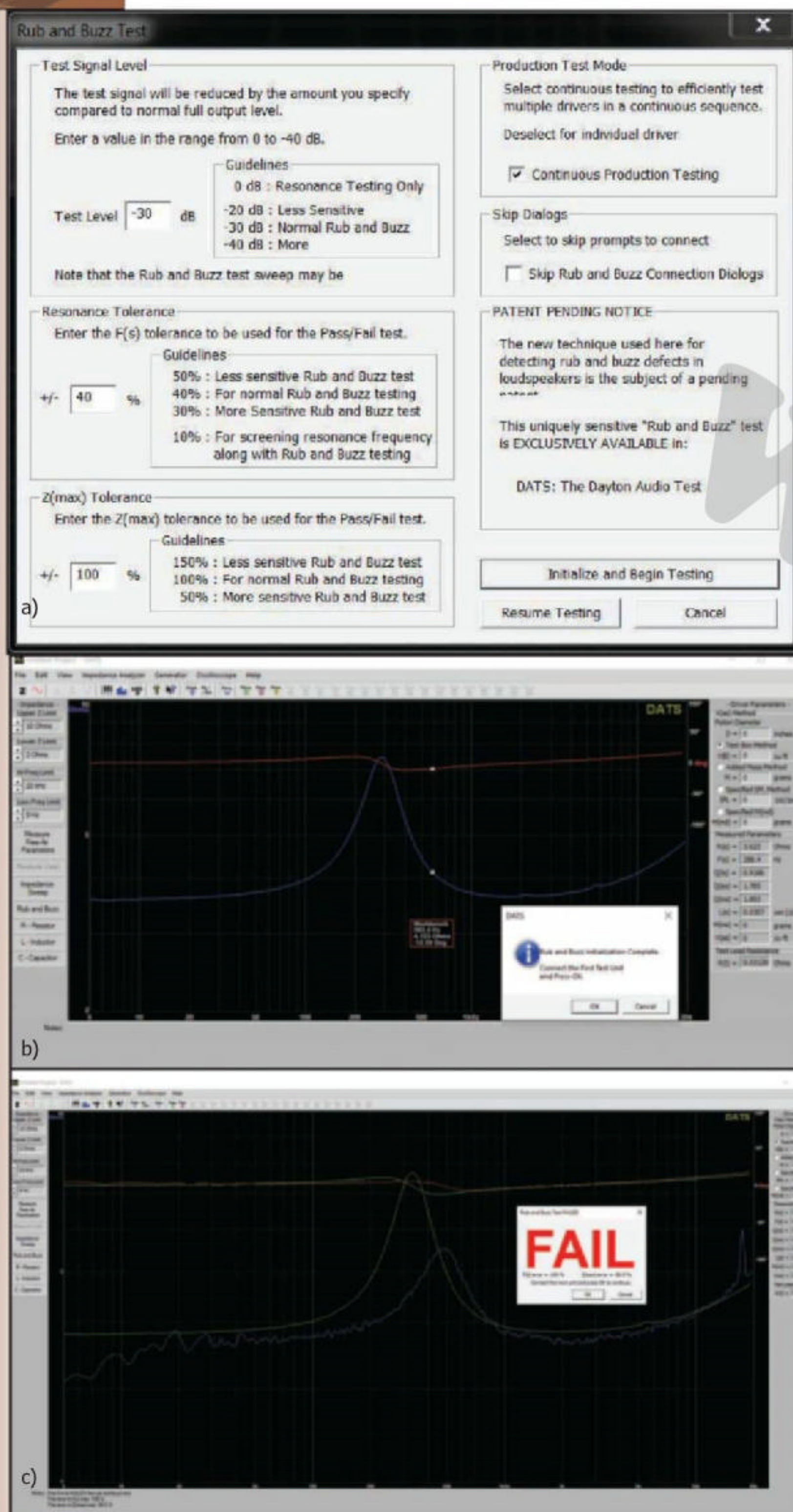


Figure 5: The set-up panel for Rub and Buzz testing provides signal level setting, tolerance setting, and the option to enable production testing (a). After entering appropriate signal levels and tolerances, the next step is to initialize the system with a known good "Golden Sample" unit to teach DATS V3 appropriate limits for rub and buzz (b). A failed unit is detected using the rub and buzz algorithm. In this case, the failure is due to a severe rocking mode (c).

to aid in woofer box simulation work. For those unfamiliar with True Audio's WinSpeakerz, it is a low-cost software package aimed at the DIY market for those users looking to design woofer boxes and passive crossover and Zobel networks.

When generating your loudspeaker driver files (using one of the four available methods), the Wright model parameters are also calculated, which provides a more representative description for the complex voice coil impedance.^[3] For more details, please see the tech notes in the DATS V3 documentation.

Final Thoughts

While the acoustic measurement system world is strongly biased toward PC-based data acquisition systems, consistent demand for audio testing portability has led to the proliferation of mobile apps that offer powerful measurement capabilities that can be run on a smart phone or tablet (Studio Six's AudioTools for iOS and Bofinit's AudioTool for Android are prime examples). Even with this flexible and low-cost measurement capability, there are times when you need an impedance measurement. In those cases, there isn't "an app for that," and some specialized hardware is required in order to collect this data. And while DATS V3 is firmly rooted as a PC application, its spirit and adaptable workflow definitely have kept up with the changing expectations for today's engineers and audio enthusiasts.

It has been interesting to observe the evolution of DATS over the past 12 years. The software has consistently added interesting features that expand significant capabilities, whether you are hobbyist, mobile installer, field technician, or pro engineer far from a rack of professional measurement gear.

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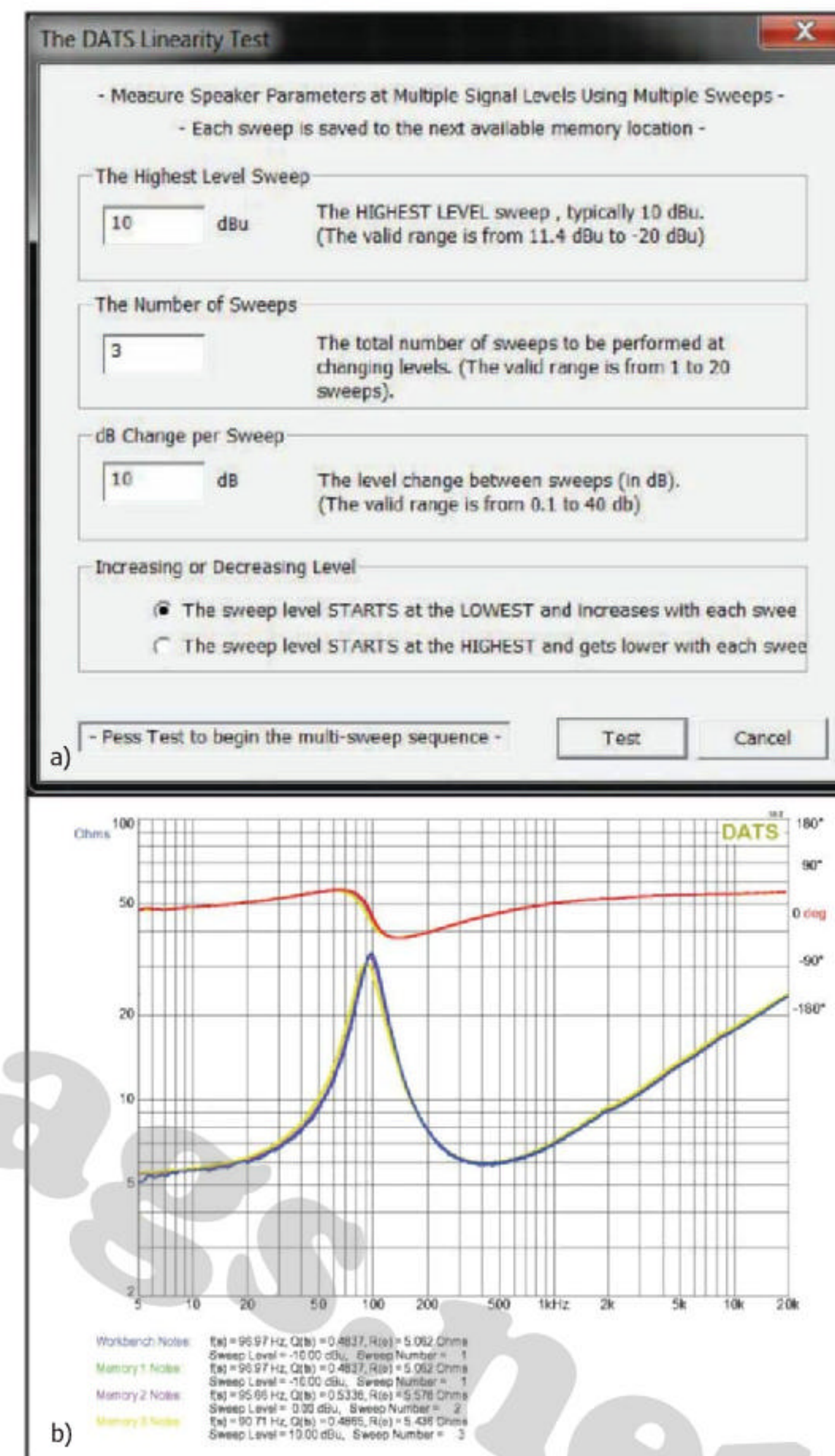


Figure 6: The set-up panel for Linearity testing provides settings for signal level, number of sweeps, and the decibel change per sweep (a). Linearity test results are plotted for three sweeps in 10 dB steps up to the maximum amplitude of +10 dBu for the Peerless 850102 (b).

Physically, it has gone from the very basic look of a low cost, specialized peripheral, to a robust and professional appearance. Feature-wise, it provides the end user with the ability to measure T-S parameters, something that can't be done with a sound card-only based test system, along with a host of other capabilities such as signal generation and linearity testing. This is a flexible tool that can be used for a variety of interesting functions unrelated to woofers, such as testing speaker cables or tweeters. Its greatest strength lies in the satisfying fact that it encourages users to use, analyze, and better understand the vast amount of information that is communicated from a humble impedance curve.



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Strategies for Optimizing Audio Production Tests

By
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(Audio Precision)

This article focuses on several key challenges found in the production and manufacturing tests of audio products, and shares strategies for optimizing the measurement system to address said challenges. Included in the discussion are test time and device throughput, test process control, and repeatability, cost-of-test, and the use of reference devices (so-called “golden units”). The advantages of a common test platform (across R&D and Manufacturing) are also reviewed.

Functional validation—along with the more robust testing found in quality control—of audio devices in a manufacturing environment is often considered a slow, expensive, and difficult process. In this article, we will discuss the realities of manufacturing tests for audio devices ranging from headsets, smart speakers and surround sound receivers to all classes of audio devices. Specifically, we will discuss the physical limitations of audio measurement and then demonstrate that audio measurements do not need to be slow or expensive. With proper automation, the process can be made fast, simple, and reliable.

The Physical Realities of Audio Testing

The simple equation $BT \geq 1$ (where the Bandwidth $\times$ Time product must always be equal to or greater than 1) defines the absolute minimum amount of time required to measure an audio, or any time-variant, signal. More simply, $1/f$ (where f is the frequency of interest) defines the absolute minimum amount of time required to sample a signal of a given frequency. Practically, this means that the lowest frequency energy we want to measure in our device will define the minimum amount of time an audio measurement will take. If we want to measure a device from 20 Hz, the measurement will take an absolute minimum of 50 ms.

For a sine wave, $1/f$ is the period, or the duration of one cycle. Practically, sampling a single cycle of a sine wave will likely have measurement repeatability problems. In addition, most audio analyzers rely on Fast Fourier Transform (FFT) analysis and apply a windowing function to the time signal prior to analysis. Therefore, the minimum is often ~ 3 to 5 (or more) cycles of the period of the signal of interest. Again, for the hypothetical 20 Hz signal, that would give us a minimum measurement time of 150 to 250 ms. As we will further discuss, 250 ms is not an unrealistic amount of time for an audio measurement.

While the frequency of the signal of interest defines the minimum possible cycle time, the type of signal and the desired measurement repeatability also interact to set the minimum measurement time required. For example, a device might be tested with a simple spot check of THD+N at 1 kHz. At 1 kHz the period of the signal is just 1 ms so one might imagine that 5 ms would be enough dwell time based on the preceding. However, THD+N is looking at several component signals—which include the 1 kHz fundamental tone, the harmonic distortion products of the fundamental, and, crucially for test time purposes, noise (see **Figure 1**).

The harmonic distortion products are integer multiples of the fundamental and, therefore,

higher in frequency than the fundamental. Because of this, if we sample the signal for enough time to accurately measure the fundamental, we are likely to be sampling the signal for long enough to measure the harmonic products as well.

Whenever a noise signal is being measured, we must define the bandwidth or passband over which the noise is measured. For all practical purposes, there is noise at every frequency in the environment. In audio, it is conventional to measure noise over the 20 Hz to 20 kHz passband of human hearing. The key being the low-frequency corner of human hearing. The fundamental may be 1 kHz, but we are including energy at 20 Hz in our measurement and that lower frequency passband controls the minimum amount of time required to sample the signal. Moreover, if the signal was a steady-state 20 Hz tone, we could rigidly define the minimum sample time.

However, noise is usually a random signal. In this case, the question becomes: What is our desired measurement repeatability? At an extreme, if we are measuring a pure white noise signal with a Gaussian distribution over a 20 Hz to 20 kHz bandwidth, we would need to sample the signal for more than 2 minutes in order to reach six sigma measurement repeatability.

Thankfully, most devices do not emit a pure white noise spectrum and most test scenarios do not need 99.99966% measurement repeatability. Most electronic devices have their noise dominated by steady-state tones, usually 50 Hz or 60 Hz power supply hum and the truly random component of the noise portion of the THD+N is not large.

With acoustic devices such as speakers, it is often not desirable to measure noise. A passive speaker does not have a noise product. Any noise emitted at the output of a speaker is energy that was input to it or is ambient acoustic noise. For passive speakers, we are typically only interested in their ability to reproduce the fundamental test signal and the distortion to it that they introduce. Finally, for manufacturing line quality control measurements, repeatability of 99% (or even 95% or better), is typically more than good enough.

While the physics of measuring an audio signal define the absolute minimum amount of time required to make that measurement, compared to human time scales, audio measurements can be quite fast. In the previous discussion, we have been bandying about figures in the hundreds of milliseconds and this is not misleading. It is indeed possible to make a complete audio measurement

in a couple of hundred milliseconds, with some limitations, or in a few seconds with no limitations.

Fast and Slow Audio Measurement Techniques

The most common and traditional audio measurement technique is the stepped sine sweep. Using this technique, a signal generator produces a steady-state tone at one frequency while an analyzer measures the fundamental level, harmonics, noise, and other products at

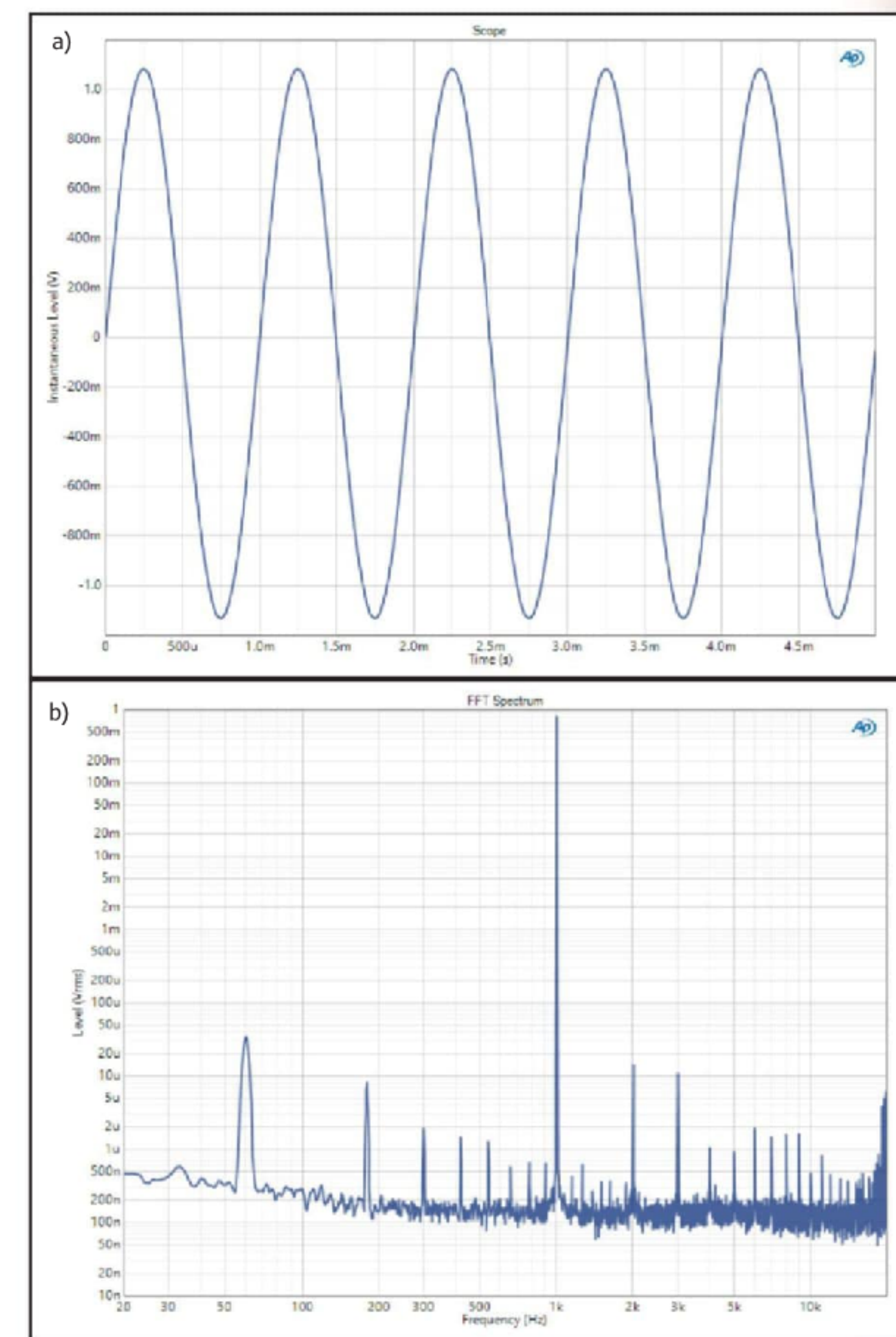


Figure 1: A device might be tested with a simple spot check of THD+N at 1 kHz (a). However, THD+N is looking at several component signals—which include the 1 kHz fundamental tone, the harmonic distortion products of the fundamental, and noise (b).

the output of the device under test (DUT). Once the measurements are completed at one frequency, the generator is tuned to the next frequency and the process is iterated across the frequency range being measured with the system dwelling at a number of measurement points according to the desired measurement resolution. Because of its long history, most if not all audio devices are characterized this way, at least when they are being developed and often even on a production line. This technique is not particularly fast—often

taking on the order of 1 second per measurement point with common audio analyzers, depending on the exact measurement you want to make at each point. The obvious solution in production test scenarios is to reduce the number of measured points. However, with modern FFT-based audio analyzers, there are much faster measurement techniques available, including multitone and chirp/continuous sweep.

Multitone—Instead of sweeping a single tone across the frequency range, pausing at each frequency point, we can simultaneously generate all the tones and simultaneously measure all the parameters we want (see **Figure 2**). Instead of waiting for 250 ms to 1 s to make a measurement just at 20 Hz, we can make measurements across the entire frequency range of interest in the same amount of time.

The main limitation of the Multitone technique is that because multiple test tones are generated simultaneously, THD+N cannot be measured directly. This is because instead of just harmonic distortion and noise, a multitone signal creates intermodulation distortion products as well. The multitone equivalent of THD+N for a pure tone is Total Distortion and Noise (TD+N). For all real-world devices TD+N will be a worse number than just THD+N. This means that multitone TD+N results are not directly comparable to THD+N results. However, in return, multitone measurements are fast with the test time purely dependent on the desired lower frequency and the measurement repeatability.

Chirp/Continuous Sweep—Thanks to innovations contributed by Angelo Farina, the continuous, logarithmically swept sine wave has become an extremely useful and fast measurement technique. With this technique the frequency of the test signal varies continuously with time. The reference signal and the response signal are deconvolved and the impulse response, frequency response, distortion, and other parameters can then be extracted. As a measurement technique, chirp can be almost as fast as multitone but has the advantage that the stimulus signal is a sine wave and, therefore, the results are directly comparable with those made with a stepped sine sweep (see **Figure 3**). The most significant limitation of the chirp technique is that it cannot measure noise. Chirp measurements return THD but not THD+N.

In short, a stepped sine sweep may take 30 seconds or more, and because of its ubiquity as a traditional measurement technique, it has become an anchor that has led to the conclusion

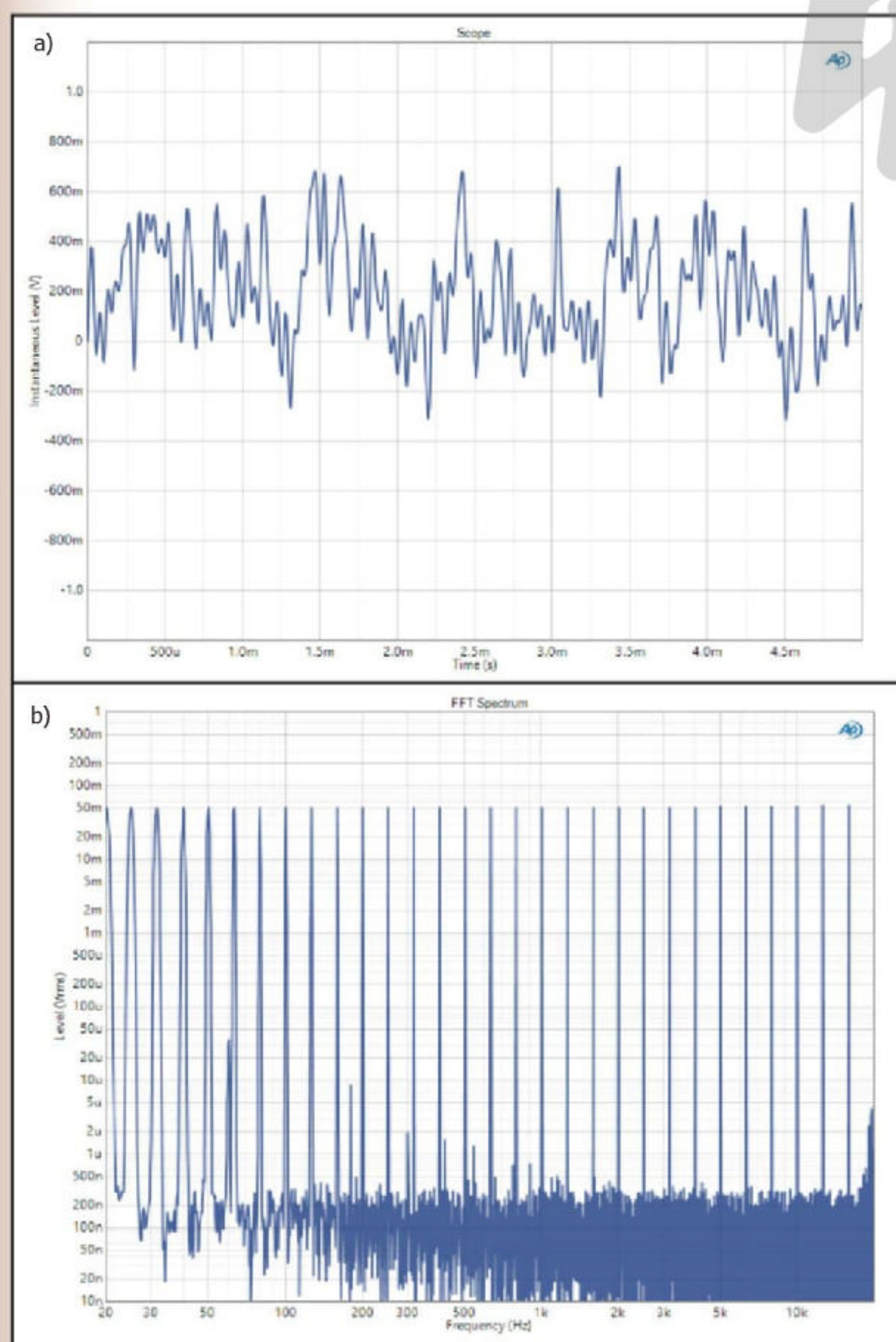


Figure 2: Instead of sweeping a single tone across the frequency range, pausing at each frequency point (a), we can simultaneously generate all the tones and simultaneously measure all the parameters we want (b).



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that audio measurements are slow. In fact, audio measurements can be quite fast; the multitone and chirp technique demonstrate that measurements can approach the theoretical limits of test time.

What Slows Down Audio Measurements?

Example 1: Audio measurements can be fast, even in real-world environments. There are test solutions in operation on a mass-scale that test devices in 250 ms. However, while the audio measurement takes just 250 ms, the total

cycle time in this case is 12 seconds. Where does the disparity come from? First, in this example, the device has four signal paths—two speakers, one microphone, and one headphone jack—so only a total of 1 second is spent making actual audio measurements. The other 11 seconds are consumed placing and removing the device from a test jig, and digitally controlling the state of the DUT for each signal path.

This, in fact, is the dominating factor in real-world audio measurement. Audio measurements themselves can be optimized to be extremely fast. Meanwhile building fully automated test jigs is difficult. From a practical standpoint, optimizing the test jig will likely yield a much bigger improvement on cycle time and measurement quality than anything else. In this case, when we were first approached about optimizing this test, the cycle time was 2 minutes. The audio test time was reduced from 12 seconds to 1 second, a very large improvement, but the device handling time was improved from 108 seconds to 11 seconds by improving the test jig and optimizing the human operator time-motion.

Example 2: While it is useful to analyze and optimize the audio measurements being made on a manufacturing line, it is quite likely that other steps in the test process are quite time consuming. In another manufacturing test scenario, evaluation of the microphone on a smart device was performed by recording wav files and then analyzing them on a computer hosting the test system. Unfortunately, the universal asynchronous receiver/transmitter (UART) implementation on the DUT was sub-optimal and transferring the wav files over the slow UART was the source of most of the test time.

Absolute vs. Relative Pass/Fail Limits

Another challenge in production testing of acoustic devices is developing and implementing pass/fail limits. Finding the best practical limits for pass/fail determination can dramatically improve test system throughput by preventing false failure results.

During product development, it is likely that figures of merit such as frequency response are derived from ideal, often anechoic test environments. Typically, this results in a set of absolute performance targets that become the production test limits. However, in a manufacturing environment, the device will likely be evaluated inside a small test box. Due to the geometric constraints of a typical test box, the frequency

responses recorded in a production environment will not reflect the free field or in-chamber responses determined during product development. Furthermore, if there are many test cells operating in parallel to support large scale production, it is likely that geometric differences from cell to cell, while small, will result in slightly different test results depending on which station the device is tested.

Alternatively, one or more “standard” or “golden” units can be selected to establish relative pass/fail limits. Commonly, these are either an ideal or perfect example of the device, or they are a collection of devices which have been selected for accurately representing the statistical spread of many devices. Once the golden units are selected, they can be run through each test station, with the pass/fail limits established as offsets from the actual measurements of the ideal device. These relative pass/fail limits compensate the limits with the unique variation from cell to cell. While this process entails some time up front, it pays high dividends over the production life of the device.

Measurement Comparability

Another challenge of manufacturing test is the consistency of measurements made on a production line with those made in the development lab. Ideally, the same measurements are made using the same test systems in both scenarios. Particularly when there is a manufacturing problem, it is frustrating to try and troubleshoot a potential manufacturing defect while at the same time trying to solve unrelated discrepancies between measurement systems (possibly produced by different test equipment vendors) used in the lab compared to the manufacturing lines.

Managing Costs

Optimizing audio test time intrinsically lowers the cost of audio testing. At the very least, fewer test stations might be required to test a given number of devices. One area where costs can be further optimized is in observing that typically the actual audio measurement is much faster than the time it takes to install and remove devices from their test jigs.

Because of the time savings, it is often worthwhile to consider multiplexing one audio analyzer between two or more test jigs. In this scenario, an operator can be replacing the device in one test box while the actual testing is happening in another. By optimizing both the audio measurement time and the analyzer utilization it is possible to drive the cost of audio testing per tested device to much smaller values.

Conclusions

Audio testing is often thought of as being slow, but it does not need to be. Complete characterization of an audio device can be done within a few seconds. While there are physical limitations that bound how quickly a measurement can be made, test speeds are still quite fast.

In many cases, unoptimized audio measurements can be optimized to be at least an order of magnitude faster. Further, while it is common to focus on the speed of the audio measurements, total cycle times are often dominated by the handling time required to place and remove devices from test jigs.

By optimizing the audio measurement technique used, the test jig, and the operator’s time-motion, and possibly sharing one analyzer between two or more test cells, it is likely that the cycle times and the test costs can be reduced by at least an order of magnitude.

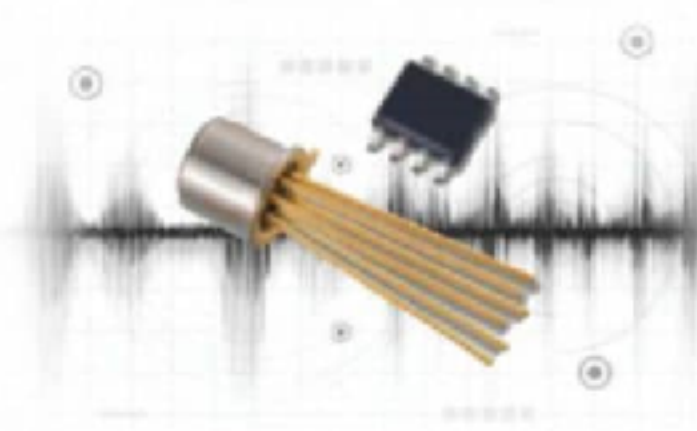
About the Author

As Vice President of Product Development at Audio Precision (AP), Dan Knighten is focused on the definition of the features, functions, and price of new products while managing AP’s engineering team and shepherding products through their complete life cycle. He also supervises the AP technical support team, a rich source of firsthand experience on how customers use, and aspire to use, Audio Precision analyzers, software, and test accessories.

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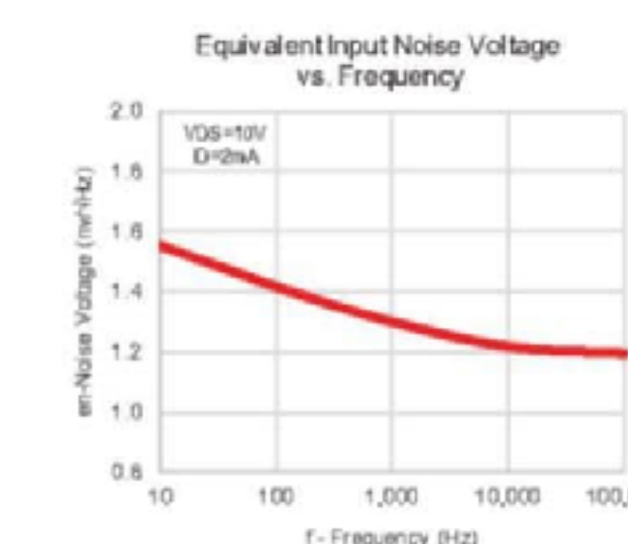
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Figure 3: The continuous, logarithmically swept sine wave (a) has become an extremely useful and fast measurement technique. With this technique the frequency of the test signal varies continuously with time. As a measurement technique, chirp can be almost as fast as multitone but has the advantage that the stimulus signal is a sine wave (b).

Miking Choirs with Orchestras or Organ

By
Richard Honeycutt
(United States)

The December 2019 Sound Control article addressed the issue of miking choirs for recording, broadcast, and live sound reinforcement. The challenges of obtaining a good pickup of instrumental accompaniment, including piano, organ, and orchestra was not discussed, and is the topic of this article.



Photo 1: This empty platform and choir area of a 665-seat church sanctuary is used for major choir/orchestra productions.

Photo 1 shows the platform and choir loft of a 665-seat church sanctuary. Starting at 15.5' above the choir, there is an acoustically transparent screen in front of the organ pipe chamber. In most services, the choir is accompanied either by an organ or by the a piano, which is located in a boxed-in area of the platform to the left of the image shown in Photo 1.

When the sound system was installed, two suspended hypercardioid mics were located 7' from the front of the choir, spaced symmetrically about the side-to-side center position, 14' apart. The choir balance was good, as was the balance between choir and organ.

Some years later, at the request of a new Minister of Music, the mics were repositioned to a distance 19.5' in front of the choir, located with the same side-to-side dimensions. After this change, the balance among singers was somewhat improved, but the organ began to overbalance the choir. In both the first and the second mic position, balance between the piano and the choir was good.

As mentioned in the December 2019 Sound Control article, a weakness of the spaced-pair arrangement of mics is that they do not sum well to mono. Since this church broadcasts its weekly service on FM radio, and the studio-to-transmitter link is a mono FM transmitter, good mono sound is important.

Recently, in order to improve the choir's sound on the broadcast, the spaced pair of hypercardioid mics was replaced by a coincident XY pair of cardioid mics, located above the head of the choir

director, about 3' in front of the center of the choir. This location improved the choir/organ balance by placing the mics relatively closer to the choir and farther from the organ. However, it also emphasized the voices of certain sopranos who stand near front row center. As a result, the coincident pair was moved another 3.5' from the front of the choir. The resulting improvement in balance among the singers outweighed the slight increase in organ pickup.

Microphone Setups

Recently, this church has begun using brass or string ensembles to accompany the choir in some services. **Photo 2** shows the choir with a small orchestra used in a Christmas program. When only the strings or only the brass are used, the instrumental ensemble is usually located left of the conductor, with the piano placed as shown in the Photo 2. The XY coincident pair works well for picking up choir and instruments—including organ and piano—in this arrangement.

Sometimes a larger orchestra is used, and in these cases, a platform extension is required, as illustrated in **Photo 3** and **Figure 1**. For this service, a stereo CD recording was made. The spaced pair in the front (bottom of Figure 1) assisted in balancing the orchestral instruments with one another. The XY pair functioned mainly as a choir mic, and to keep the aural image centered in the stereo recording. Used in combination with the XY pair, the spaced pair of mics mixed well to mono, because the left and the right signals from the XY mics prevented comb filtering

from the spaced pair from becoming objectionable.

The mics used in this setup included small cylindrical condenser mics on a coincident mic bar for the XY pair and large-diaphragm hypercardioid condenser mics mounted at a height of 8' above the sanctuary floor (5.5' above the platform), with the stands placed at the ends of the third central pew from the front. Although the front four pews were made unavailable for seating, shock mounts were used for the spaced pair (see Photo 4). Figure 2 and Figure 3 show the frequency response and directionality of these mics.

The XY mics have nearly flat response (+3, -2 dB) across the audio range. Their polar response is an excellent cardioid pattern at 1 and 5 kHz, with a small rear lobe at 200 Hz and 8 kHz. The front-to-back ratio at 1 and 5 kHz is about 30 dB, but at lower and higher frequencies, it drops to about 15 dB. A more perfect cardioid pattern, available in some more expensive, large diaphragm mics, would improve stereo separation and favor choir pickup over orchestra pickup in this application.

The spaced pair mics also have a very flat frequency response. Unlike some large-diaphragm condenser mics, the presence peak of this mic is very small: about 4 dB at 12 kHz. The manufacturer of these specific mics only shows the polar response at one frequency, presumably 1 kHz. The maximum rejection is about 25 dB at the nulls, 112.5° off axis. The normal rear lobe of a hypercardioid mic reduces the front-to-back ratio to about 10 dB at 180° off axis. In this application, sounds from the rear are predominantly room reverberation, so lower rear rejection is not very important.

During the service using the larger orchestra, there was congregational singing to be picked up for broadcast and recording. A bidirectional ("figure-8") mic was suspended below the balcony face, centered right-to-left, about two-thirds of the way back in the sanctuary, and hypercardioid mics were also suspended below the balcony face, centered above each of the side rows of pews. The height of the congregation mics was 10' above the floor. This arrangement allows for a full congregation sound without emphasizing individual voices, and without allowing the organ or orchestra to overbalance the voices. The fact that sound picked up on each side of the bipolar mic is out-of-phase with sound picked up on the other side creates the perception of added width in the stereo mix.

Since these are not professional singers, occasionally soloists need to be miked. For this purpose, a handheld wireless mic is used. For two reasons, it is not stand-mounted. First, the platform



Photo 2: The choir is sometimes accompanied by an orchestra.



Photo 3: A platform extension is used for a larger orchestra.

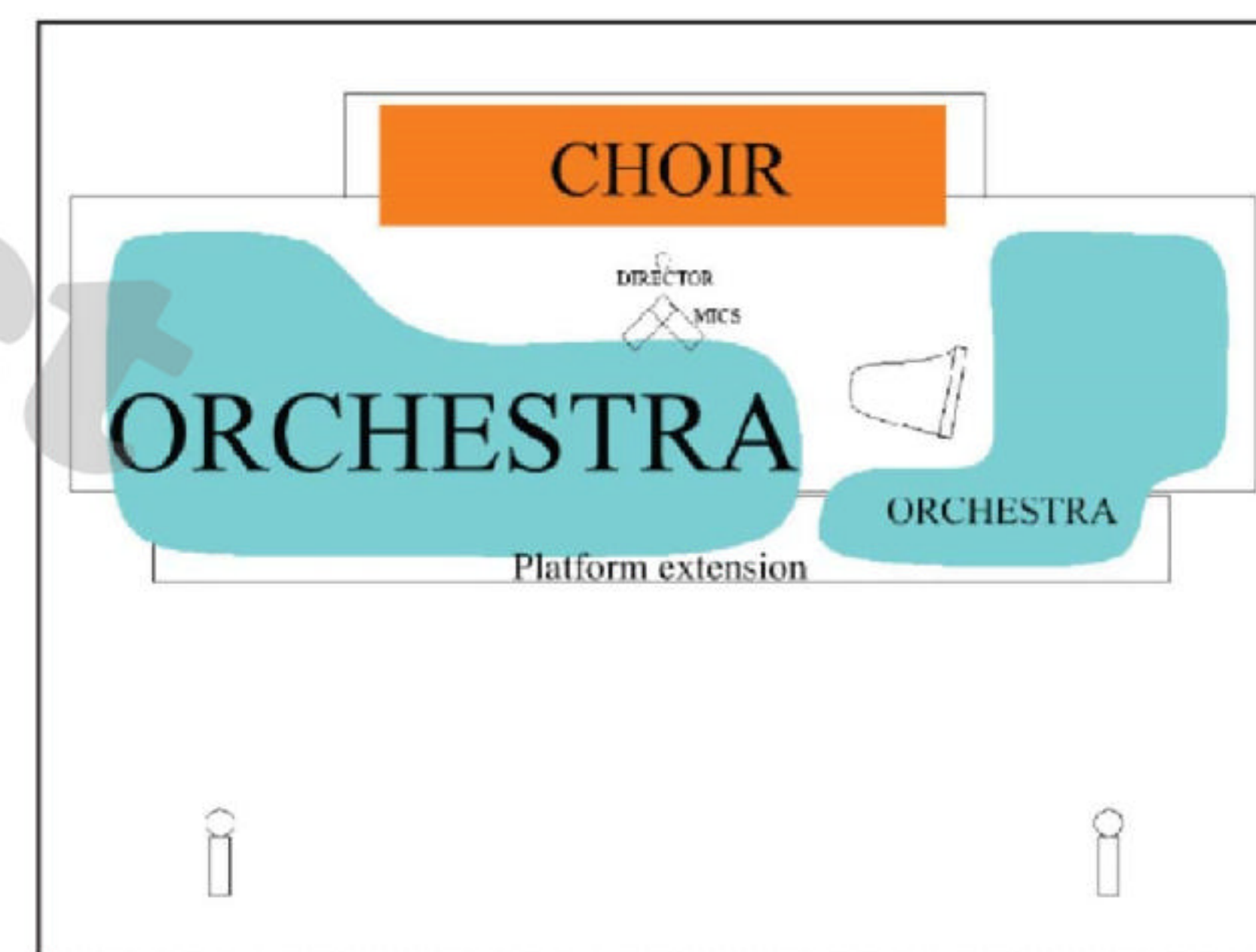


Figure 1: This plan view shows the positions of the choir, the larger orchestra, the piano, the director, and the mics.

is very crowded, even with the extension in place, and a mic stand might cause choir members to trip and fall. Second, if more than one soloist needs miking, the same handheld mic can be passed from one soloist to the other.

During this particular service, the choir was amplified in the house system, but the instruments were not. This provided freedom from feedback squeal, which was a great blessing for the operators doing the house and recording/broadcast mixing.

Pitfalls to Avoid

Aside from using sub-optimal mic positions, what pitfalls should a technician avoid when miking this type of program for live sound, recording, or broadcast?

Photo 4: A "spider-type" shock mount is used for each mic of the spaced pair.

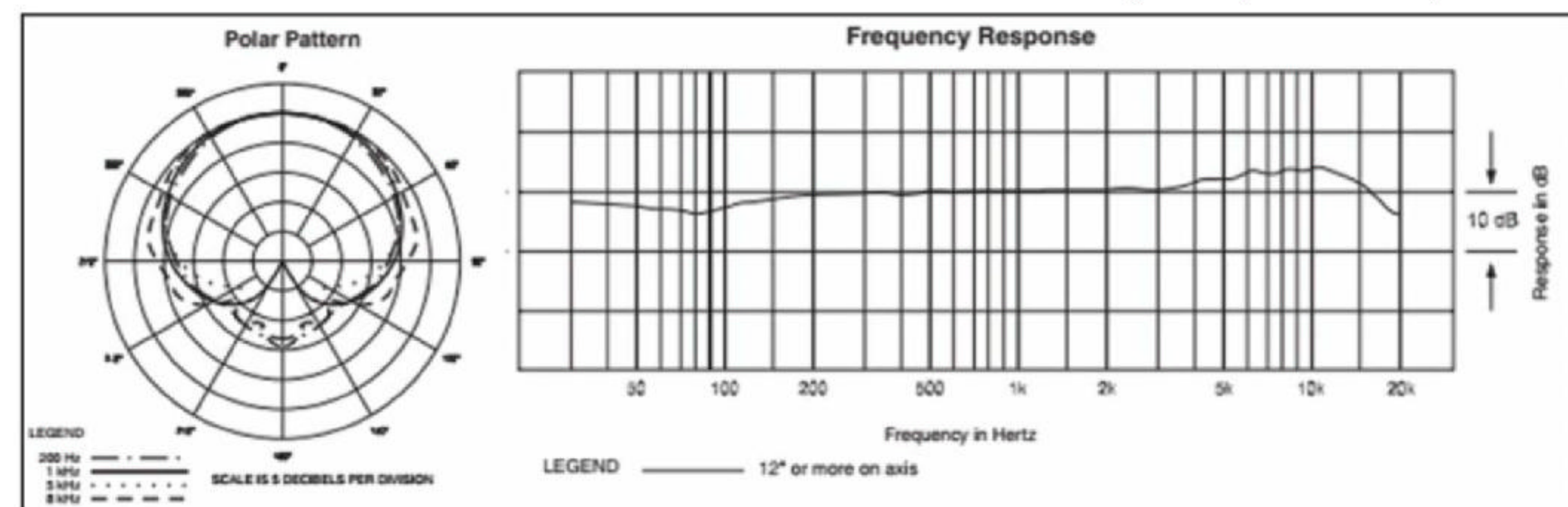
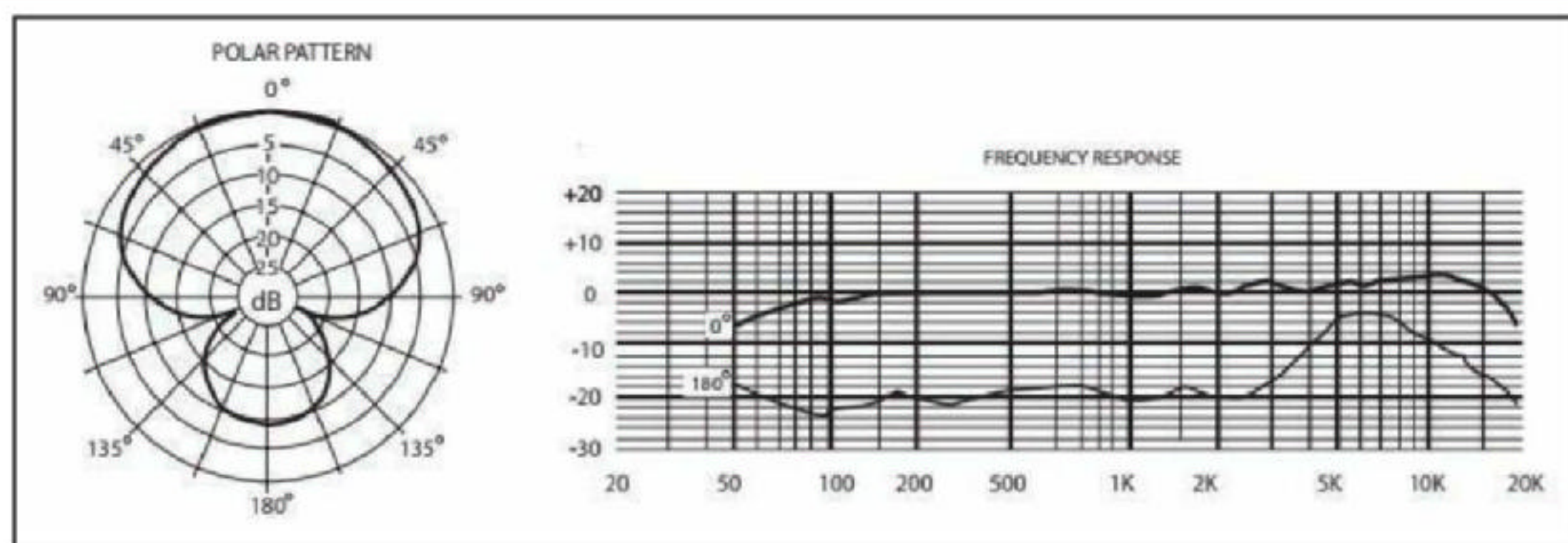


Figure 2: The frequency and polar responses of the mics used in the XY pair illustrate the excellent performance of these mics.

Figure 3: The frequency and polar responses of the mics used in the spaced pair work well in this application.



The obvious ones include using the wrong mics, overusing EQ, and adding artificial reverberation. Many venues—and not a few amateur recording engineers—have a suitcase full of their favorite vocal mics (see **Photo 5**) that have deliberate frequency-response tailoring (see **Figure 4**). This mic's 8-dB peak at 6.5 to 7 kHz was beneficial for TV and radio broadcasts back in the 1950s and 1960s, when radio and TV receivers typically did not reproduce frequencies above 5 kHz well. But it would cause a major feedback problem if this mic were used for live sound reinforcement, and would cause choir and instrumental music recordings to sound harsh.

The most common reasons that an engineer might overuse EQ include:

- Making up for poor mic performance.
- Compensating for poor mic location.
- Trying to compensate for the operator's hearing deficits (such as high-frequency dips caused by noise exposure).
- Attempting to make the sound clearer when the room is too reverberant.

The last reason betrays an unrealistic expectation on the engineer's part. Certainly too much

reverberation can muddy the sound of a recording or make it harsh, especially if the venue's low, mid-, and high-frequency reverberation are not balanced. But EQ is never the answer to such a problem. Instead, the engineer must recognize that acoustical problems of the venue are not within his/her ability to compensate, and accept that the best recording/broadcast (s)he can produce will have to be good enough.

Although a recording made in a dead room will not sound good, it is best to record "dry" and add synthetic reverb, if you decide to do so, after the fact. Sometimes recordings made in dry rooms do benefit from added synthetic reverb, but knowing how the reverb should sound, and how much to add, is best done later in a studio instead of during the recording. As for broadcasts from dead rooms, remember that almost all radio and TV stations use extensive compression of the audio signal, and compression emphasizes reverberation.

Hopefully, the discussion in this Sound Control article will help you to produce better recordings and/or broadcasts of choirs with instrumental accompaniment. **ax**



Photo 5: Some engineers are addicted to a particular vocal mic and use it for all applications.

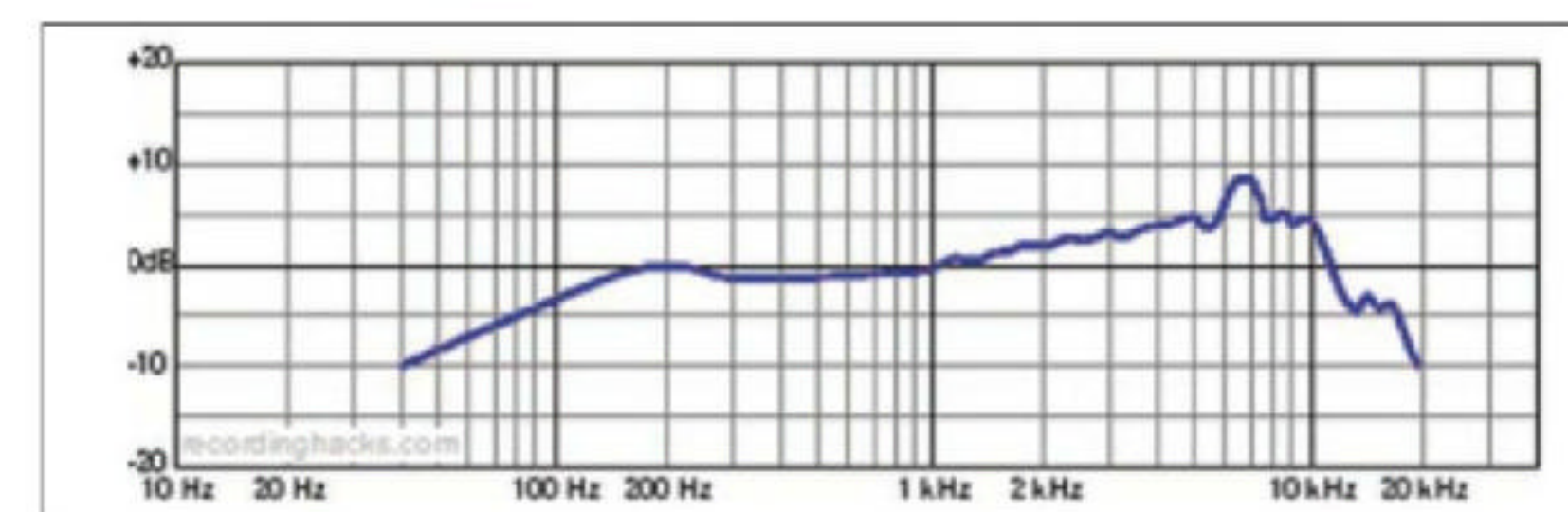


Figure 4: The frequency response of a vocal mic may well be unacceptable for miking choir and orchestra.

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FDA Errors to Consider Driving Audio ADCs and Active Filter Design Options

By
Michael Steffes
(United States)

Last month’s Audio Scope article (February 2020) introduced the intrinsic advantages using the newest low-power fully differential amplifiers (FDAs) to interface measurement mics to 24-bit delta-sigma ADCs. Some of the error terms to consider will be discussed here along with an alternate filter design.

Once the AC input coupled mic signal is available as a differential DC coupled output to the ADC, what added operational constraints need to be considered? First, what are the effects of output DC errors in the signal chain? Fortunately for audio

applications, these recent fully differential amplifiers (FDAs) have been developed with precision industrial automation and test and measurement applications in mind. Often driving precision 18-bit to 22-bit SAR ADCs, extremely low output common mode

Supplier	Device	Vs (minimum)	Vs (maximum)	SSBW@ Gmin	Slew Rate	Icc Maximum/ Channel	Vn_ flatband	Vos Maximum	Part Description
ADI	ADA4940-1	3 V	7 V	260 MHz	95 V/μsec	1.4 mA	3.9 nV/√Hz	0.350 mV	Ultra-low power, low distortion, fully differential ADC driver
TI	THS4551	2.7 V	5.4 V	150 MHz	220 V/μsec	1.4 mA	3.3 nV/√Hz	0.175 mV	Low noise, precision, 150 MHz fully differential amplifier
ADI	ADA4945-1	3 V	10 V	145 MHz	600 V/μsec	4.2 mA	1.8 nV/√Hz	0.05 mv	High speed, ±0.1 μV/C offset drift, fully differential ADC driver
MChip	MCP6D11	2.5 V	5.5 V	82 MHz	27 V/μsec	1.8 mA	5 nV/√Hz	0.15 mV	Low noise, precision, 90 MHz differential I/O amplifier
TI	THS4561	2.9 V	12.6 V	70 MHz	130 V/μsec	0.9 mA	5 nV/√Hz	0.20 mV	Low power, high supply range, 70 MHz fully differential amplifier
TI	THS4531A	2.5 V	5.5 V	36 MHz	250 V/μsec	0.4 mA	10 nV/√Hz	0.40 mV	Ultra-low power, RRO, NRI, fully differential amplifier with improved offset
LTC	LTC6363	2.8 V	11 V	35 MHz	75 V/μsec	1.9 mA	2.9 nV/√Hz	0.10 mV	Precision, low-power differential amplifier
LTC	LTC6362	2.8 V	5.5 V	34 MHz	45 V/μsec	1.1 mA	3.9 nV/√Hz	0.20 mV	Precision, low-power rail-to-rail I/O differential op-amp/ SAR ADC driver

Table 1: Recent precision FDAs with less than -130 dBc audio frequency distortion.

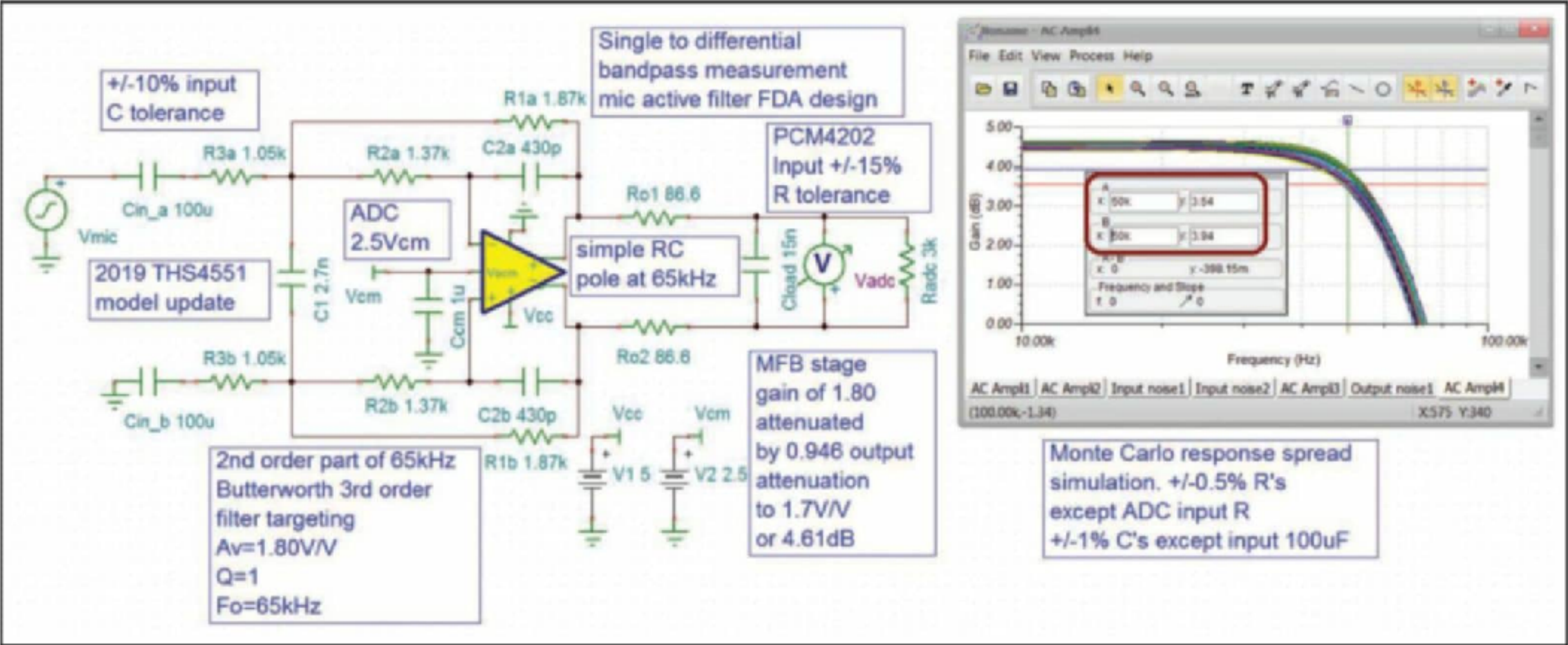


Figure 1: A third-order AC coupled Butterworth at 65 kHz shows the response spread using a Monte Carlo simulation.

and differential DC offsets are being delivered by the latest low power FDAs, which are shown in **Table 1**. These are all unity gain stable voltage feedback designs with negative rail inputs and rail-rail outputs—sorted here in descending unity gain bandwidth. Note the very low differential input offset voltages (V_{OS}). With an AC coupled input signal path, that differential input offset voltage has a gain of 1 V/V to the output and will not impair full-scale swing range.

The DC accuracy concerns at the ADC inputs also include common mode (CM) shifts from midscale. The audio delta-sigma ADCs will have an allowed input CM tolerance—rarely specified, the precision FDAs in Table 1 all offer less than 20 mV CM offset even with their control input floated. That by itself should not impair ADC performance, but include supply voltage variation as well if the V_{OCM} control input is floated as that output CM is just a divide by two from the FDA supply.^[1] Use a buffered ADC reference voltage source into the FDA V_{OCM} control pin if this becomes a concern. Simply buffering the ADC reference output to the FDA V_{OCM} inputs with a low-noise unity gain op-amp can reduce the output V_{CM} variation and common mode noise vs. a floated V_{OCM} control input condition. If the FDA supply voltage is well controlled, simply adding a capacitor on the FDA V_{OCM} pin to ground may be enough to remove the V_{OCM} noise from consideration.

RC Tolerances

The effect of RC tolerances will show up in a few ways. Their “tolerancing” will, of course, introduce a variation in the filter response shape as shown in **Figure 1**, using a third-order 65 kHz Butterworth design. Using $\pm 0.5\%$ Rs and $\pm 1\%$ Cs in the filter design shows only a 0.6 dB roll-off spread at

50 kHz. Most of this spread is assuming a $\pm 15\%$ ADC input resistor tolerance (unspecified). Removing that effect tightens the 50 kHz roll-off spread to 0.15 dB considering just the filter RC element variation.

An often more critical concern is the effect of resistor tolerancing on the feedback ratio mismatch in the FDA design. Feedback ratio mismatch creates several error terms—the most concerning here is the conversion of CM output noise to differential output noise. That conversion gain is shown as Equation 1 where G_1 is the R_F/R_G ratio on one side while G_2 is the R_F/R_G ratio on the other (R_1/R_3 here):

$$V_{od} = \frac{V_{ocm} \times (G_1 - G_2)}{1 + \frac{G_1 + G_2}{2}}$$

Assuming 1% Rs, with the nominal gain set to $R_1/R_3 = 1.78$ V/V as shown in Figure 1, gives a worst-case conversion gain of 0.0144. Floating the V_{OCM} input pin to get midscale for the THS4551 will show a CM spot noise of 300 nV/√Hz at 100 Hz (Figure 45 as shown in Reference 1). This converts to a worst-case differential output spot noise contribution of 4.3 nV/√Hz. Not significant at 100 Hz, but much more so at lower frequencies. This effect can be reduced using tighter tolerance resistors and/or driving the FDA V_{OCM} input pin with a low noise buffer from the ADC reference voltage. Another contribution to gain mismatch is the measurement mic output impedance. If known (and not zero), include that value in series with the blocking capacitor to ground on the non-signal input side of the FDA circuit to more closely match the feedback divider ratios.

Going the other way, feedback ratio mismatch also converts the single-ended input signal to a small output CM term identical to the desired output differential signal but attenuated by Equation 1.

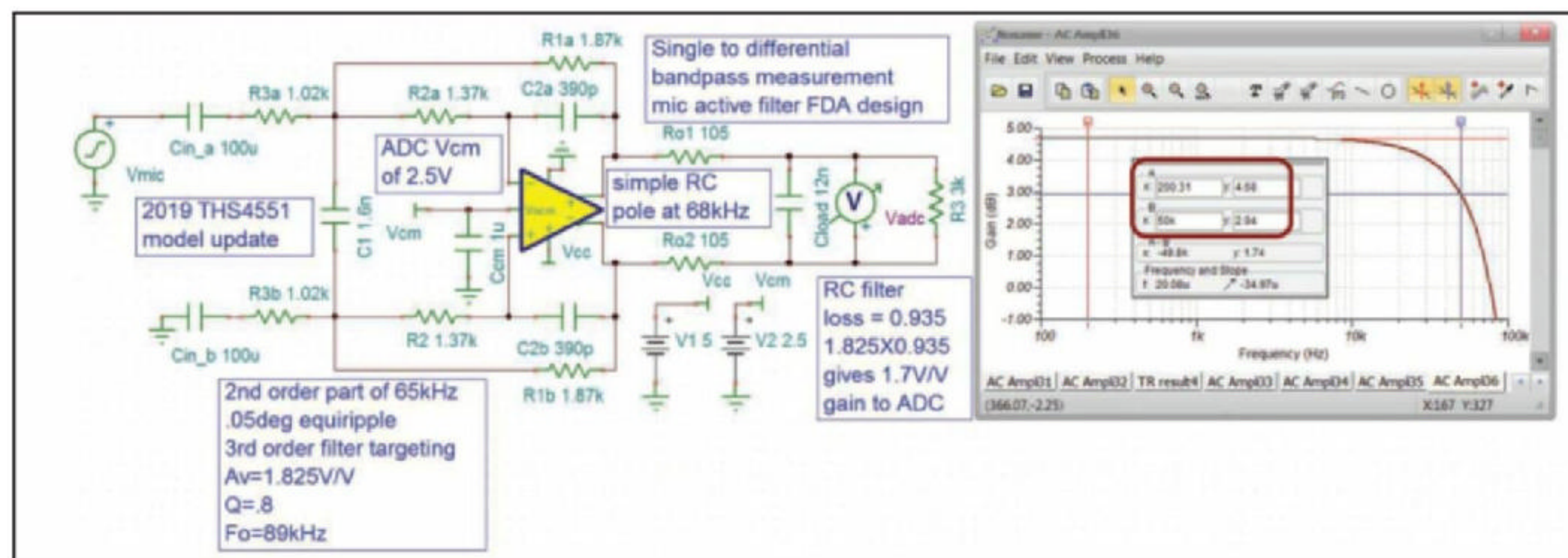


Figure 2: A third-order 0.05° Equiripple active filter design is shown driving an audio ADC.

This signal-related CM term is rejected by the ADC's common mode rejection ratio (CMRR) to be inconsequential. For instance, no effort to match feedback ratios were used in generating the less than -130 dBc harmonic distortion plots for the THS4551 (Figure 3 as shown in Reference 1). That characterization interface (Figure 61 as shown in Reference 1) also rejects CM in converting back to single ended for harmonic distortion measurements using a very linear passive network.

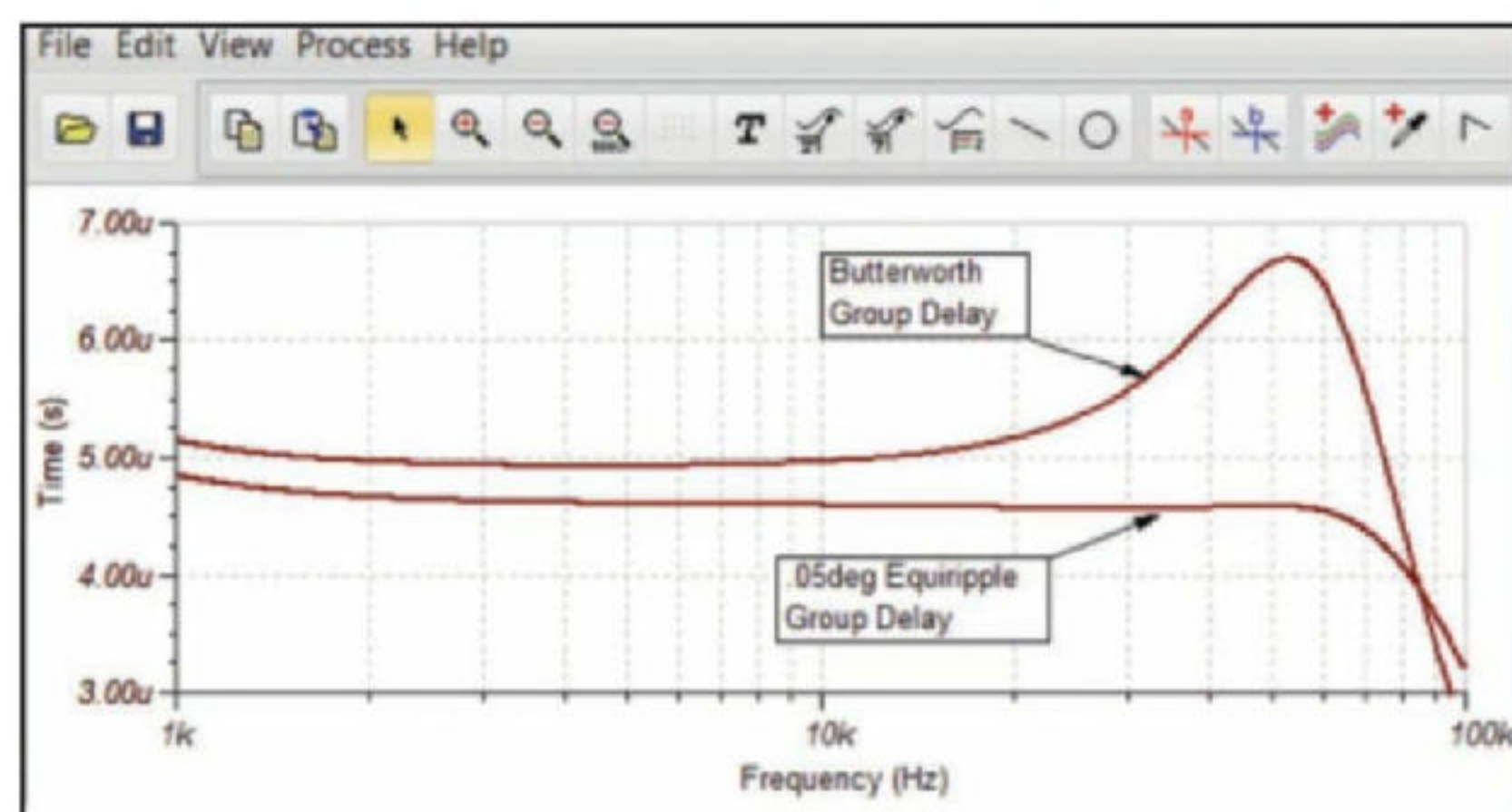


Figure 3: Group delay is shown for the two third-order active filter options to the ADC input.

Group Delay

While the third-order Butterworth shape used so far is very common in audio, it does suffer from poor group delay through the entire 50 kHz span. Gain flatness and constant group delay are competing goals in any active filter design. There are literally an infinite range of filter shape choices where perhaps an easy one to try (from the Burr-Brown Filterpro design tool) might be the common 0.05° Equiripple design. This shape is not quite as constant group delay as the Bessel filter, but has slightly less roll-off in the target gain flatness range through 50 kHz. Targeting the same 65 kHz cut-off, gain of 1.7 V/V, third-order design using a 0.05° Equiripple target gives the active filter design of **Figure 2** (just adjusting RC values) again using the THS4551 FDA.

The nominal 50 kHz roll-off of -1.74 dB slightly exceeds the Butterworth third-order design of -0.84 dB roll-off. However, the group delay is much more constant through the desired 50 kHz span as shown in **Figure 3**, comparing the Butterworth to Equiripple 65 kHz third-order designs.

Both designs exceed -100 dB attenuation at 6 MHz by more than 10 dB so extending the target cut-off frequency slightly can be used to reduce the 50 kHz roll-off, while still meeting the stopband target. The simple second-order multiple feedback (MFB) filters used here, with the post RC filter, are very effective at broadband attenuation.

Next month, I will continue this topic by discussing loop gain (LG) considerations (that partially determine harmonic distortion) and I will discuss phase margin analysis (and improvement) for this active filter multiple feedback (MFB) solution.

Reference

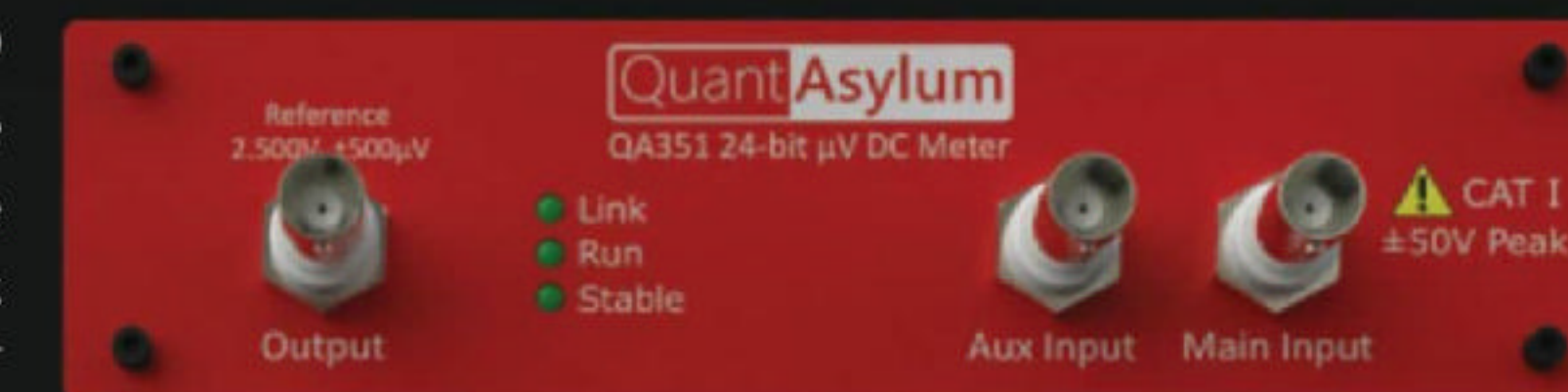
[1] "Low Noise, Precision, 150MHz, Fully Differential Amplifier," Texas Instruments THS4551 Datasheet, revised July 2017, <http://www.ti.com/lit/ds/symlink/th4551.pdf>

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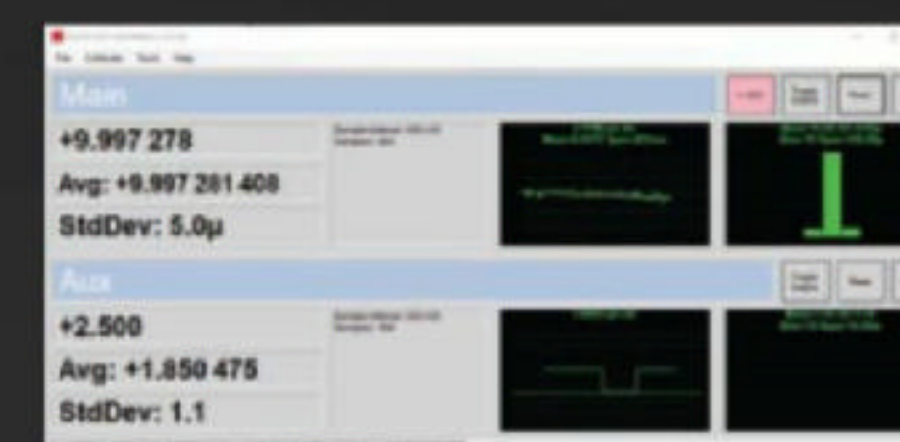
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Audeze Mobius Headphones Reviewed

Almost Essential, Almost Wonderful

High-end planar magnetic headphone manufacturer Audeze breaks out of the mold with the Mobius: a wireless Bluetooth headphone capable of LDAC high-resolution playback, virtualized surround sound, and a number of surprises Audeze continues to invent. We explore what this headphone can do, the use-cases for its unique features, and the limitations of this boldly conceived product.

By
Luke McCready
(United States)

Sometimes a fascinating product passes right underneath our noses because we don't look closely enough to see it's potential. I can't think of a better example than the often misunderstood Audeze Mobius, an over-the-head pair of surround headphones generally marketed toward gamers. Many reviewers explored its form factor, sound quality, and accessories. But Mobius aims to be a platform for creatives and professionals as well, and is chock-full of unique features and untapped potential.

Meet Audeze

Audeze (pronounced like "odyssey") is a high-end headphone company based out of Southern California; founded in 2008. I got the chance to interview Audeze CEO and co-founder Sankar Thiagasamudram, who shared how he and a friend met an aerospace materials engineer with a film perfect for planar magnetic drivers. And from an Orange County, CA, garage, Audeze grew into a company producing top-quality headphones widely respected in the audiophile community.

All Audeze headphones, from the bottom to the top of its product stack and even the in-ear models, use planar magnetic drivers, consisting of an extremely thin film with electrical traces sandwiched between

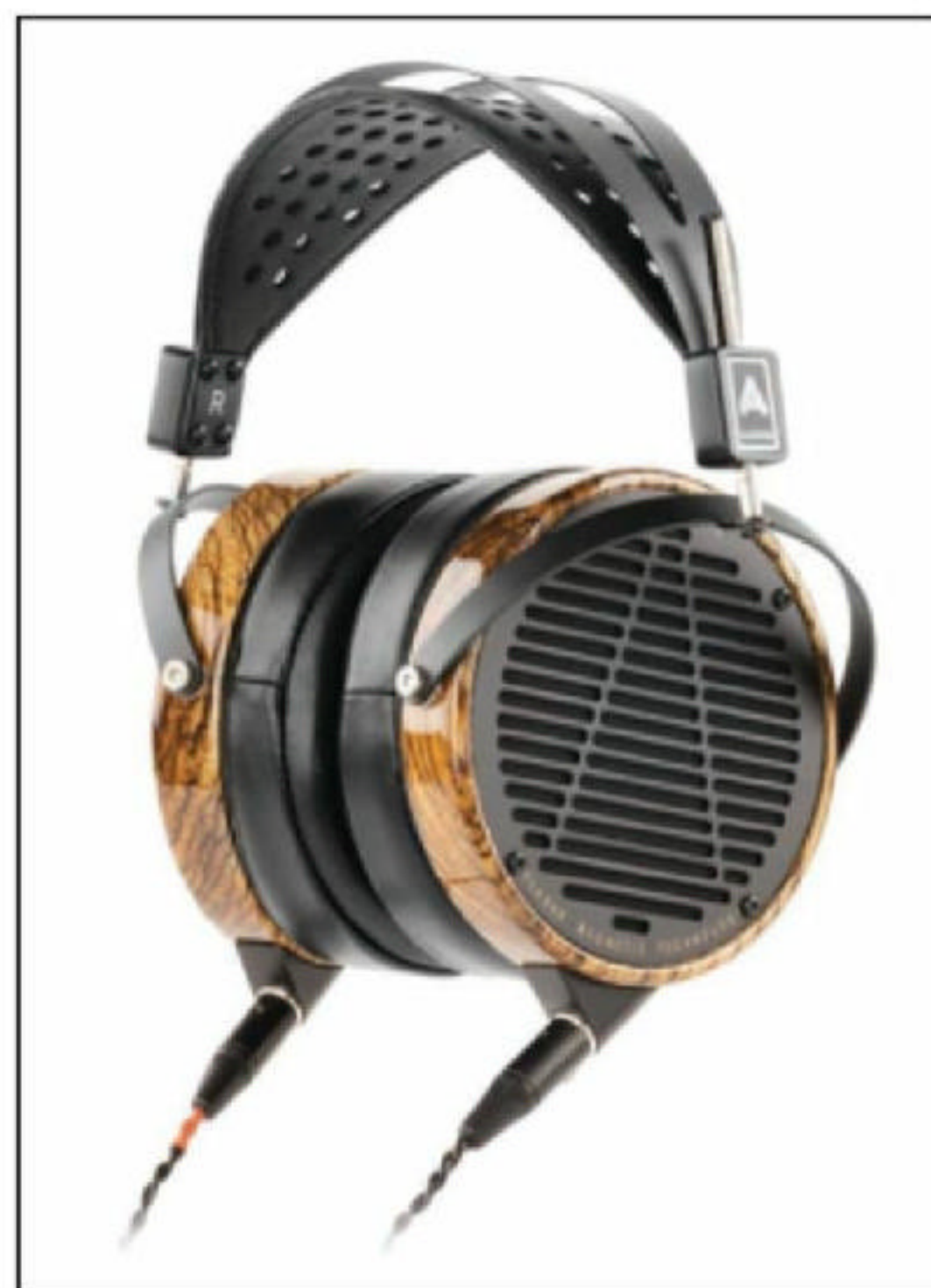


Photo 1: The LCD-3, headphones more typical for Audeze, feature planar magnetic drivers like all of the brand's products.

two arrays of rare earth magnets. Because the driver is so light, in theory, planar magnetic drivers have excellent transient response and low distortion. Thiagasamudram takes pride in the development and manufacture of Audeze headphones. "All our transducers are designed and manufactured in-house by our R&D team." I've had the pleasure of listening to the LCD-3, a \$2,000 pair of over-the-ear headphones (see **Photo 1**), and I found them exceptional. The Audeze over-the-ear lineup runs from the \$400 Mobius to \$4,000.

The Unusual Mobius

Mobius is the most unique product Audeze has ever brought to market: it not only packs planar magnetic drivers into a wireless Bluetooth headset, a very rare design choice, but it goes all-in on software-simulated surround sound, a feature at the very heart of Mobius (see **Photo 2**).

Thiagasamudram shared information about his history with Harvey Landy of NBCUniversal, who oversees digital delivery of movies to streaming providers. Landy's team struggled to format and quality-check the numerous versions of each movie required for varying languages, subtitles, and video formats. "In early 2016, Landy knew that there was a need for change in their workflow with 5.1 and 7.1 deliverables becoming the norm," Thiagasamudram said. "We reached out to Waves for help with 3D audio emulation." And Mobius was born as a tool to help post-production houses expeditiously perform quality control (QC) checks on movies.

And the vision for Mobius evolved: if a product is designed to excel in virtual surround, gamers become an obvious market. Mobius was fitted with an optional microphone boom for teammate communication in multiplayer gaming. As gamers are consumers, wireless Bluetooth connection becomes a compelling convenience. To appeal to the audio enthusiast interested in wireless planar magnetic headphones, Mobius includes a Bluetooth 5.0 radio and support for Sony's LDAC, a high-quality codec with the ability to adjust bit-depth per frequency band of audio. "We wanted to make sure that the Bluetooth experience on Mobius was very close to the wired performance," Thiagasamudram said. "LDAC can go up to 96 kHz over Bluetooth, so we used LDAC."

Even though Audeze is an established company, it opted to crowdfund the launch of Mobius on Indiegogo. I spoke with Jaren Cloud, Marketing Manager of Audeze, who shared why: "Not only was Mobius targeting a completely new market for us—gamers—but it was also the first product of its kind ... Indiegogo gave us a platform to reach a lot of new customers and to gauge how many units would be needed."



Photo 2: The unusual Mobius: the wireless Bluetooth surround sound headset with planar magnetic drivers.

Cloud was excited. "Mobius can do a lot of things that other headphones can't: onboard multichannel processing, head-related transfer function (HRTF) with multiple 3D modes, firmware updates that expand functionality—the list goes on." And Audeze has yet another surprise in store.

Head-Tracking

Waves Audio, a software developer specializing in mixing plug-ins for studio and live sound, launched a Kickstarter campaign in 2016 with the aim of "3D audio on any headphones." The company designed Nx Virtual Mix Room, a plug-in to virtualize stereo and surround mixing within an acoustically treated environment for the headphones you already own. Waves designed the Nx plug-in to combine with its Nx Head Tracker, a Bluetooth device to attach to the bridge of a user's headphones (see **Figure 1**). Using Nx Virtual Mix Room, the Nx Head Tracker follows the position and orientation of the wearer's head along multiple axes: when you turn your head left, the soundscape stays fixed, seeming to come from your right.



Figure 1: Waves Nx Head Tracker technology is incorporated into Mobius for use with and without Waves Nx software.



Photo 3: Each Mobius earcup features a single 100 mm planar magnetic driver, relying on DSP to virtualize surround sound.

Audeze partnered with Waves to incorporate head-tracking directly into Mobius without mounting a separate, battery-reliant device to the headphones. Head-tracking could be useful not only to audio engineers working to mix surround material on headphones, but also movie watchers and gamers looking for extra realism in their entertainment. "Head-tracking is handled by a built-in gyroscope and accelerometer," Cloud said. "This is the exact same technology found inside the Waves NX Head Tracker. It enhances the realism of the out-of-head emulation, which is critical when using Mobius as a tool for any kind of multichannel or spatial audio production." Thiagasamudram added, "We work very closely together with Waves, and not just on the management level. Our developers also work closely with theirs to make sure the user experience is seamless."

Head Gestures, another Mobius first, utilizes head-tracking to turn head tilts and nods into the hotkeys of your choosing. It works well, though I imagine a gamer in need of more controls would prefer a mouse with macro buttons. Cloud said, "Head Gestures can also be used as a supplemental tool for gamers with disabilities, which is an application that is of particular interest to us." I have a friend with cerebral palsy that enjoys gaming but struggles to play with excellence as his hands lack coordination.

Mobius's Head Gestures could be a valuable tool for him and others like him.

Inside Mobius

But quality headphones are more than just tech. Thiagasamudram shared on the design of the drivers: "Our goal was to create a transducer that would offer the maximum efficiency and output for a given headphone size." The design team opted on a single 100 mm planar magnetic driver per ear cup (see **Photo 3**). "We wanted a very linear transducer that can reproduce low frequencies down to 20 Hz with essentially no dipole drop off," he said.

With surround sound headphones, there are two paths one can take: true surround headphones pack many small drivers around each ear, whereas virtual surround headphones use one driver per earcup and simulate surround sound with processing. I probed on the decision to implement virtual surround, and Thiagasamudram said, "With either approach, you are trying to reproduce very small changes in level, phase, and frequencies to convince your brain you are hearing audio from outside your head. Our opinion is that it's hard to achieve with multiple drivers per ear since it is very difficult to produce the many transducers

within tolerance levels." Also, scaling becomes an issue. "With 7.1.4 being widely used in the field, it's not practical to cram 12 speakers inside a headphone. Using virtual surround allows us to easily implement a scalable solution."

I asked what makes planar magnetic drivers so special. "They offer full range reproduction with very low distortion," Thiagasamudram answered. "With very thin diaphragms, the impulse response is very fast and clean. We add a lot of digital signal processing (DSP) for 3D emulation on top, and since the planar drivers do not distort even when you add significant changes, they are ideally suited for these applications."

AudezeHQ

AudezeHQ is the free desktop software counterpart to Mobius, and its primary function is to customize HRTF, the processing that aims to mimic the intricacies of locationality. AudezeHQ includes adjustments for head circumference and inter-aural arc, and artificial room ambiance that's adjustable to taste (see **Figure 2**).

When Mobius is plugged into a computer via USB, AudezeHQ includes modes for receiving standard stereo, hi-res stereo, and 7.1 surround. Because



Figure 2: AudezeHQ, a desktop application, supports setup and options for the Mobius. The animated head's angle is updated in real-time by head-tracking. HRTF settings are on the left.

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Bluetooth doesn't support multichannel audio, a USB connection is required to listen to surround content. Curiously, for a product with professional production in mind, sample rates are not adjustable—stereo mode is locked to 48 kHz/16-bit, surround mode is locked to 48 kHz/24-bit, and hi-res mode is locked to 96 kHz/24-bit.

AudezeHQ includes Sound Profiles to tailor the EQ of the headphones to individual tastes from the eight provided presets, including choices such as Ballistics, Racing, and Warm. The changes are subtle, which is appreciated. "We don't believe in extreme frequency boosts," Cloud said. He also confirmed that the profiles are limited to EQ changes without further DSP applied. To my ears, Default sounds the most natural, though I'd love to see some custom EQ options to manually tailor the sound.

But it's the Mobius headphones themselves that are something special, not AudezeHQ. I asked Cloud about where the virtualization happens, and he said, "All

surround sound, spatialization, and tracking functions are performed inside the headphones themselves."

Stereo Sound Quality

Mobius, designed to be listened to via USB, is an excellent headphone that's neutral, exacting, and full. Compared to my favored AKG K702, Mobius has a noticeably flatter frequency response—as it should: Audeze admits heavy DSP in tuning the drivers. A fairer test is Mobius against the K702 corrected by Sonarworks True-Fi, a consumer-oriented program that compensates for headphone frequency aberrations. Relative to the Mobius, my K702 with True-Fi sounds more spacious and articulate, though it's benefitting from the advantages of open-backed headphones. Mobius adds a slight forcefulness to midrange frequencies that can be overbearing on

occasion, and it lacks a certain texture and breathiness heard on a singer's voice. That said, Mobius sounds more natural and detailed than most headphones, and the K702 with True-Fi makes for very tough competition. Compared to the closed-back Sennheiser HD 280 Pro, the Mobius sounds significantly more precise, spacious, and with soundstage subtlety beyond left, right, and center. True-Fi helps close the gap with the HD 280 Pro, though it can't add that spaciousness or keep certain midrange frequencies from biting. I feel these are fair comparisons: though Mobius costs more, it also includes far more hardware along with software capability.

However, there are more options than just USB listening. I used my mid-level DAC and headphone amp to feed the analog input of the Mobius. The sound was not bad, but perhaps suffering from some distortion on the input side of the Mobius. Put simply, USB performance sounds like a more expensive headphone relative to analog-in performance (see **Photo 4**).

Thiagasamudram achieved his goal, as Bluetooth sounds similar to wired USB performance. I hear a little more detail and a little greater bass response over USB, but the performance of properly configured Bluetooth is genuinely impressive.

LDAC Performance

Does the LDAC Bluetooth codec make a difference? Certainly, though it takes a bit to configure as you need to enable Android's Developer Options to access LDAC settings. LDAC is not supported on iPhones.

Many devices offer SBC and AAC Bluetooth codecs. Enabling LDAC at 44.1 kHz sounds slightly better than AAC and noticeably better than SBC. The difference is in the grit and texture of sound that gets lost in compression. But things go awry when I switch to a 96 kHz LDAC stream: panning sounds more dramatic, there's an audible low/mid-frequency bloat, and vocals sound less centered and

Photo 5: Mobius was designed to appeal to cinephiles as well as gamers and audio engineers.



Photo 4: Mobius headphones are best driven by USB. While using analog-in, Mobius's battery-powered amplifiers remain in use, and audio is converted from analog to digital and back again.

About the Author

Luke McCready is an audio engineer, a technical writer, and a music lover based out of Denver, CO. He graduated with a degree in audio engineering from North Central University with a passion for audio and a fascination for speaker setup and acoustic treatment.

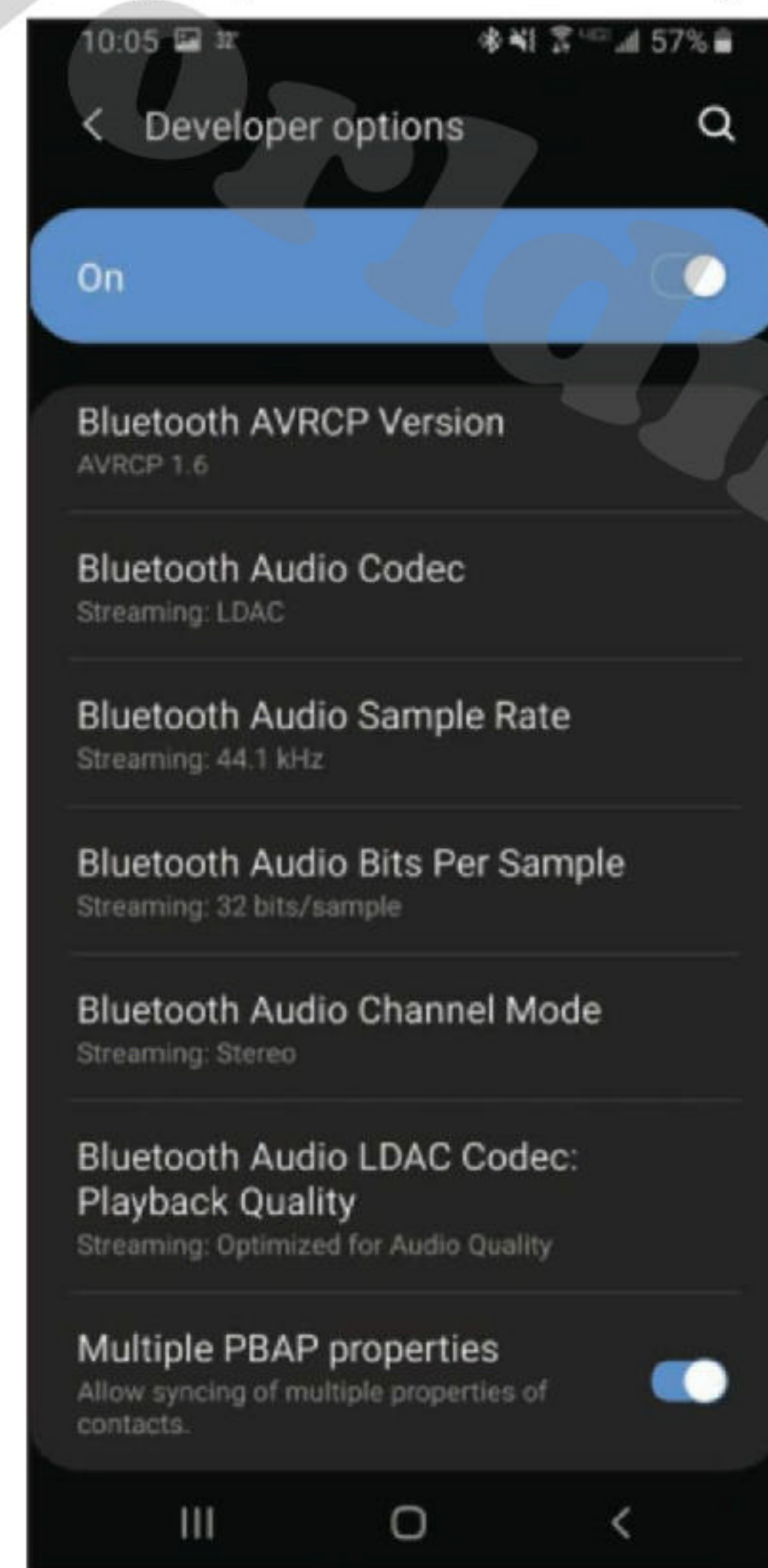


Figure 3: LDAC requires configuration in Android's hidden Developer Options. 44.1 kHz at 990 kbps sounds very similar to wired USB performance.

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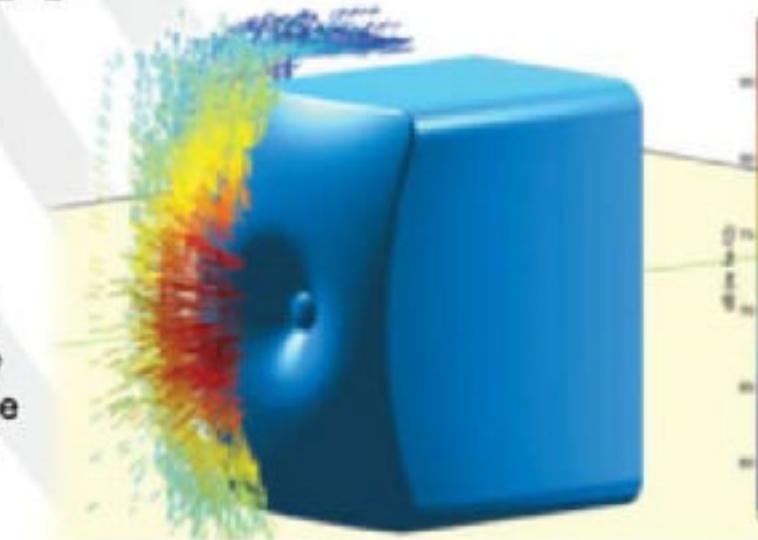
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focused. I experienced this across two media players and a streaming service, testing with uncompressed hi-res files, uncompressed CD-quality files, and MP3 streaming. Hi-res sample rates over LDAC are far from purist: 44.1 kHz LDAC sounds significantly closer to a wired connection than 96 kHz LDAC. What clearly sounds like additional processing could be due to how Mobius's DSP is programmed to handle each codec, or more likely, the LDAC encoder in my Galaxy s9 phone. Regardless, locking LDAC to 990 kbps and 44.1 kHz ensures exceptional performance (see **Figure 3**).

On the topic of LDAC, www.SoundGuys.com wrote up an excellent exploration (linked under Resources) into the frequency response, noise floor, and real-world bitrates for LDAC, discovering that Sony's "hi-res" is not exactly true hi-res, connection speeds often drop from the top tier of 990 kbps, and that frequency response and noise floor suffer as the connection speed drops. Admittedly, 330 kbps sounds pretty brittle, but when forcing 990 kbps, I experienced dropouts only rarely with my Samsung Galaxy s9 in my pocket or a room or two away. All things considered, LDAC allows for

above-CD-quality file playback, and the experience of CD-quality content with LDAC is excellent: imperfect technologies can still reap benefits.

Configuring the Surround Modes

There are multiple modes for 3D audio within Mobius, and you cycle through them with the physical button labeled "3D." The first two modes are variations on surround emulation with head-tracking, the third skips the head-tracking, and the fourth disables 3D audio, treating Mobius as stereo headphones even in 7.1 mode. Thiagasamudram clarified, "When 3D is switched off, we do a fold-down without any 3D emulation."

The first time I twisted in my desk chair with 3D audio on, I was sure I had left my desktop speakers playing in addition to the headphones. The head-tracking and Mobius's ability to localize sounds in the stereo spectrum is effective, but for it to work there has to be some room emulation enabled. The effect is less realistic as you spin all the way around: audio imaged to the rear sounds very similar to audio imaged to the front, just with a lower source-to-reverb

ratio and rolled-off highs. I don't find Mobius's rear surround emulation convincing.

I measured my head and dialed the numbers into AudezeHQ. Accurate measurements do not make surround emulation more convincing, but they do make 3D modes sound more neutral. The Room Ambiance slider is adjustable. Above 40 sounds awful to my ears, 30 sounds reasonable, 20 is my preferred setting, and 0 sounds relatively studio-dry, though it makes surround emulation even less convincing: raising the slider is a trade-off in that it increases the surround emulation's effectiveness while decreasing the desirability of the sound. Thiagasamudram clarified, "3D is meant to spatialize the audio for you, so even with Room Ambiance turned to 0, there is still HRTF active, and the HRTF changes with your head position. Though HRTF provides clues to the directionality of sound, without room reverb, depth information will be missing, causing the image to collapse."

Turning on any of the 3D modes adds gain, a certain harshness, a distinct lack of clarity, and artificial ambiance even with the Room Ambiance slider at zero. It takes away from the precision of good headphones and sounds noticeably more veiled and lo-fi. "With 3D off, you're just listening to a regular pair of stereo headphones. The difference in distinction/clarity/intimacy would be akin to the difference between listening to headphones vs speakers in a dead room," Thiagasamudram said. "One can't create an artificial sense of space and distance without giving up some of the experience of listening to a regular pair of stereo headphones."

Poor Surround Emulation

Listening to surround-encoded music and concerts with 3D mode on is not convincing: I can't decipher any surround material unless I deliberately game the system by turning 90° away from the screen with head-tracking enabled to hear front channels in one ear and surround channels in the other. Disabling 3D provides the impact and punch that music deserves—the cost of surround emulation with music is not worth the benefit.

Surround sound in movies fares slightly better on the Mobius. It's difficult to pinpoint exactly which sounds are coming from behind, but it does add a sense of immersion (see **Photo 5**). Unfortunately, the sound with 3D mode on is reminiscent of down-firing TV speakers: lacking clarity, definition, and transient response. Because of this, again, I'd rather use Mobius in stereo (see **Figure 4**).

Gaming with Mobius suffers from the same experience: surround content is vague and not worth the penalty to audio clarity on the sounds of gunfire and guns reloading, dialogue, and even footsteps and



Photo 6: Mobius marketing places gamers front-and-center.

environmental effects. The lo-fi aspect of 3D simply removes more immersion than it adds.

Cloud, enthusiastic about gaming (see **Photo 6**), said, "What I always try to emphasize to people is that head-tracking isn't there because we expect you to constantly be turning your head when playing a game—obviously you're not going to do that. The idea is that if you aren't tracking your head's movements, then you aren't creating a convincing real-world listening simulation, because in the real world your head is constantly in motion, and that plays a major role in how your brain localizes sound." Maybe, but I couldn't detect Mobius adjusting to the subtle, continuous movement of my head. I think the best use case for head-tracking in gaming is with an ultrawide monitor.

During testing, I plugged my computer into my home theater for the first time. To make the test fair, I wired up my Andrew Jones-designed Pioneer speakers: a budget but quality 5.1 system. The sound was full and dynamic, and I heard an abundance of surround content I had missed on the headphones; all of it sounding crisp and natural and distinctly surrounding me. It was so much fun that I lost track of time and just played the game. Gaming on a true surround system showed just how lacking virtualized surround headsets are relative to the real thing.

Examining the Shortcomings

One might wonder where the fault lies in the complaints we've collected. To be fair, we need to address each aspect individually: playback quality, the binaural renderer, and the loss of fidelity with surround emulation. I've already shared that the stereo sound quality of Mobius is excellent: not quite world-class, but a fair margin better than most headphones. Playback quality isn't the issue.

Some fault lies with the nature of binaural rendering: Acoustics are complicated. The physics

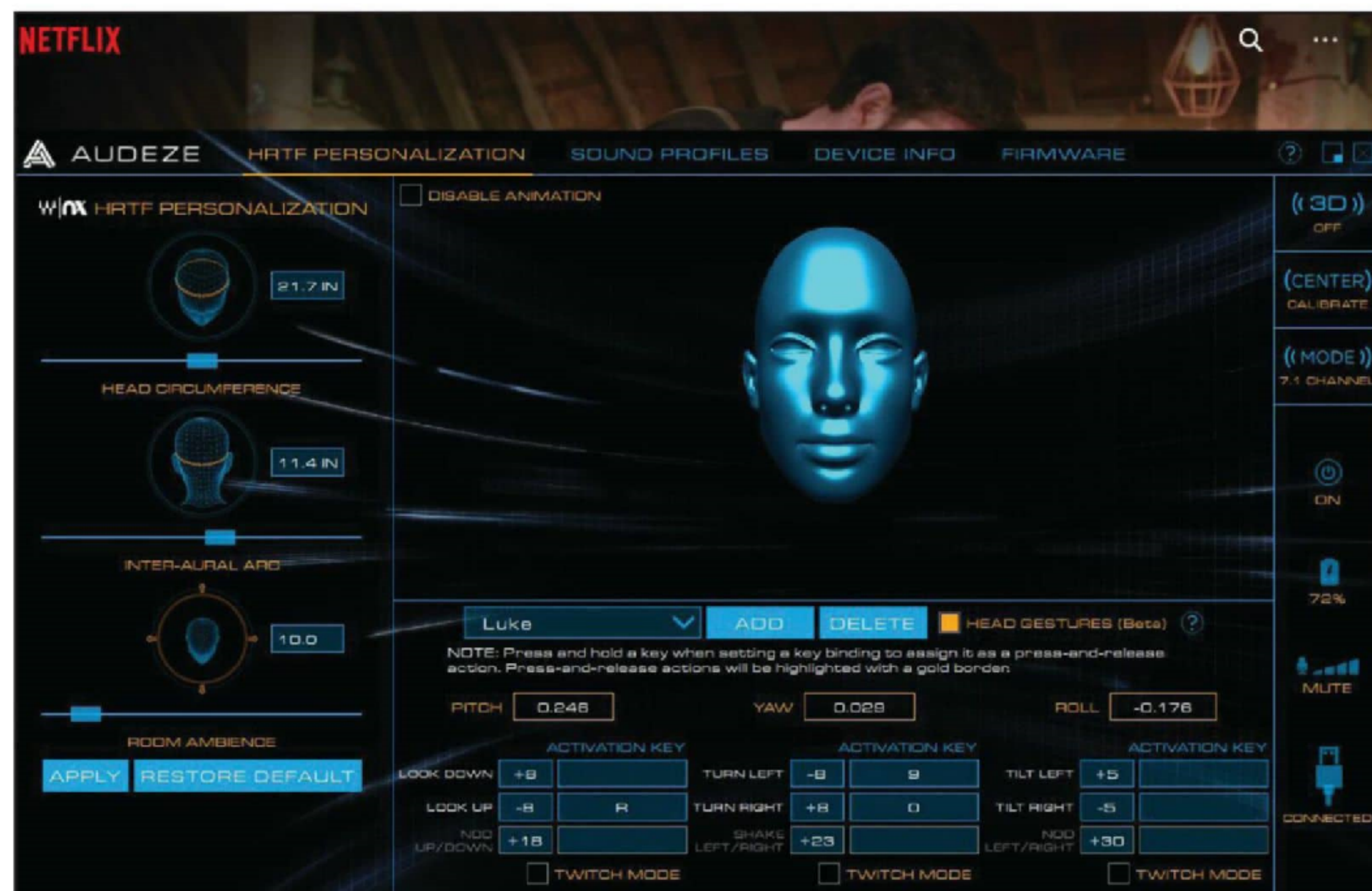


Figure 4: Browser-based Netflix is limited to stereo, but the Netflix desktop app can stream surround sound. Still, because of the fidelity cost of Mobius's binaural renderer, I prefer 3D mode off, with any surround channels summed down to stereo.

Photo 7: Audeze CEO and co-founder Sankar Thiagasamudram



of sound passing through the skull and surrounding tissues are complicated. Moreover, every human has differently shaped ears, and locationality is different for every person. "We're accustomed to listening through our own ears and our brain is not easily fooled when a different HRTF is used," Thiagasamudram said. "However, the use of head tracking in Mobius helps aid in our brain adjusting to the new HRTF; it does need some time."

We may not have the technology to properly emulate surround sound on headphones in 2020. And if we do, it's more likely to be in the form of something like the Realiser A16 by Smyth-Research, a powerful standalone surround processor that relies on exacting measurements per ear of each user. And Thiagasamudram (see **Photo 7**) may agree. "The best form of virtualization is making true measurements of one's HRTF in the setup/room one wants to emulate," he said. "Though it is technically possible to make such HRTF measurements, it's not practical to do such



Figure 5: Waves Nx, the \$10 consumer desktop app compatible with all headphones, performs the same 5.1/7.1 surround emulation as Mobius with less degradation to sound quality.

measurements on a large scale. Moreover, using high quality, true-HRTF measured in real-time within the headphone requires memory and processing power that would not be possible to squeeze into headphones with the technology available today."

As for the loss of fidelity, the best test was to pit Mobius's binaural renderer against that of Waves Nx, the \$10 desktop program that contains the same binaural renderer as Mobius, but to be used with any headphone (see **Figure 5**). I tested both renderers with Mobius. Surround emulation in Nx was not superior, but subjectively, the fidelity loss from Mobius's 3D processing is four times worse than Nx's processing. I can forget that Nx is active as I'm watching a movie, but I can't forget when Mobius's 3D mode is on.

Surround Formats and Mobius

A quick look at surround sound formats is worthwhile. In the DVD-era, channel-based sound was the only option. A studio would mix a movie's audio into 5.1 surround, with one channel intended for one speaker.

Object-based sound, with standards like Atmos and DTS:X, allows for sound "objects" to be mixed anywhere in the room, including on the height dimension. Object-based audio is speaker-agnostic in that it doesn't matter if you're listening in a 5.1.2 home theater, a 29.4.18 movie theater, or on a pair of headphones: the mix leaving the studio is the same, to be interpreted by your playback device.

Scene-based sound goes even further by creating an entire sphere or "scene" of audio around the listener that could contain hundreds of sound sources. Ambisonics B-Format is becoming the standard for VR games and 360° video, as it's able to coarsely create the sphere of sound using just four audio channels. And fourth-order Ambisonics, included in the MPEG-H standard, requires at least 25 channels of audio to create and results in a significantly more intricate sphere of audio—enough to feed up to 64 speakers.

Mobius operates on channel-based audio. There is no sensation of height, as 7.1 surround exists on the horizontal dimension. Atmos and MPEG-H support would be a nice addition to the headphone-wearing cinephile, and they're increasingly mandatory to create rough mixes and perform QC checks for movie and TV post-production houses.

Atmos is also quite attractive to gamers, as the leading edge of games are adopting object-based sound just as they are ray-traced lighting. Lacking support for height and object- and scene-based audio, Mobius is noticeably feature-uncomplete. Still, Mobius is an evolving platform, and the decision to use virtual surround makes this a possible addition via firmware. As Thiagasamudram said, virtual surround is scalable.

Professional Production

Mobius was initially designed as a tool for performing surround mix QC (see **Photo 8**). As an engineer that works primarily in stereo, I asked Cloud if Mobius has any advantages in the stereo realm. "The head-tracking and spatialization functions of Mobius don't really provide any dramatic benefits for stereo workflows," he said. "It's the surround monitoring potential of the Mobius that really makes it shine as an audio production tool."

For my first production test, I used Mobius as the output for a 7.1 channel surround mix with numerous vocal layers panned in all directions. Using 3D with head-tracking, it was pretty fun to swivel my chair and hear each vocal layer come marginally more into focus. But the binaural rendering was unconvincing, particularly when dragging a single voice around the sound field. The 3D hit to fidelity makes Mobius sound far too skewed to create a surround mix that will translate. And significantly, Nx Virtual Mix Room performs the same emulation with a smaller sonic penalty than Mobius.

If Mobius's production value is that its head-tracking works with Nx, it's not making a compelling argument: Try as I might, I couldn't get any of the

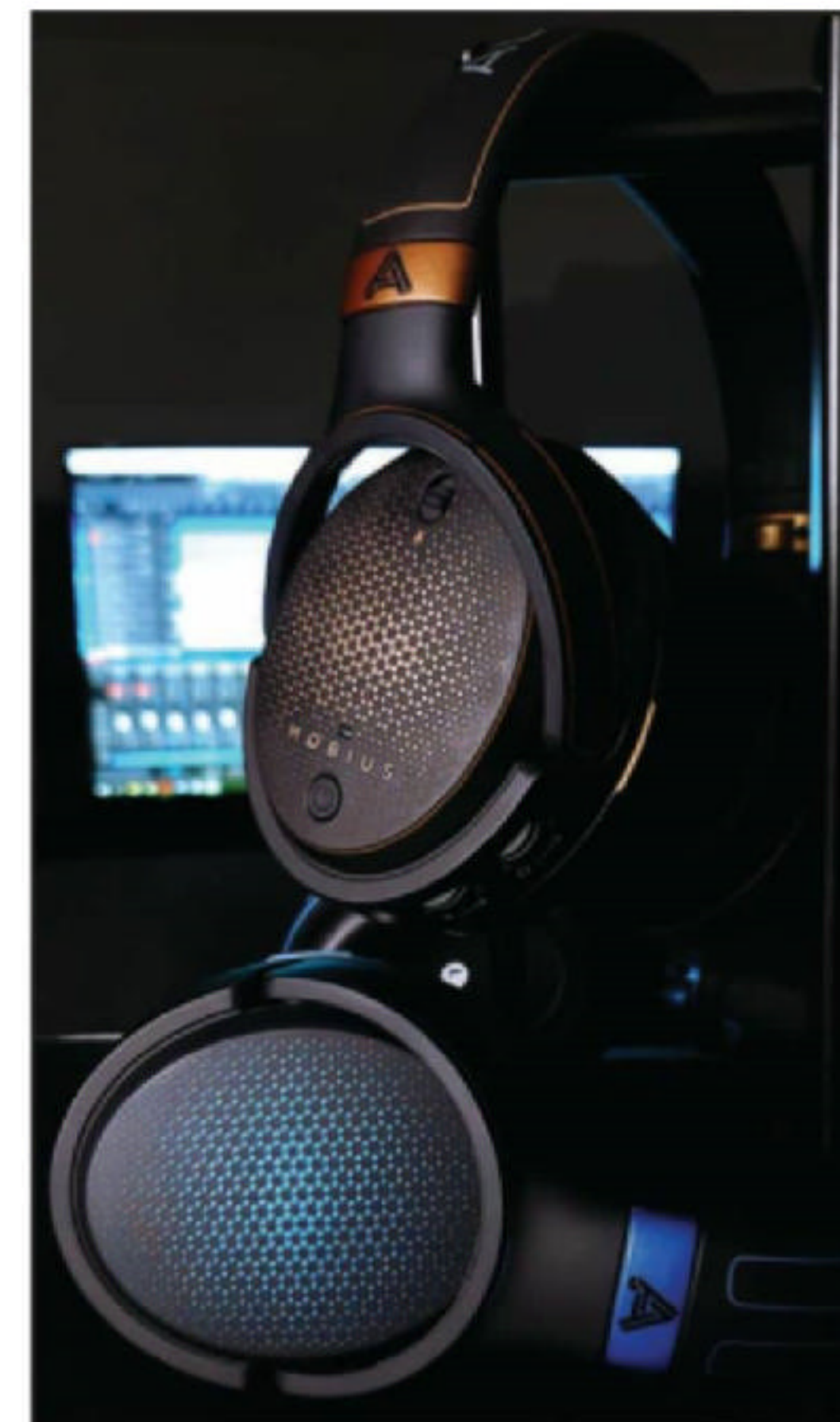


Photo 8: The original vision for Mobius targeted professional audio production first.

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Virtins Technology

Waves plugins to recognize Mobius as my head-tracker. Google and YouTube offered no solutions, and the Audeze website only added, "Because Mobius uses the same technology that is found inside Waves' own Nx Head-Tracker, it can route all of its head-tracking information straight to the plugin."

For my second production test, I explored the uses for Nx Virtual Mix Room in a stereo mix session. Using Nx on every channel allows for surround panning and binaural rendering before the audio heads to the master bus. A neat trick, though lacking realism. And as the master bus is stereo, Mobius is a passive participant: a quality stereo headphone with no other useful additions.

For my third production test, I created a B-Format Ambisonics mix using Waves B360 on every track (see **Figure 6**), allowing me to position sounds anywhere in a sphere, including height. Like Nx, the effect was far from spell-binding. Supposing Mobius can be used for head-tracking with Waves plug-ins, it could help one create an Ambisonics mix. But again, Mobius is operating as a traditional two-channel headphone.

Challenging the Narrative

"Mobius is starting to see some use as a tool for creators in the XR space [of virtual reality and 360° video]," Cloud said, "and I think Mobius makes the idea of spatial audio experiences a lot more accessible and relevant. Jonathan Rowden, CEO of Overworld, has

been using Mobius to help with that kind of content production for over a year now."

I struggle with the claim that Mobius is a tool for production work, XR or otherwise. Used as a 7.1 sound card, Mobius can't play back surround content without 3D turned on. And Nx plug-ins in the DAW already offer the same binaural rendering for any headphones with a lower sonic penalty. The only advantage Mobius has for surround work over traditional stereo headphones is the built-in head tracker that won't talk to my Waves plug-ins. Every feature advertised in the Mobius Creator Package, from real-time monitoring of Ambisonics to FuMa/AmbiX conversion to DAW support, is referring to software from Waves, not hardware from Audeze.

Conclusion

I'm beginning to think of Mobius less as a pair of surround headphones and more as a platform, ripe for innovation and continuous improvement. Audeze's ability to add functionality after release and its track record of doing so is reminiscent of Tesla, rolling out safety features and even acceleration and braking enhancements to its vehicles after they've been sold. In May of last year, the headphones gained a new 3D mode, improved bass response, and even improved imaging. And recently, Audeze updated Mobius with an ASIO driver, aiding in bit-perfect music playback and DAW integration. "We see Mobius as a platform

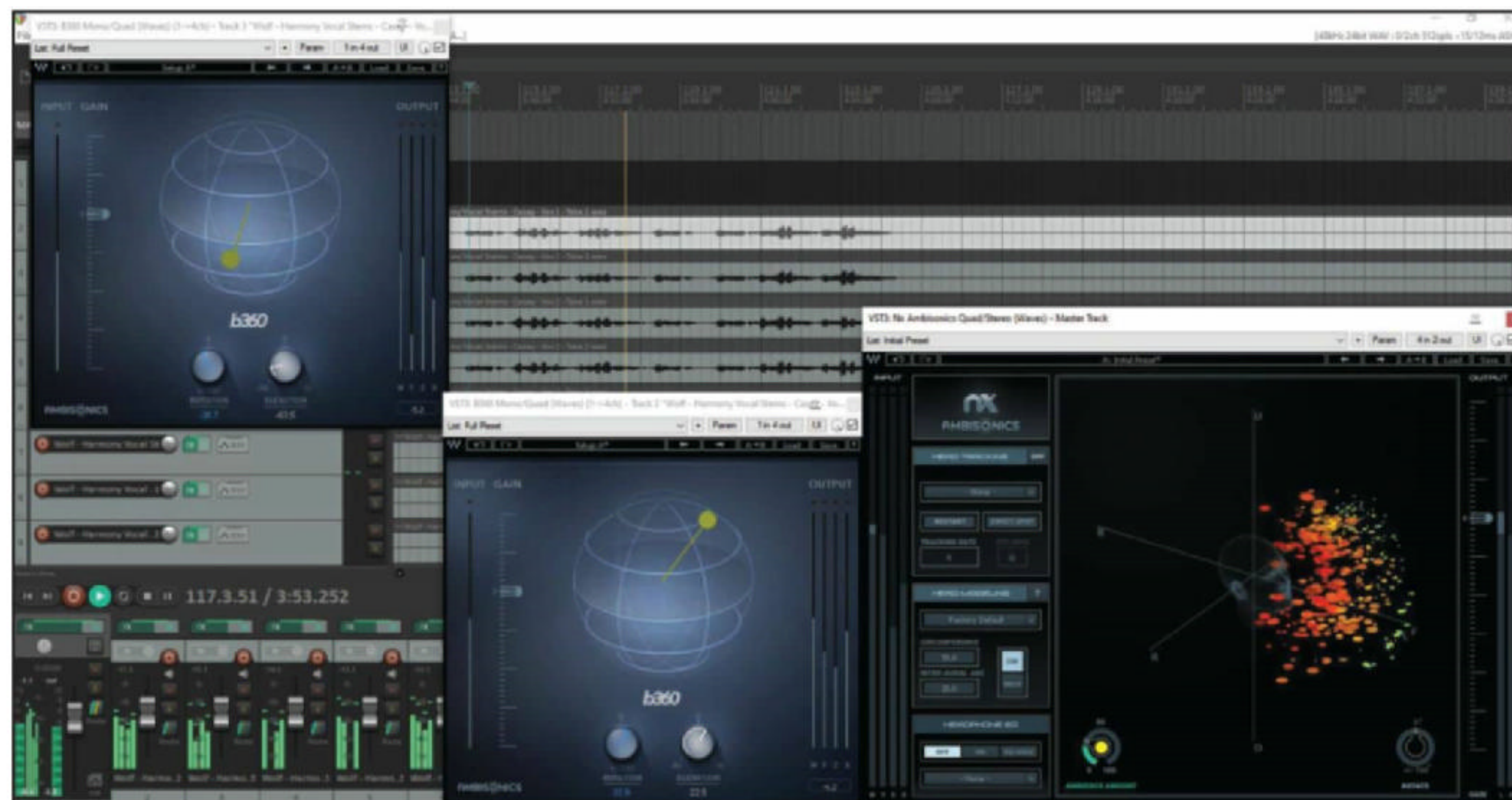


Figure 6: The Audeze FAQ on support for Ambisonics is misleading—I now understand it to mean that Mobius does not need to decode Ambisonics B-Format because Waves has a plugin for that purpose.



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that continues to evolve,” Thiagasamudram said. “We get feedback from customers, and if possible, implement these changes and push them out using firmware updates.”

We’ve thoroughly covered the issues in using Mobius 3D for gaming and consuming movies and surround-encoded music. The case for Mobius as a production tool is equally weak with the possible exception of XR. And though artificial reality feels in its infancy, virtually all current VR headsets offer native head-tracking, and in many cases, body-tracking too.

I’m impressed by everything Mobius can do in theory, but it feels like an unfinished solution for a large handful of problems. Binaural rendering at a smaller sonic penalty feels very much within reach for Mobius over future updates. But in its current state, I would rather use headphones I already own and optionally use the \$10 Waves Nx as a mildly-effective binaural renderer for my multi-channel PC content.

As much as I want Mobius to be more, I can only recommend it for the standard use-cases of listening to stereo content over wired USB and wireless

44.1 kHz LDAC (see **Photo 9**). And for those tasks, Mobius genuinely excels. Truly, a single major improvement would create a large market for Mobius. If Atmos and MPEG-H are added and nothing else, Mobius would be useable for QC checks on modern film projects. If 3D modes can operate at a lower cost to fidelity and nothing else, Mobius becomes the gold-standard of gaming headsets. And if surround emulation improves and nothing else, Mobius would become the must-have tool for surround mixing engineers, hobbyists and professionals alike.

There’s good news and bad news on those fronts. When I asked if Atmos support is possible, Thiagasamudram simply stated, “The current version of Mobius will not support Atmos. The processor is quite powerful, but there is no easy way to send object-based audio over USB. But doing it on the software side is possible even today.” As 7.1 mode requires a USB connection to a computer, I think an AudezeHQ-based software decode of Atmos and MPEG-H would be perfectly convenient. The emulation could happen on the software side with the binaural render sent as

a stereo signal to the headphones.

I asked if Mobius will ever part from the Wave’s Nx renderer, perhaps because Audeze is cooking up something more advanced. “No,” he said. “Mobius models will use Nx.” But that’s not to say Waves’ emulation might not improve over time. Thiagasamudram later shared, “At NAMM, we will be showing a new plugin for our (non-Mobius) headphones that uses machine-learning HRTF to emulate 3D audio.” That could be very compelling.

Fortunately, Mobius is only the beginning, as Audeze is working on other high-tech projects beyond

traditional, analog headphones. “2020 will perhaps be the busiest year for us,” Thiagasamudram said. “Some of our new projects will be radical departures for Audeze. We plan to do an event early in Q2 showing all the products.” If Mobius ever grows to feature more convincing surround emulation with greater fidelity, even to include object- and scene-based decoding via a host PC, it would be a truly incredible product. But even if it doesn’t, Mobius appears to be just the first such radical infusion of hi-fi and high-tech from Audeze, and future products are only going to build beyond this innovative platform.



Photo 9: I want to like Mobius, but it’s made special by everything related to its surround emulation, and its surround emulation just isn’t as good as it needs to be.

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Harman International Industries | www.akg.com

Audio-gd Fun integrated DAC, preamp, and headphone amplifier
Audio-gd | www.audio-gd.com

Denon AVR-X1500H receiver
Denon US | usa.denon.com

Pioneer SP-FS51-LR floorstanding speakers and matching surround set
Pioneer Corp. | global.pioneer

Reaper 6 DAW
Reaper | www.reaper.fm

Samsung Galaxy S9 smartphone
Samsung | www.samsung.com

Sennheiser HD 280 Pro closed-back headphones
Sennheiser | en-us.sennheiser.com

Studio One 4.6 DAW
PreSonus | www.presonus.com

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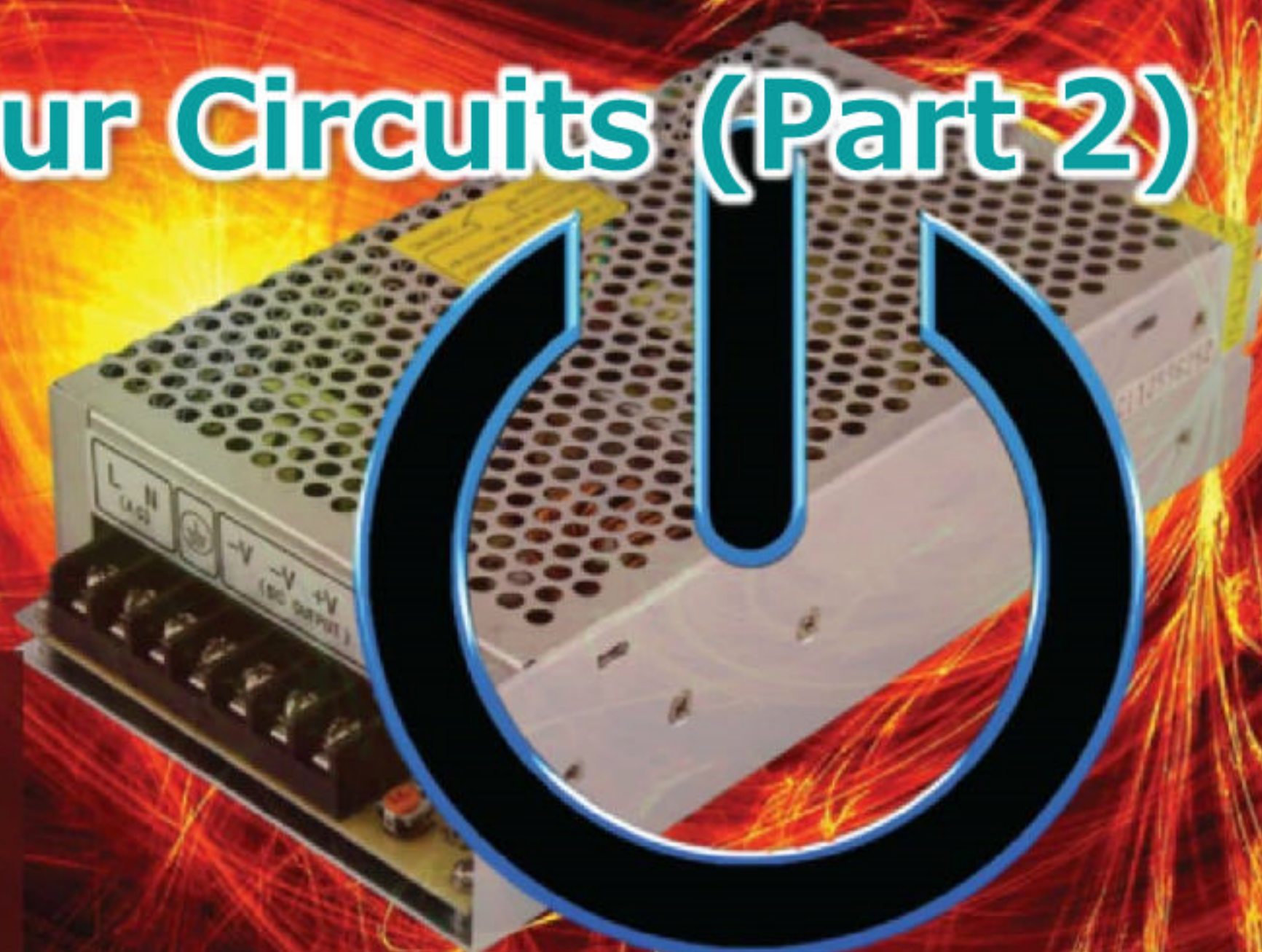
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Powering Your Circuits (Part 2)

AC Power Supplies

By
Ed Simon
(United States)

Ed Simon has provided *audioXpress* with a six-part article series that details ways to power circuits. After reviewing battery power in Part 1, Ed now tackles all aspects in AC power supplies, starting with transformers.



The second most popular method of powering sound equipment is to convert the alternating current (AC) power line into useful lower voltage direct current (DC). (Vacuum tube aficionados may actually want to step up the AC line voltage.)

The easiest method to safely do this is with a transformer-based linear power supply. In its simplest form, this would require a chassis of some sort to hold the parts and provide protection from the line voltage terminals, a power cord with a plug, a strain relief or IEC connector set, a fuse and holder, a transformer, a diode, and a capacitor.

A Transformer's Basic Operation

The basic operation of the typical 120 V, 60 Hz AC to DC power supply is that AC power goes from the local AC line source through the fuse to the power transformer (see **Figure 1**). The transformer usually steps down the voltage and provides isolation from the power line. The diode only lets current pass from the anode connected to the transformer to the cathode into the capacitor. This gives us a positive voltage at the diode-capacitor junction referenced to the direct transformer-capacitor connection. In most applications the safety ground wire (not shown here) will connect to the chassis and the transformer's mounting lugs. If there is an insulation fault the AC line's fuse or circuit breaker should open.

The basic operation of a transformer is to use an electric current to produce a magnetic field (called flux), coupling the input

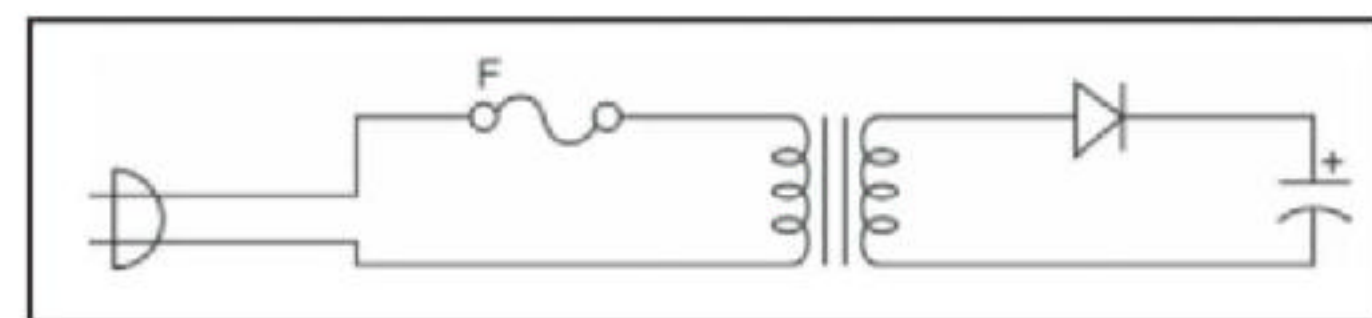


Figure 1: The basic operation of the typical 120 V, 60 Hz AC to DC power supply is that AC power goes from the local AC line source through the fuse to the power transformer.

winding to the output winding(s), which turns that back into an electric current. The voltage depends on the ratio of input turns to output turns. A core (see **Figure 2**) is used in power transformers to increase and restrict the available magnetic field to the windings. The core is made of alloys of iron and other materials.

Now the transfer from current to magnetic field is not completely linear. At first, any remaining magnetism in the reverse direction has to be cleared, and then as the current continues to be applied all, of the magnetic domains eventually line up with it. Now when all the domains have lined up, the current to magnetism relationship slows down quite a bit. The more magnetic field produced by the current, the faster the changes occur.

With an AC power line, the current reverses every cycle so the magnetic field also changes. As the AC power line is never a true sine wave, there may be a trend for the operating point to shift from zero. If there is a lot of imbalance on the AC line (often called DC on the AC Line), the operating point may move enough to cause operation more into the bent tips of the BH curve (see **Figure 3**). This will increase distortion and to some extent, result in less capacity for the core to carry the magnetic field.

One problem is that the core material, although capable of handling more magnetic flux than air, has this fully magnetized limit. When you reach this limit, it is called saturation. At saturation, the transformer begins to transfer the energy much less efficiently until the current (and magnetic field) reverses. If under-sized, the core also saturates in the reverse direction, repeating the process.

When any core saturates, it behaves in some ways as if there was a short circuit. To avoid saturation more material is used in the core. The lower the frequency, the longer the time is available to saturate the core so more material is required. At higher frequencies less core material is required. To keep costs low, the least amount of magnetic material is used that will allow the transformer to function. So part of the magnetic cycle will be near saturation and not quite linear. The advantage of this is a bit

of greater voltage regulation on the secondary and slightly more charging time for the filter capacitors. The disadvantages are harmonic distortion due to the nonlinear behavior and more heating due to the energy loss.

The Core Design

The core's design is also a consideration. The goal is to get as much of the magnetic flux generated by the primary coil into the secondary coil(s). Also, it is good to have as little as possible leaking out that would affect other components.

The most efficient core is a toroid (donut shape.) The disadvantages are that the windings have to loop through the core. With the increase in efficiency, there can also be very high start-up surge currents, depending on the phase angle of the input AC line during switch on and the magnetic field remaining in the core from prior use.

To overcome the winding problem, the E-I core is used (see **Figure 4**). It is not quite as good as a toroid, but the coils can be wound on a bobbin and then the core assembled around it. The other feature is that when punching steel, two Es could be made face to face and two Is for the core would be created at the same time.

There is a bit more magnetic flux loss at the corners of an E-I core compared to a toroid. The start-up current is lower than the equivalent toroid. In some applications, this leakage flux may be a problem. Of course, when a core goes into saturation, some of the extra magnetic energy tries to complete its path and goes outside the core to do this. As the surrounding air is not a very good magnetic conductor, this excess field will affect surrounding circuitry for a much greater distance than normal.

Virtually all linear power supply transformer cores are made of layers called laminations. By processing the laminations, the electrical resistance on the outside is increased so that the core itself does not act as another secondary winding that is shorted! There is some loss from any core. These induced current inefficiencies are called eddy current losses. They are usually very small for power transformers used directly from the AC line. These losses are of some concern in audio transformers and inductors. They would be a real design issue in switching power supply transformers if a laminated core was used. That is why most switching power supplies use molded ferrite cores. Lower-frequency transformers do not use ferrite core as they saturate at a lower level for a given size.

E-I transformers are often made with the primary coil wound first on an insulating form, an insulating

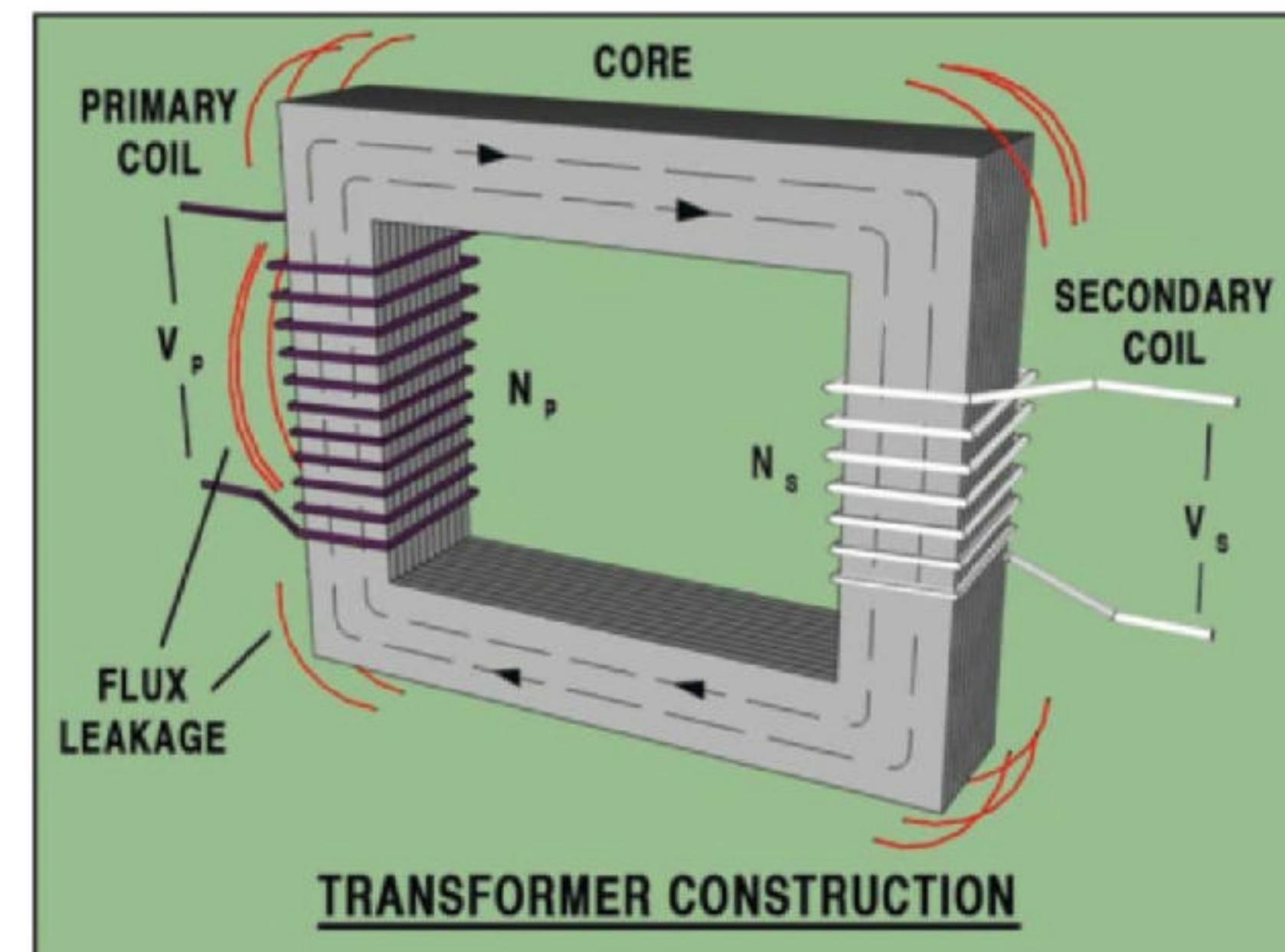


Figure 2: A core is used in power transformers to increase and restrict the available magnetic field to the windings.

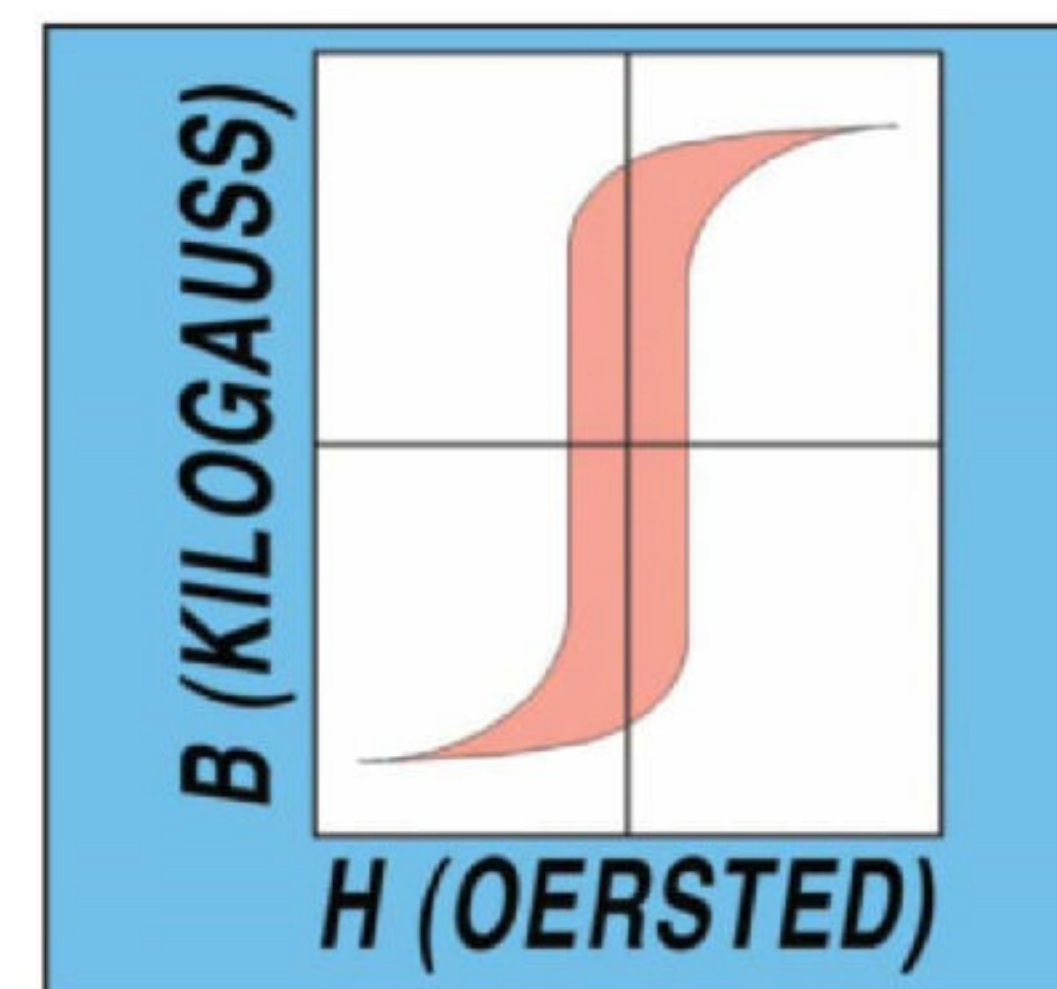


Figure 3: If there is a lot of imbalance on the AC line, the operating point may move enough to cause operation more into the bent tips of the BH curve.



Figure 4: To overcome the winding problem often experienced with the toroid core, an E-I core is used.

layer over the primary coil(s) for extra insulation even greater than what is on the wires, and then the secondary windings go over that. There are two issues. One is that the wire leading to the primary winding that is closest to the core should be connected to the AC power line's neutral lead. That will minimize unwanted capacitance from the power line to the chassis! Many transformers come with a dot or phase markings. These indicate the power is matching in phase to the other windings so marked.

The other issue is that there is also coupling via capacitance from the primary winding to the secondary. Some transformers use a copper shield foil over the primary to reduce this. The foil is wrapped with an insulator so that it does not form a shorted

turn. It also has a wire connection so that it may be grounded. These transformers are often custom-made and rarely available as stock items. A good source for small quantities of these transformers is often to remove them from older retired sound equipment.

A more modern method is to use a bobbin with a divider in the middle so the primary and secondary are wound at the same time next to each other. Some transformers use two dual-winding bobbins. These provide flexibility in that the two primary windings may be placed in series for 240 V operation or in parallel for 120 V use. The same can be done with the dual secondary windings. An advantage for audio is that the coupling between primary and secondary coils is lower. This results in less line noise getting through. (However, they do have more magnetic flux leakage as a general rule.)

Another popular power transformer core is called an O core (see Figure 2). It can be used with a primary on one side and the secondary on the other. If a very thick bobbin and extra insulation are used, this can be good for very high voltages. The more common use is to use two dual bobbins with a primary and secondary next to each other. This design often produces a low-profile transformer that can be PC mounted and used with either 120 V or 240 V lines and can be used as two secondaries, a center tapped secondary, or a double current single primary.

The Energy

Another transformer requirement is that there are enough turns of wire to properly couple the energy into the core. If there are not enough turns then the primary will saturate the core. Too many and the resistance of the wire will drop some excess power and reduce the efficiency of the transformer. For higher power transformers, more current is required, so the core must also be larger.

To make things more interesting, the transformer action is not entirely linear in allowing energy transfer from the primary to secondary. That combined with other sources of loss means that no transformer is 100% efficient. Typically, half of the loss is from magnetic issues and half is from the resistance of the windings. The wasted energy heats up the transformer.

In an ideal transformer, when there is no load placed on it, there would be no current drawn from the input. In the real world, there's always some minimum current draw. This is often called the "magnetization" loss.

It seems intuitive that if you have 250 turns on the primary and 25 turns on the secondary you will have a 10:1 voltage step down transformer. Due

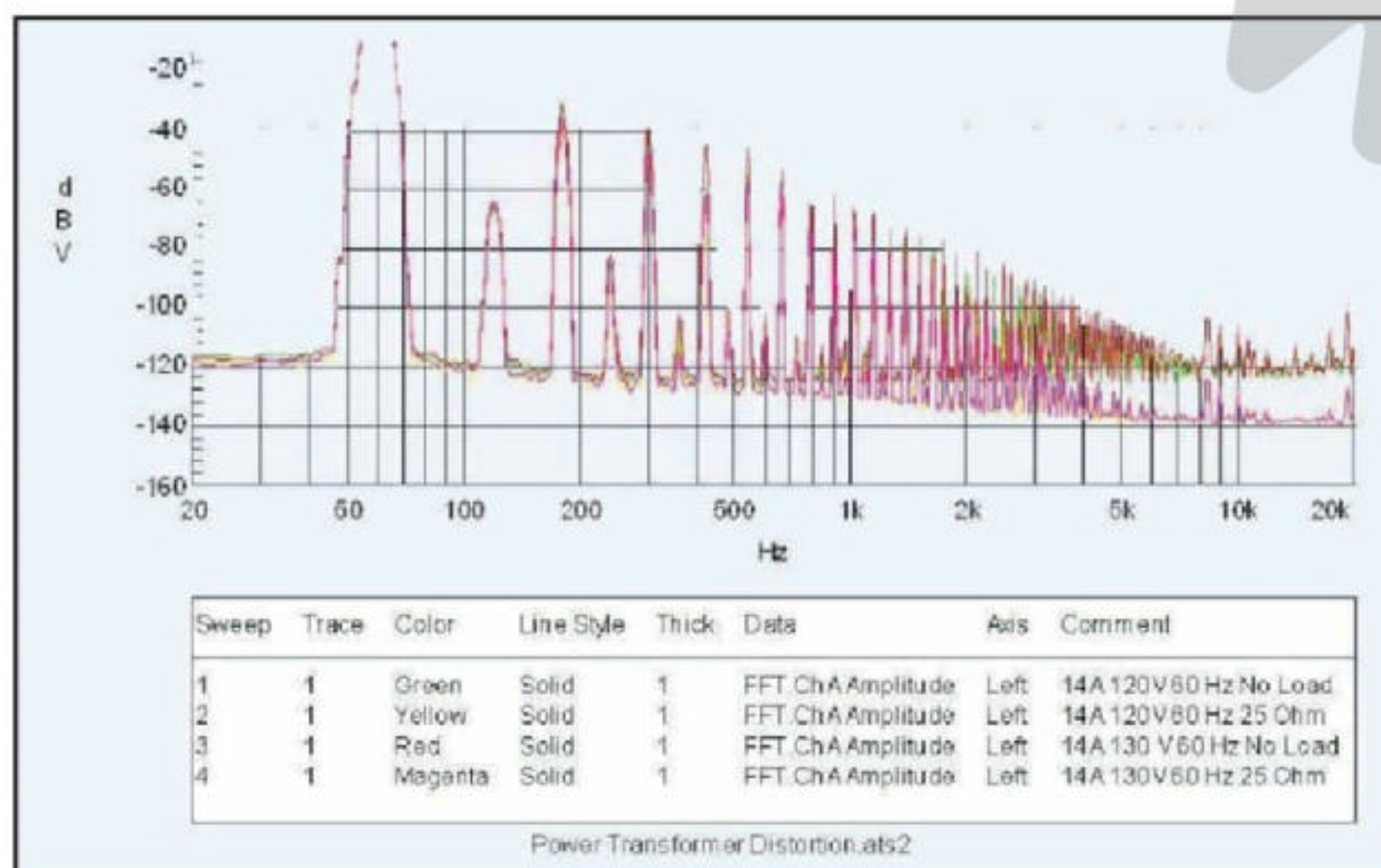


Figure 5: A fairly standard E-I core 20 VA rated Signal Transformer model 14A-20-24 showed an output voltage of 24 VAC RMS when tested.

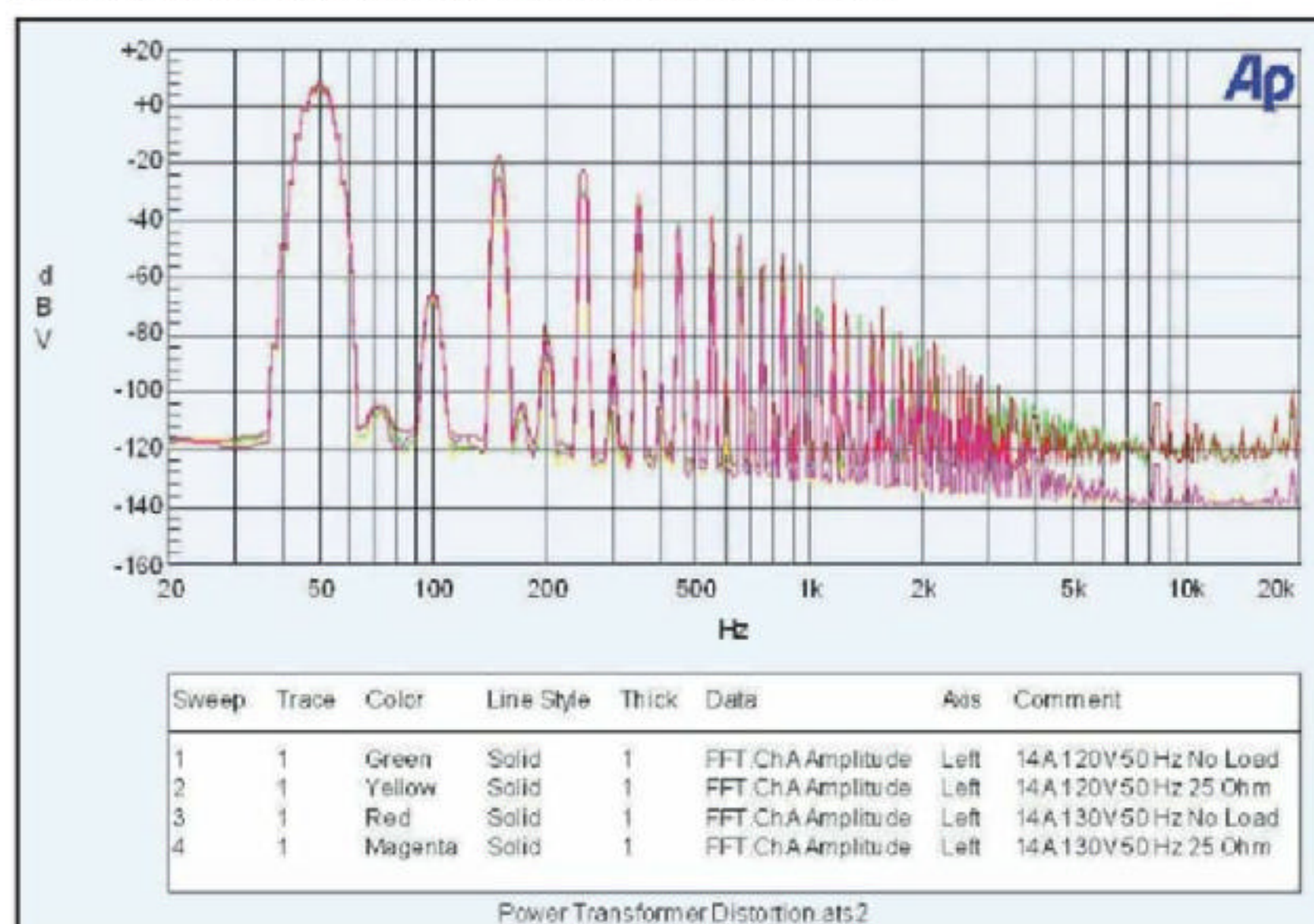


Figure 6: Here, I used the same transformer that was tested in Figure 5, but I conducted the test at 50 Hz.

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to the losses, it may actually have 27 turns on the secondary. Now if the transformer is not loaded to the maximum current for which it is designed, some of the losses will be lower. The result is that under light loads, the output voltage can be higher. This is often called the transformer's regulation. For small transformers, it might be a change of as much as 20%. Large transformers are frequently designed with lower losses so they do not overheat. They might have regulation listed as 5% or even lower. The output voltage rating of a transformer is given for the rated load. So the output voltage will drift higher at lower currents.

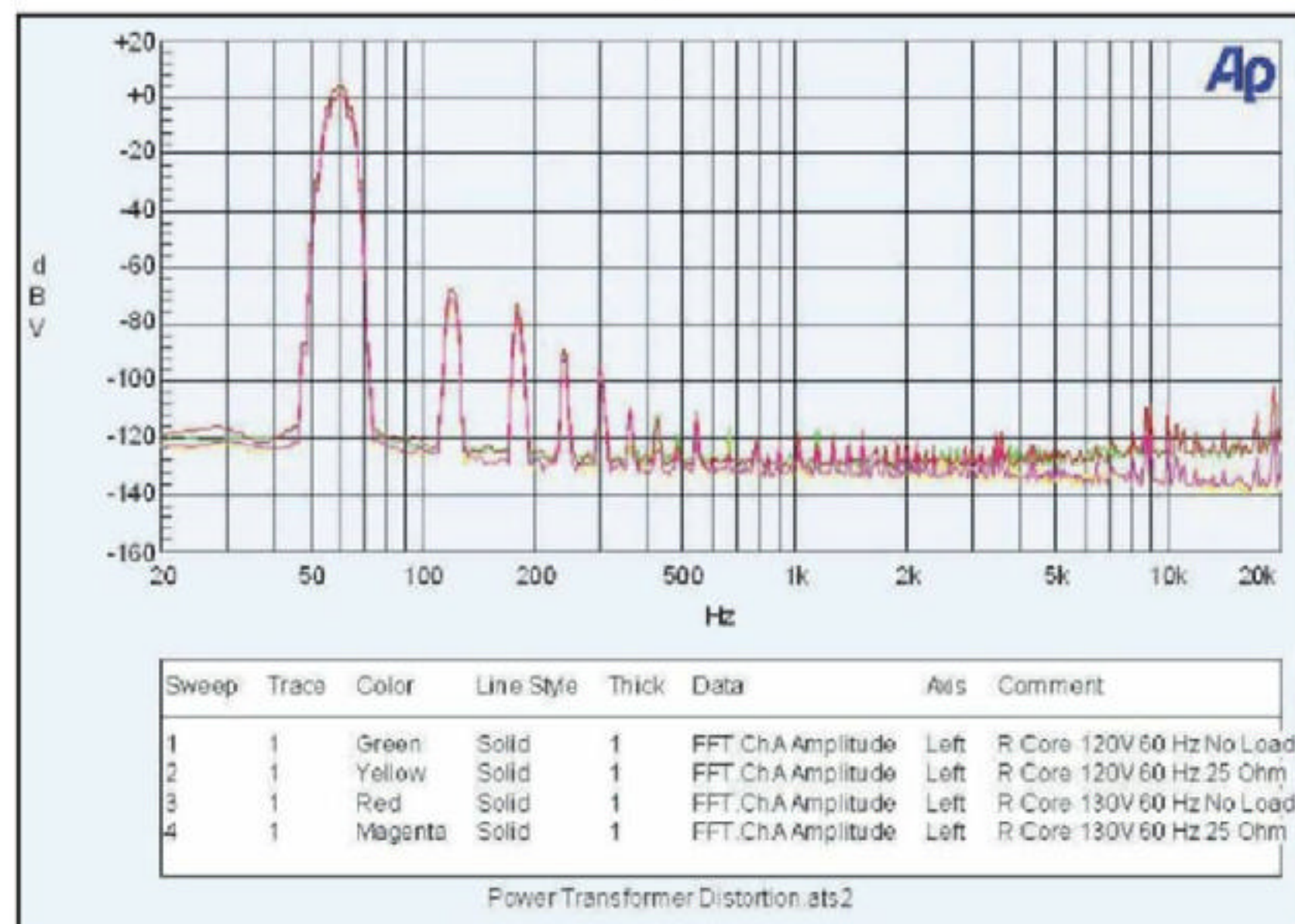


Figure 7: I also measured an R-core transformer rated for 220 V input at 120 V input.

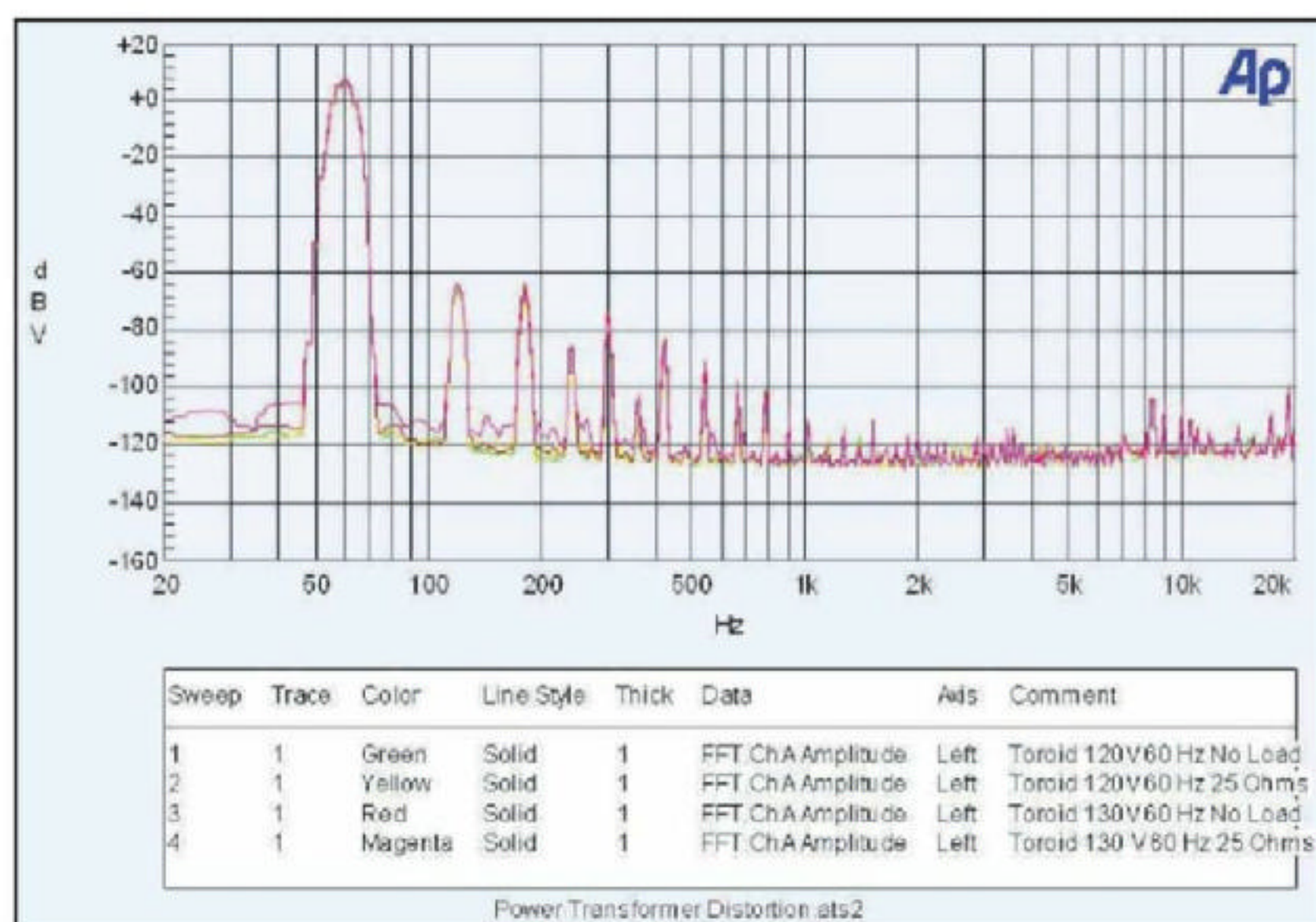


Figure 8: I conducted a third test on a Triad VPT24-1040 toroidal transformer, connected for a 120 V input.

An important feature is that as you step down the voltage, you step up the current available as power is conserved. So the 250 turns stepping down to 25 turns would in theory provide 10 times the current out as it draws. With losses, this can be closer to nine times the current due to the core losses and the resistance of the windings.

Now it is less expensive to use a bit less core and smaller gauge wire and just add a few turns to make a lower cost transformer. This is actually done for some large AC distribution transformers. These are often not just rated by efficiency but also by in-use temperature rise! (They also buzz a lot more!)

Measuring Transformers

When testing a transformer, it is often useful to measure the open circuit voltage, the short circuit current, the winding resistance, and the actual primary to secondary voltage ratios. Some models also short one coil and measure the inductance of the other for a value of modeled inductance. These are useful if you want to more accurately model how a linear power supply transformer will perform.

I connected my audio test set to a power amplifier and tested several different styles of transformers to see how much distortion they produced at rated voltage and frequency, and again with a slightly higher line voltage.

First up was a fairly standard E-I core 20 VA rated Signal Transformer model 14A-20-24 (see Figure 5). It is rated for international use and has extra insulation to meet many different codes. It was measured with a 1/12 voltage divider. The output voltage was 24 VAC RMS. The Fast Fourier Transform (FFT) display shows this as +6 dB re 1 V RMS.

Four plots were done: 120 V, 60 Hz open circuit, and then with a 25 Ω load that should fully load the transformer. These tests were repeated with a line voltage of 130 V.

A careful look will show the distortion components are a bit higher when there is no load. The distortion also increases when the line voltage goes up. An interesting issue to note is the high-frequency noise level drops by almost 20 dB when a load is present. That shows that the transformer has a rising impedance with frequency. Please note that the major distortion products are the odd harmonics.

Virtually all of the noise and distortion is being produced by the power transformer under test. This noise and the other noise from the rest of the power supply, not only goes into our load but also back into the AC line feeding the circuit.

Figure 6 shows the same transformer and test but done at 50 Hz. As you can see the distortion at low frequencies has risen by quite a bit. The 150 Hz harmonic is only down by 20 dB! Again distortion and high-frequency noise dropped under load. I also noted that at 130 VAC with the 25 Ω load, the transformer was warm to the touch. (It was being operated a bit above rating at 27 VA.)

I also measured a transformer rated for 220 V input at 120 V input (see Figure 7). This unit was an R-core unit that I purchased on eBay. An R-core is very similar to a toroid except the laminations are tapered to form a rounded core cross-section rather than the rectangular core of a toroid. Both are overall round. It has a lower secondary voltage (15 VAC RMS) when powered from a 120 VAC line than the first unit. The impressive difference is how much lower the distortion is now that the core is being run at half of the design value. Of course, this also demonstrates the distortion on the first test is in the transformer and not the test setup. As with the E-I core unit, there is high-frequency noise that is reduced by a load resistor.

I conducted a third test on a Triad VPT24-1040 toroidal transformer (see Figure 8). It was connected

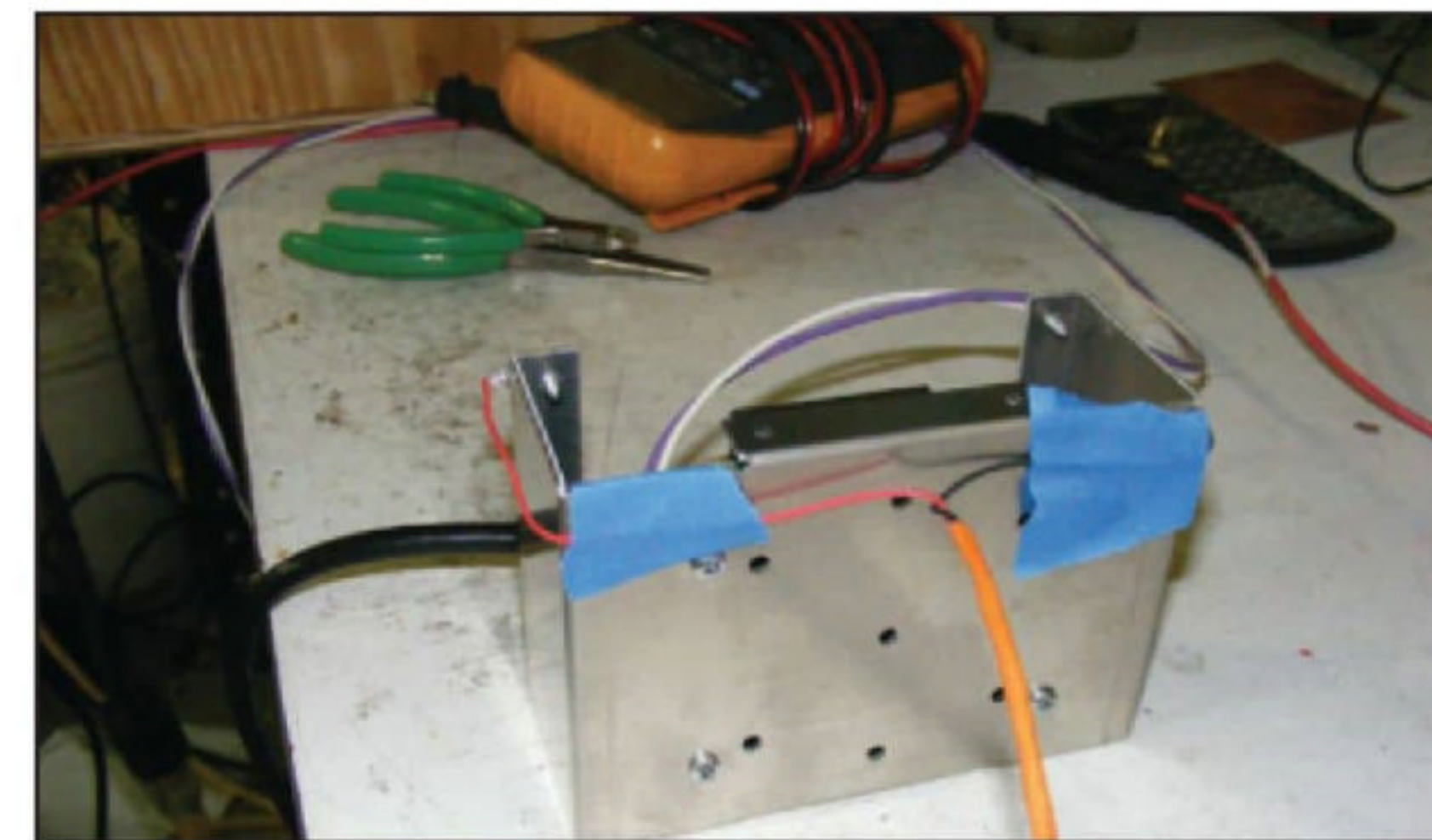


Photo 1: I made identical chassis for six different types of transformers and looked at the voltage difference across the chassis. One of the chassis was 0.062" thick aluminum (shown here). All the other chassis were the same thickness but made of mild steel.

About the Author

Ed Simon has a BSEE from Carnegie Mellon University. He was employed by Cameradio, a regional electronics parts distributor from 1973-1975. In 1975, Ed started Edward Simon & Co. He has published more than 40 articles and taught lectures for the NSCA, Carnegie Mellon University and the University of Washington among others. His company specializes in sound and communication systems, systems for public facilities, houses of worship, and allied engineering. The business has completed more than 400 church installations, and 150 other installations including shopping malls, teleconferencing facilities, performing arts centers, television studios, and radio stations.

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for 120 V input and performed quite well. As you can see, it has lower distortion. But a comparison with the R-core distortion is not quite fair, as here we are using the 120 VAC input not a 220 VAC input.

The noise behavior does not change very much with load. This is likely due to a wider transformer bandwidth. There also may be some advantages in the noise behavior due to the use of grain orient steel in the core. Also it would seem the core is

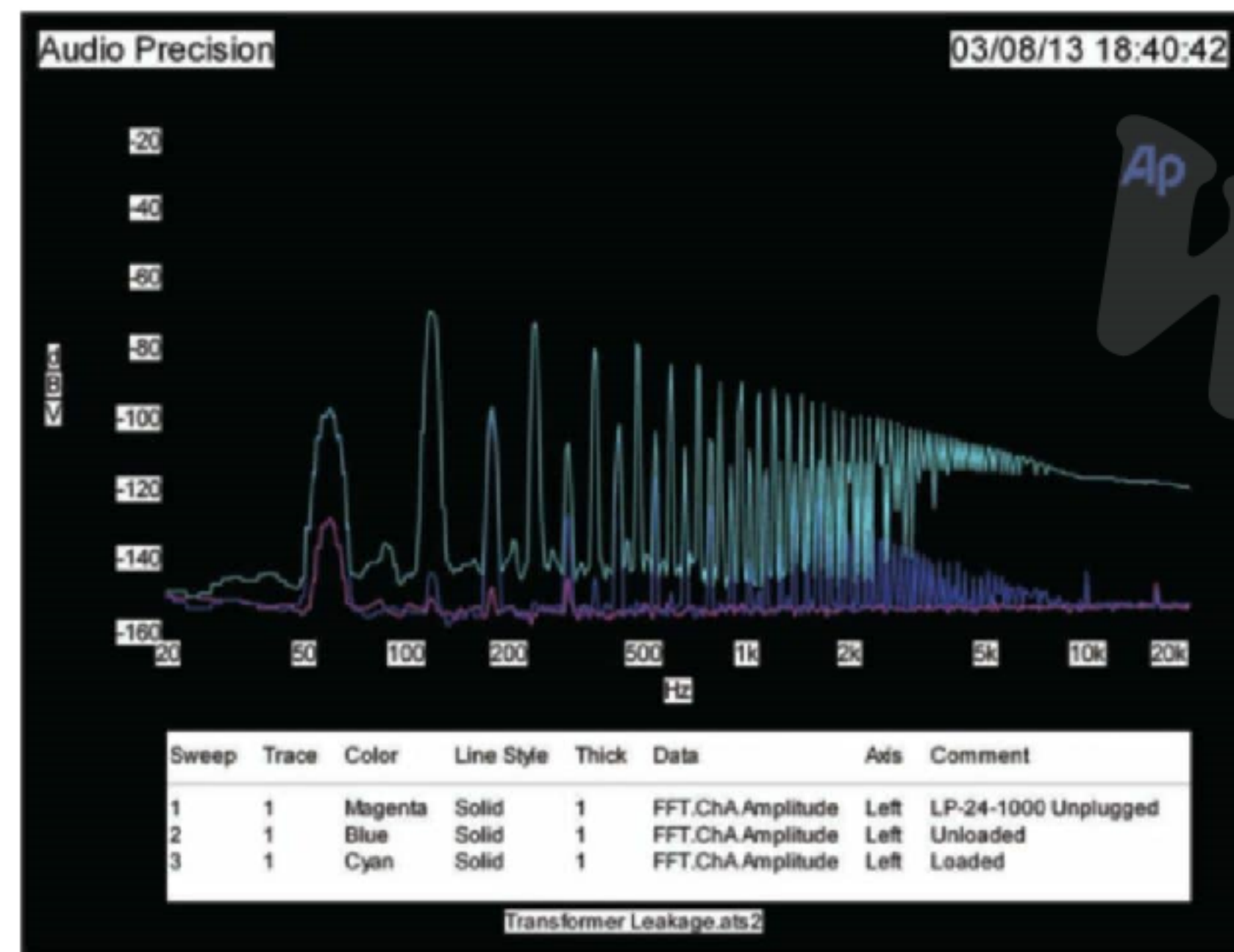


Figure 9: This is the result from a Signal Transformer LP-24-1000 transformer on a steel chassis.

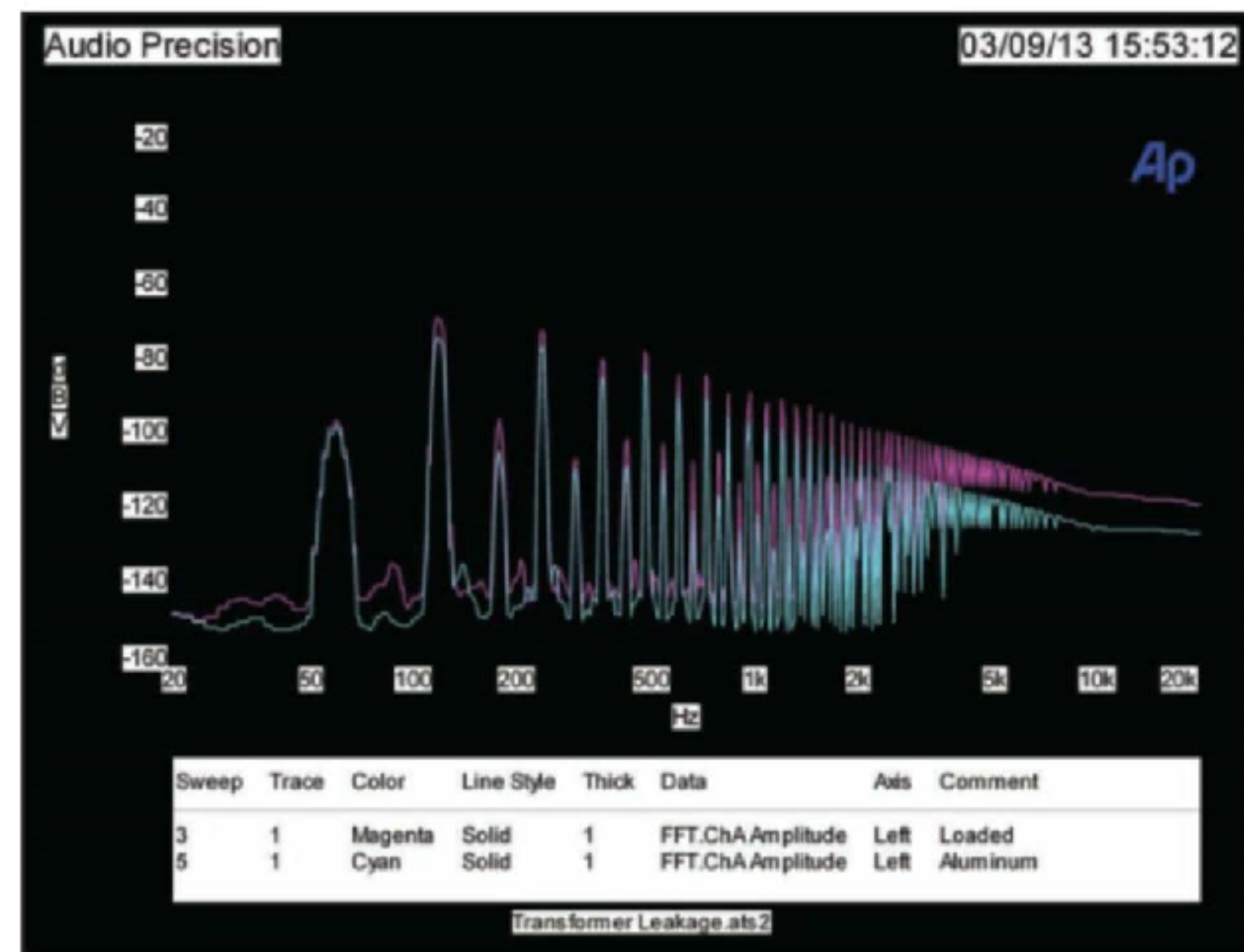


Figure 10: I received this result from the same Signal Transformer LP-24-1000 transformer on an aluminum chassis.

sized to be capable of handling the magnetic flux better than an E-I core unit. The better bandwidth may actually be a disadvantage as it not only lets noise in, it will also let noise out and back into the AC power line!

Before you select any transformer for a particular DC power supply, there are other issues to consider.

Transformer Leakage and Leakage Effects

One issue that will affect our surrounding circuitry is how the transformer flux leakage affects surrounding circuitry. I made identical chassis for six different types of transformers and looked at the voltage difference across the chassis. One of the chassis was 0.062" thick aluminum (see **Photo 1**). All the others were the same thickness but made of mild steel.

Now, the flux leakage out of a transformer will vary a bit with load. So a simple power supply circuit was attached. It consisted of a pair of rectifiers filtered by a 4700 μ F axial capacitor and loaded by a 50 Ω wire wound resistor. Looking at the voltage across the chassis shown in Photo 1 with an FFT analyzer seemed to give a reasonable comparison of the noise different types of transformers when used in power supplies would cause in surrounding circuitry. The lead dress was kept as short and consistent as possible.

Testing the Chassis

Figure 9 shows the result from a Signal Transformer LP-24-1000 transformer. The bottom trace is what is seen when the small power supply was not plugged into an AC power line. The middle trace is from just the transformer when the rectifier, the capacitor, and the load resistor test circuit were not connected. The highest trace shows the voltage with everything connected. Note that when the transformer is loaded with the rectifiers, the filter, and the resistor that large even-order harmonics appear. These are created from the load circuit. The even harmonics drop out on the unloaded curves and you can see the transformer saturation odd order ones at a lower level. The 0 db reference level is 1 V RMS. So voltages across the chassis as high as 1 mV were seen.

If this power supply introduced 1 mV near a phono preamplifier's inputs, the hum and buzz would make it useless. Even with a CD player producing the full rated 2 V output, the signal-to-noise ratio (SNR) would be limited to less than 70 dB, much less than a CD is capable of producing. So good design practice would be to keep this power

supply well away from any sensitive circuit.

Figure 10 shows is the same transformer on an aluminum chassis. What may surprise some readers is the induced noise is almost the same as the steel chassis. If one remembers, in most transformers it is a copper coil that converts the magnetic flux back into electrical current. That is why the aluminum picks up similar voltages to the steel chassis.

The most common type of power transformer is the E-I core. This transformer is frequently oriented so that the core is standing up from the chassis. As you can see it is almost 20 dB quieter at high frequencies than the flat pack "O" core transformer (see **Figure 11**).

Now if the core is mounted standard PC style, the orientation is the same as the flat pack "O" core transformer and we get a result that is between the vertical E-I core mounting and the "O" core (see **Figure 12**), which suggests the leakage is a bit directional.

Next I tried the toroid transformer. The results are shown in **Figure 13**. The toroid had much less leakage flux as was to be expected from the design.

The R-Core transformer was a bit noisier than the toroid (see **Figure 14**). Keep in mind that it also

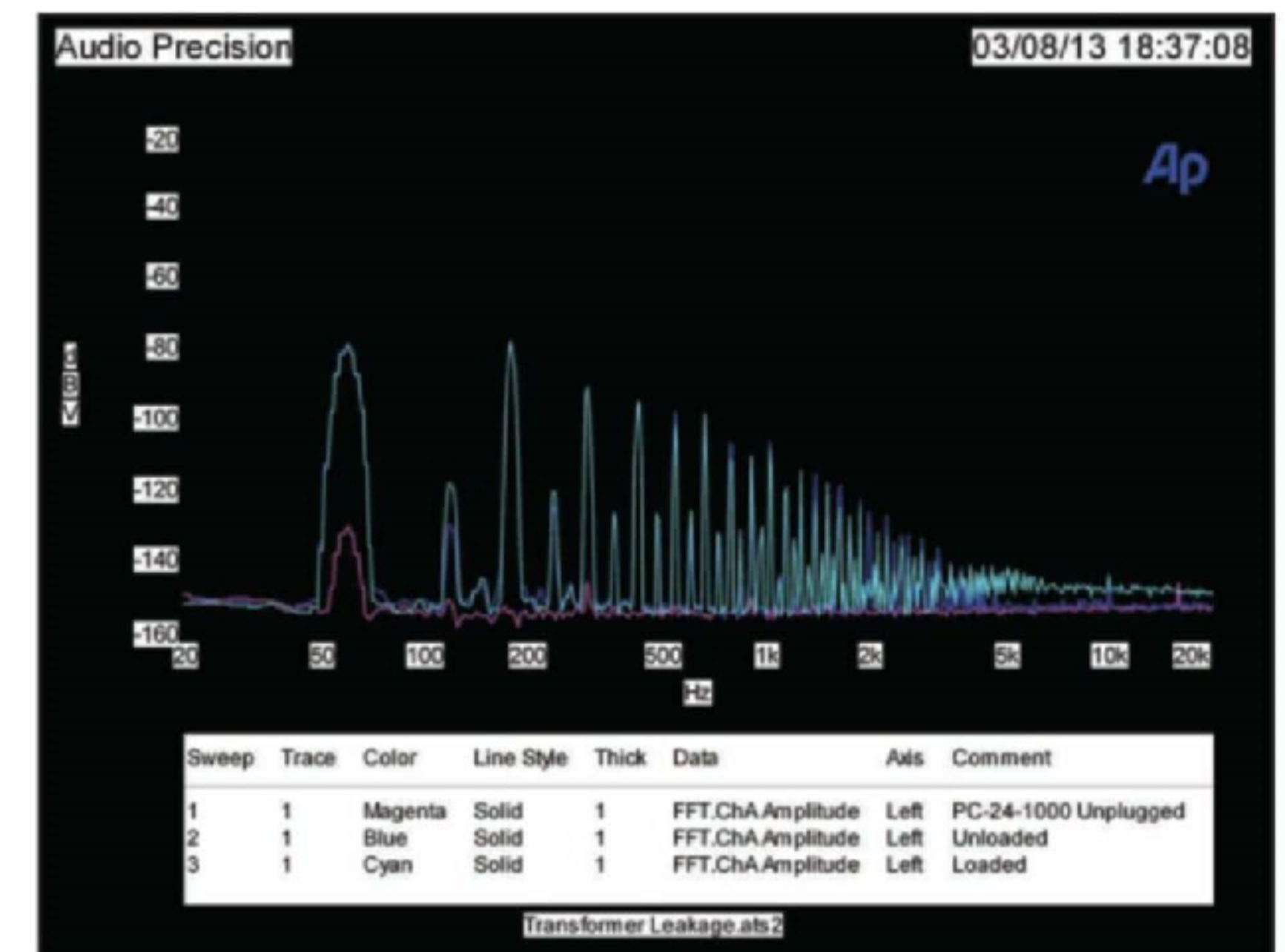


Figure 11: The transformer with the E-I core is almost 20 dB quieter at high frequencies than the flat pack "O" core transformer.

Reducing Flux Noise

Now, you may be wondering is there a way to reduce the flux noise into a chassis? One method is to mount the transformer to a "sled." The sled has at

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the corners four rubber grommets. Using a washer, a nut, and a bolt, the sled may be attached to the chassis to prevent electrical contact and reduce vibration. The sled's thickness will affect the noise reduction as will the type of metal and chassis to sled distance. Generally 0.062" mild steel will work well with most small power transformers. At higher frequencies, thinner metal would work as well.

So far we have seen how the transformers behave

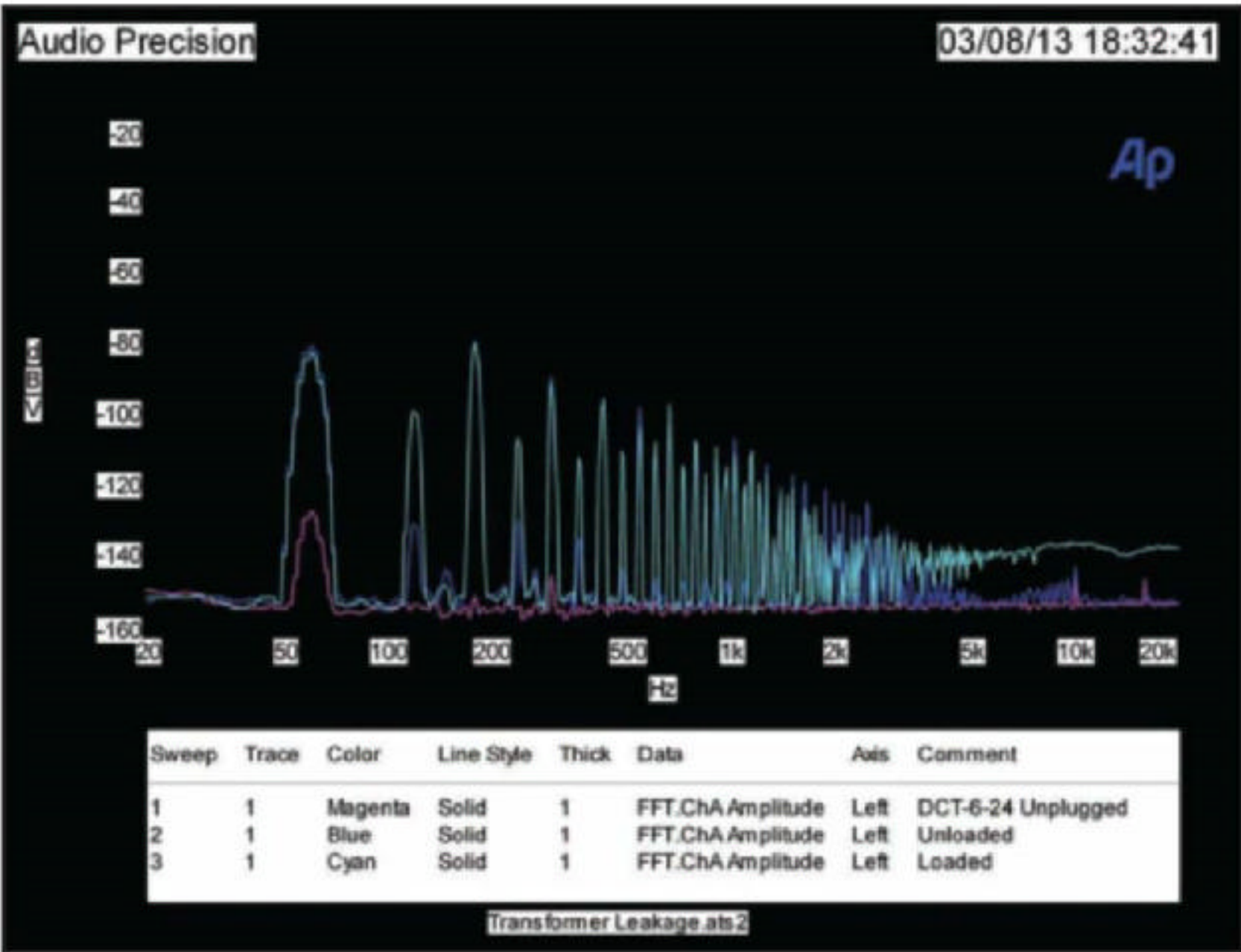


Figure 12: Now if the core is mounted standard PC style, the orientation is the same as the flat pack "O" core transformer and we get a result that is between the vertical E-I core mounting and the "O" core.

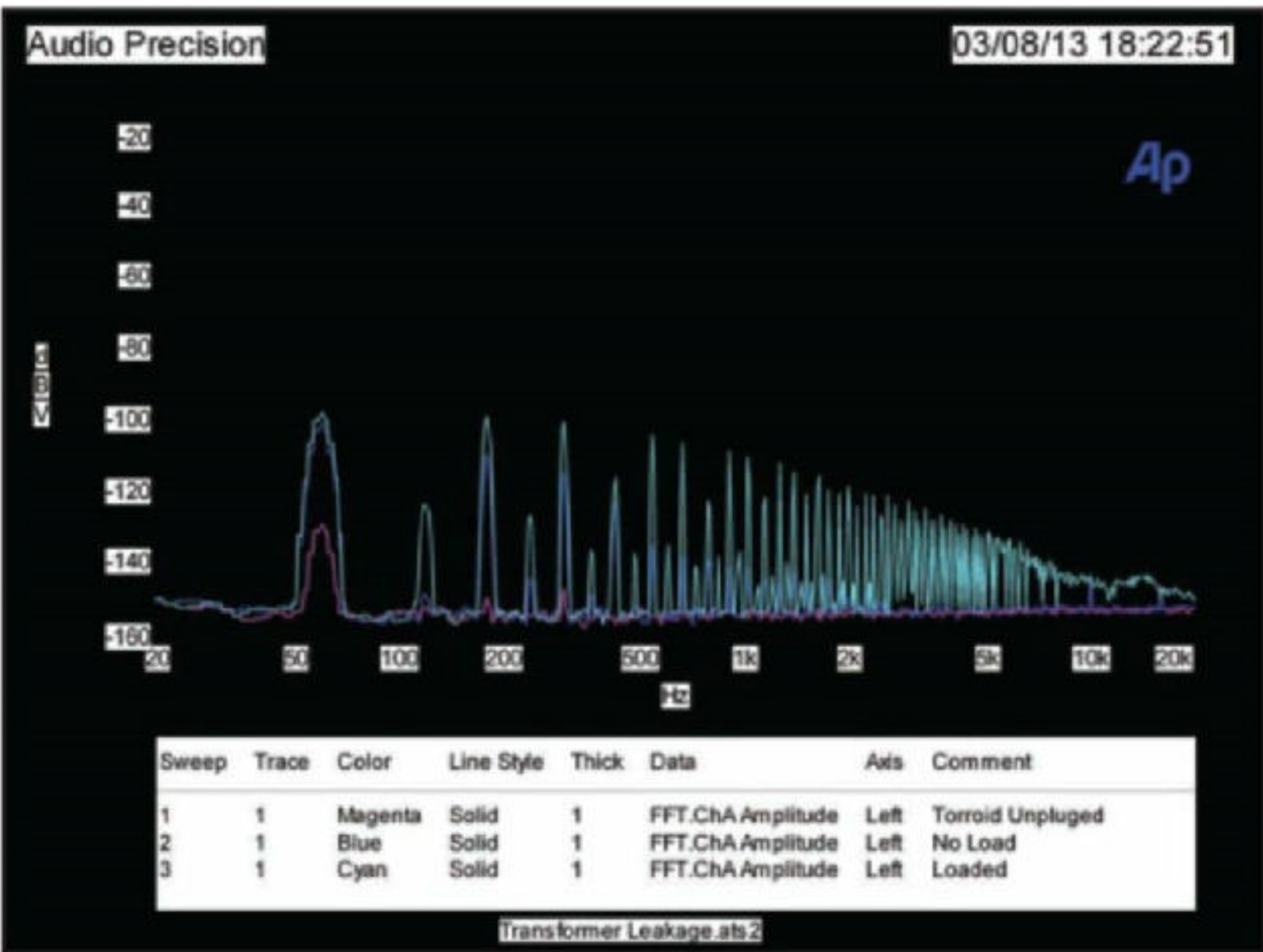


Figure 13: I obtained these results with a toroid transformer.

in creating and managing their own noise. So let us look a bit more at the rest of the power supply.

In the basic power supply circuit from Figure 1, using a transformer rated for 120 V primary and 6 V secondary, we will get just about 8 VDC at the output at full load.

An AC power line voltage is rated by the Root Mean Squared (RMS) method. One characteristic of an AC voltage is that it goes both positive and negative equally, so the average value is zero. At first, this confused folks who did not understand how any power could be delivered to a load when the voltage keeps reversing. (What they missed is that when the voltage is positive, so is the current so the power is a positive value. When the voltage is negative, the current is also negative, so again the power is a positive value!) So to get an RMS value, you square the voltage, which always yields a positive number, and you then take the square root of that and use the positive result.

With DC, Watt's law could be applied. Watt's law allowed that the Voltage times the Current would give the Power (in watts.) So if you had 10 VDC at 1 A, you would have 10 W. Now if you have an AC source that has a peak voltage of 10 V and a minimum voltage of -10 V and connected it to a resistor, you would find that the resistor did not get as hot as if you used 10 VDC.

The AC voltage from a power line is generated by a rotating machine. As a result, the output voltage is very close to a sine wave. It is not a perfect sine wave as there are gaps in the windings in order to allow the wire to actually fit into the rotating core! Then as the loads on the power lines are not just true resistors, there is distortion in the power distribution system, some of which is actually caused by the nonlinear flux in the distribution and load transformers.

For simplicity, we presume the input voltage is a sine wave. So to get the same energy or heating value as DC, the peak level of a sine wave must be 1.414 times the voltage of an equivalent DC voltage. That mathematically allows for the same area under a sine wave curve as a straight line (DC) would give. For a more complex AC voltage source, you would use the RMS formula to calculate the actual equivalent value.

When the voltage of the AC waveform peaks, the capacitor is usually not fully charged due to the effective resistance of the circuitry, so the current continues to flow for a bit longer into the capacitor. Also as the capacitor charges, the current required lessens so there is less loss from the internal losses resulting in the output voltage of the transformer rising a bit. As the input voltage continues to fall, at some point the transformer's output voltage into the diode is less than the voltage stored in the capacitor.

At this time, the diode stops conducting current. The circuit load will draw current from the capacitor. This lowers the charge stored and the voltage. It will continue to drop until the AC from the transformer not only goes back to positive, but actually has increased enough to be a bit larger than the voltage still stored on the capacitor. Then, the diode begins to conduct again and recharges the capacitor. There is a small voltage drop across the diode that varies with the magnitude of the current through the diode.

The amount of variation in voltage across the capacitor during the charge-discharge cycle is called ripple. The amount of time the capacitor is charging is often expressed in the amount of the cycle for which it charges in degrees.

The first problem with the basic circuit shown in Figure 1, is that the transformer is only delivering current for one half of the AC cycle. This means that the magnetic field in the core is not the same for both halves of the AC cycle. So the core slowly becomes a bit magnetized, as there is only time dependent fading of the flux without the current reversal effects, reducing the available capacity to handle the flux swings in one direction. This reduces

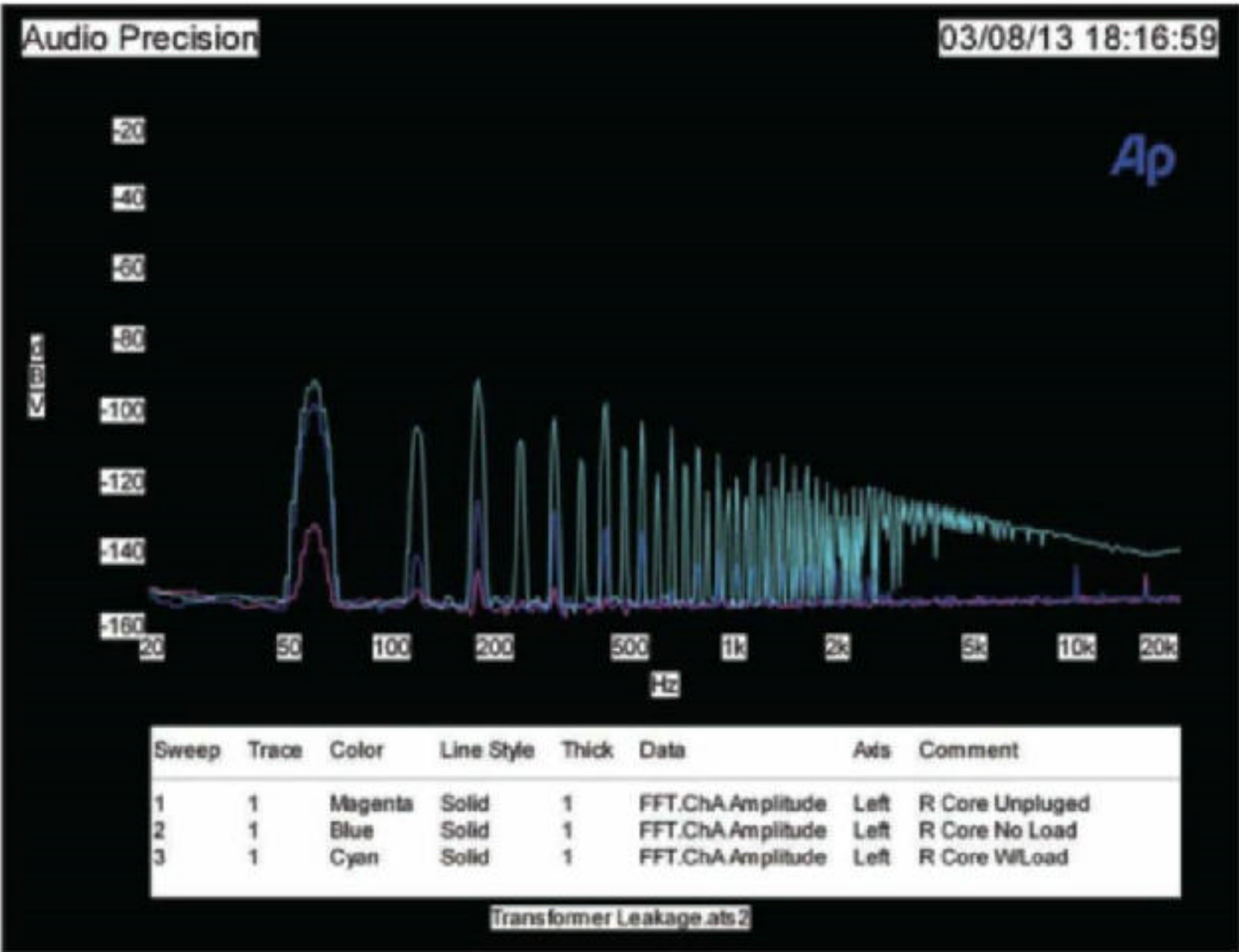


Figure 14: The R-Core transformer was a bit noisier than the toroid.

the magnetic flux saturation points. As a result, the transformer cannot handle as much current as it could with a more uniform load.

Next month, I will discuss AC power rectification and smooth in the third part of this six-part article series. ax

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