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ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

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Traditional design

TIMELESS SOUND

EXICON lateral MOSFETs are designed specifically for high fidelity integrated and power amplifiers. Launched in 1993 and improved in 2015, the current range is optimised to offer great advantages to the design of high-end linear amplifiers.

Simplicity

Simplicity in design minimizes the need for additional components in the audio path. The negative temperature coefficient means that Exicon MOSFETs are naturally immune to thermal runaway, and are well matched so there is no need for source resistors. Being voltage driven there is also no need for driver transistors, thus removing at least one amplification stage.

Extreme Linearity

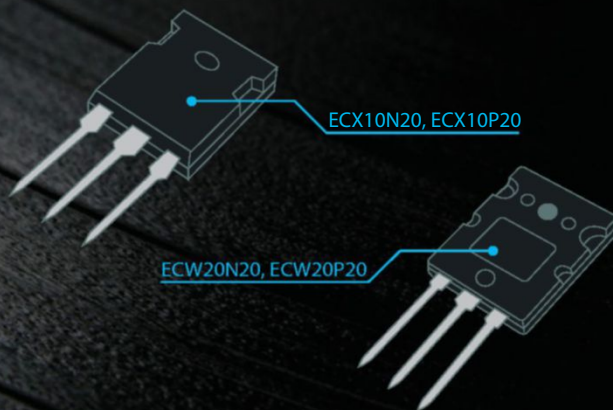
Extreme linearity gives naturally low distortion even before the application of negative feedback, which reduces it further. Lateral MOSFETs are easy to bias - the natural bias point is where the temperature coefficient is zero. For EXICON MOSFETs this is further into the Class A region to minimise crossover distortion.

Very High Speed

Very fast response is desirable to preserve the detail in the signal when it gets challenging, such as with an impulse response or percussion.

Rugged and Robust

EXICON MOSFETs are rugged and robust, and unlike bipolar transistors they are free from secondary break-down; this means the power rating does not derate with applied voltage. Devices are easily paralleled for very high-power applications.



EXICON

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Audio Sensing in 2021

The last 12 months have been challenging but they also paved the way for positive changes. In just one year, communications, online commerce, and the digital economy in general, evolved at an accelerated pace. The global pandemic made clear that investments in digital needed to be a priority in many areas that were still lagging behind, as well as reinforcing the digital infrastructure that now supports the entire economy. While we all agree that a bigger focus on privacy and security is needed, we have seen how beneficial it can be to have systems networked, connected, and more accessible than ever.

In education, the availability of communications together with connected resources is making access to knowledge, experimentation, and investigation more universal than ever. And that's also the foundation for all research and development activities—fostering better collaboration and the creation of dynamic communities.

For companies, these accelerated changes have also created a sense of priorities in the way they will be able to leverage technology to quickly adjust to foreseeable challenges and create more robust working methods that protect business and the workforce. This is true across all industries and applications.

While the pandemic is causing enormous economic impact in many industries, there was growth in specific technology segments, and the positive impact of some disruptions will be beneficial simply because it accelerated much-needed changes. Using technology and connected tools is an essential part of the recovery.

This month, *audioXpress* looked more carefully at the evolution of test and measurement solutions over the past year, and saw promising signs. Most test and measurement companies, including those focused on audio, quickly recognized the need to accelerate and adjust their existing solutions according to these pressing concepts. Implications of which enabled distributed testing, and introduced processes for remote collaboration—meaning connected measurement solutions that can be collaboratively explored, even by teams working remotely. In the same way as distributed computing, sharing large data sets and frameworks has long been acknowledged to foster progress, relying on distributed processes sharing dozens, hundreds, or many thousands of data acquisition points enables extraordinary achievements. Together with connecting users in communities, allowing remote access to measurement devices that otherwise may be sitting idle on a bench can make a huge difference to foster research or even simply for learning.

For test and measurement manufacturers, this could actually mean selling more instrumentation to be used as calibrated front ends, together with possible software-as-a-service models. It's just a new architecture with much greater scalability.

We analyzed that perspective for this edition's Market Update on Audio Test and Measurement where we naturally also discussed specific solutions used for measuring electronics and acoustics, as well as development and certification solutions.

And for this edition, David Logvin interviewed Matt Taylor, the president and founder of QuantAsylum, a fast growing test and measurement company based in the state of Washington, that adopted a unique modular product/solution approach that seems to be incredibly agile, adjusting well to current times. As Matt Taylor shares, his company has seen demand for its products doubling in the second quarter of 2020. And reliance on open-source and web- and community-based development has appealed to its growing user-base, while enabling the company to better understand real requirements and respond to actual demands.

These are interesting trends, even in those product categories that have showed greater resilience and expanded in these challenging times. I am certain these will resonate across the audio industry, particularly as we all face the need to embrace solid strategies for 2021 and beyond.

We need to consider how we could expand the benefits of streamlined and highly-efficient connected experiences for audio product development in general.

J. Martins
Editor-in-Chief

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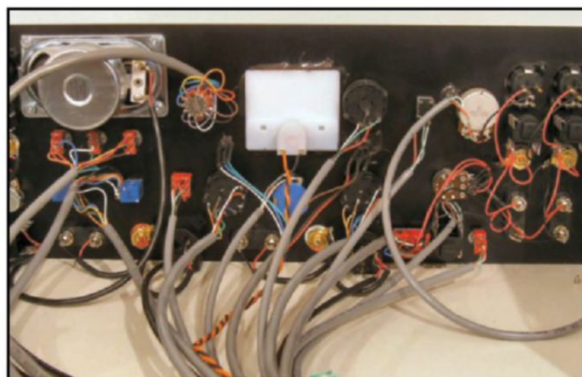
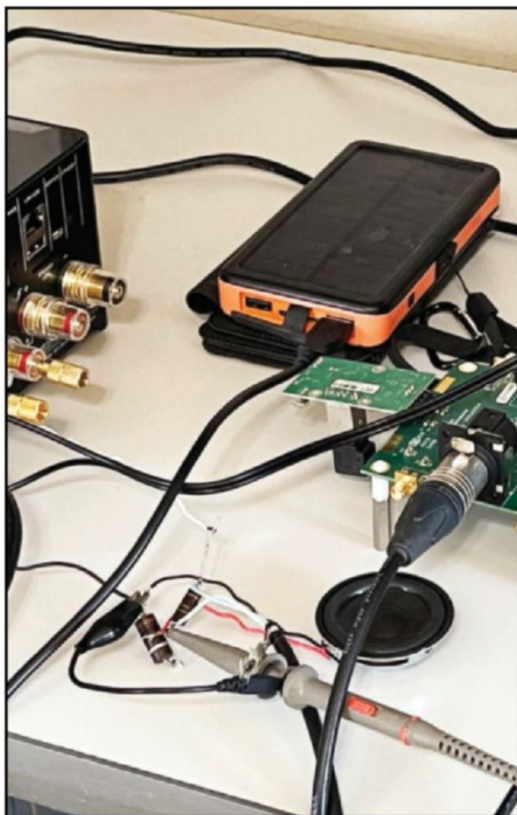


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By Mike Meechan

In Part 1, Mike Meechan considered the fundamental physics, which play a primary role in microphone pre-amplifier design. Now he discusses its practical role and how the mic pre-amp can retain the fidelity of the mic signal along its path from transducer to output.

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By Andrew C. Russell

In the second part of this article, the author details the Moving Magnet EQ stage, system gain amplifier and filters, along with the system's power supply unit for the X-Altra design. The article concludes with some measurements of the X-Altra preamplifier.

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By Ethan Winer

One of the problems when designing large audio devices, is the enormous amount of wiring that's needed. Wire is expensive, and so is hand soldering all those connections. In this article, Ethan Winer shows how electronic engineers can reduce the number of wires using rotary encoders, serial communications, and other solutions.

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AUDIO SCOPE

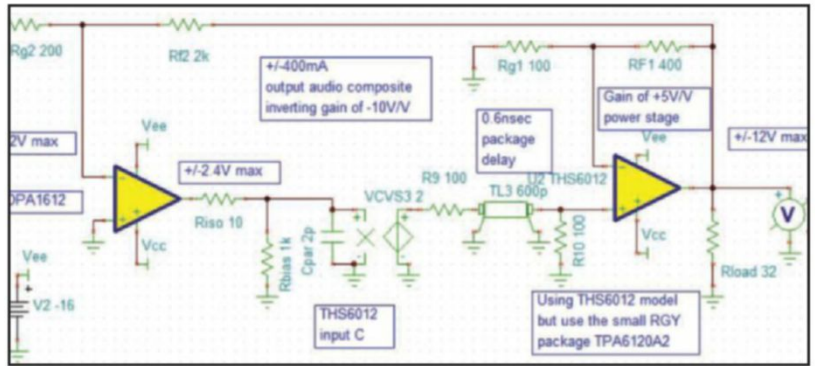
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Changing Processes for Remote Collaboration

By J. Martins

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A Real-World View of Test and Measurement

By David Logvin

Following the review of two of QuantAsylum’s products—featured in the December 2020 edition of *audioXpress*—a lot of engaging information about the company and its activities, inspired us to learn more. This follow-up article provides an overview of QuantAsylum, its history, philosophy, and product roadmap.

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The Research

By Stuart Yaniger

In 2015, Stuart Yaniger wrote a series of *audioXpress* articles discussing how to adapt home computers and sound cards for audio test and measurement. In this article, he experiments with 2021 technology to see what sort of measurement capability he can now achieve and updates his findings and recommendations for our readers.

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Distributed Testing

Changing Processes for Remote Collaboration

By
J. Martins
(Editor-in-Chief)

The latest global pandemic challenges have changed working environments and the need to leverage communications and connectivity in order to keep working. From research and development to production line quality control, test and measurement instruments and software are evolving from their traditional isolated “bench” and “island” approach, into fully distributed resources. This article reviews some of the key 2020 announcements on this front.

2020 was a difficult year for many industries and markets, but positive changes have also taken place. Research and development didn't stop and in fact, it accelerated. Some companies and organizations quickly adapted to new methods of working, and remote collaboration actually enabled a stronger focus on the work that needed to be done—or simply because the pandemic allowed for a bigger focus on priorities. This also accelerated the need to update test and measurement platforms.

These trends have affected even traditionally more conservative test and measurement technology applications in ways that we didn't foresee. Even when expensive and highly specialized instruments couldn't be updated and connected, organizations made an effort to create data sharing interfaces for heavier facilities.

As research and development continued and even partially accelerated in certain critical areas, many teams previously working from labs and fully equipped buildings are now mostly working from home and collaborating remotely. This, together with the evolution of cloud-collaboration tools, created new dynamics that are not likely to change.

While China and many Asian countries were able to keep manufacturing running and have mostly recovered from the pandemic, restrictions in travel and geo-political uncertainties regarding pandemic control have created the need for interfacing differently with the whole production and supply chain. These circumstances increase the need for more advanced planning, predictability, and resilience, and are fostering the adoption of more online management tools. The installation of remotely accessible equipment was a logical step, as distributed data resources and intelligence are gaining importance across processes as well.

At any stages where evaluation and assessment of parameters are critical and test and measurement is key, the updated working methods require instruments that are connected to the outside world and can even be, in some special circumstances and when advisable, remotely controlled. Distributed processes also pave the way to a better use of critical lab resources across all stages.

Accelerated Evolution

In 2020, people got more connected. In 2021, we will start to see equipment and solutions also being connected and resources collaboratively available.

In reality, it will take some time to change the way entire teams are able to leverage precious instrumentation in cutting-edge applications—but the more cutting-edge, the more connected those instruments probably are or can be, and the more they are able to be adapted to current methods of remote collaboration. On the other hand, tools that are used for education, training, research, and technical support are quickly being adapted and being transformed to be more accessible to anyone who needs them, when it needs them.

Some test and measurement companies are already showing us how these tools can change by unveiling new instruments that are directly connected to software and cloud services. This will not be an easy transition for all companies, some of which have long been conservative in their approach to hardware, trying to keep it as proprietary and closed as possible, and have limited software development resources. From dedicated self-contained lab instruments, the test and measurement companies that have embraced PC-based solutions have been highly successful in the past 20 years—and even the leading test and measurement companies had to embrace some kind of “open” approach in that regard.

The price to pay, unfortunately, is that those companies distribute low-quality software that is not updated as often as it should (many don't keep up with the pace of technology and OS updates). And even some of the most valuable test and measurement software that is used in production lines at critical stages has been pirated and distributed illegally at some point.

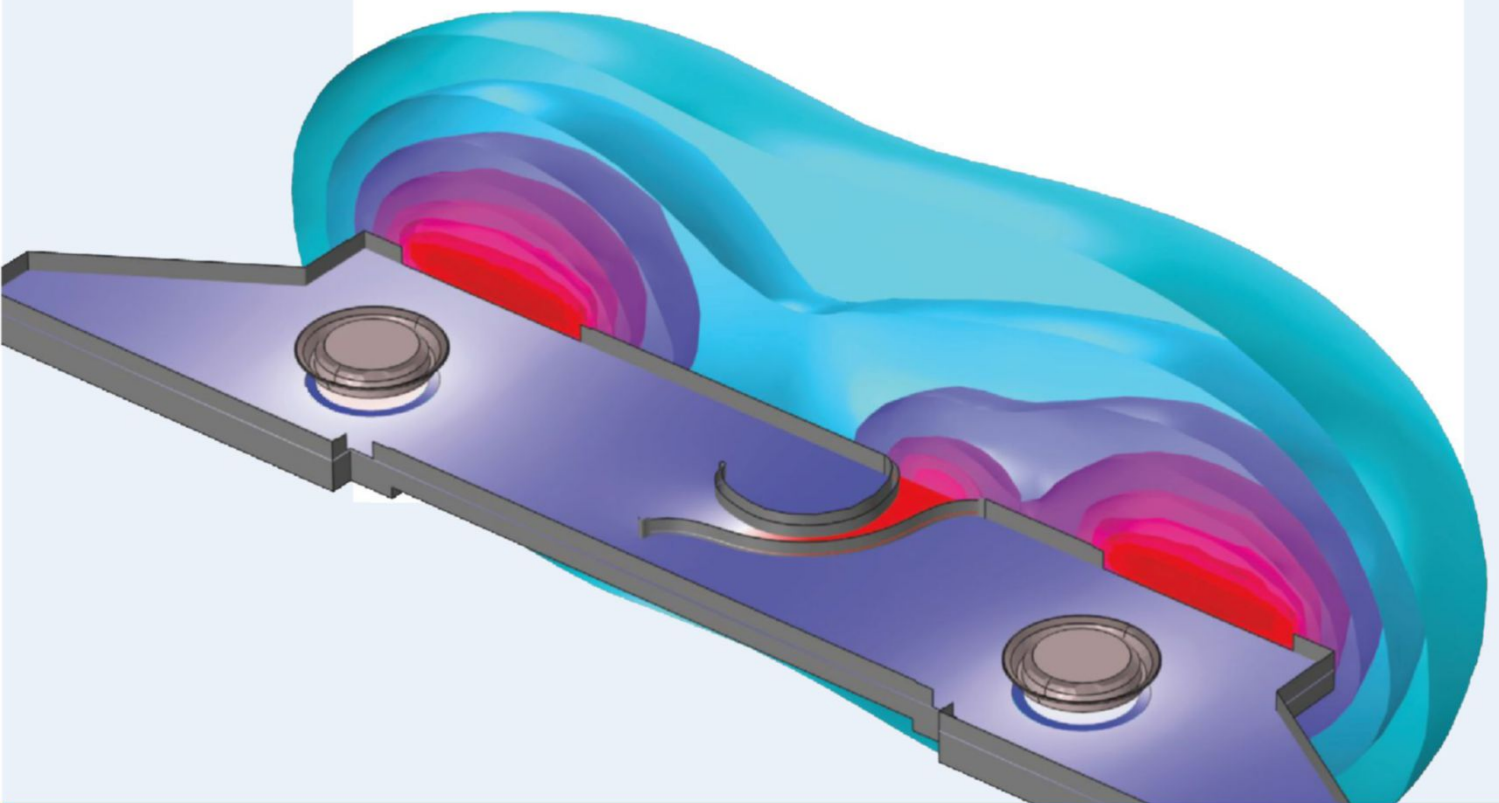
By transferring easily copied software licenses to distributed services and software-as-a-service models, continuously upgradable by the manufacturer, and available by subscription in flexible tiers, more organizations will be able to implement test and measurement tools and more users will learn to use them. From expensive and self-contained instruments, these tools can focus on more affordable

SIMULATION CASE STUDY

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hardware signal interfaces—where calibration is critical—and keep processing power available in a scalable model that will grow as needed—also accelerating innovation.

Another important factor to bear in mind is that a new generation of engineers is now learning highly sophisticated research methods by using mostly computer simulation, while modeling and simulation is increasingly a requirement for product development in any industry. Having test and measurement solutions that are connected, with software user interfaces and data models that are easily accessible will enable even more integration and validation of simulation tools. This, more than anything else, is the frontier that will change test and measurement forever.

It means the end of using old PCs with obsolete interface technologies and discontinued versions of Windows to run obsolete software that “it’s-good-enough-for-what-we-need,” just for cost reasons.

Pico Technology expanded its PicoScope 6000E Series oscilloscopes with the launch of three new four-channel models, each of which can be configured with 16 optional digital channels. These instruments with bandwidths of 300MHz or 500MHz are ideal for users who need professional-grade test equipment at their home office.

Resources

K. Alicke, X. Azcue, and E Barriball, “Supply-chain recovery in coronavirus times—plan for now and the future,” McKinsey & Co., March 2020, www.mckinsey.com/business-functions/operations/our-insights/supply-chain-recovery-in-coronavirus-times-plan-for-now-and-the-future

“Bluetooth Low Energy (LE) for wearables, peripherals and IoT,” Rohde & Schwarz, www.rohde-schwarz.com/ble-testing

“How manufacturers can power through disruption and gain competitive advantage,” PwC, www.pwc.com/us/en/library/covid-19/manufacturing-operations-strategy-coronavirus.html

National Instruments Corp., <https://www.ni.com>

Pico Technology, www.picotech.com

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SCAN FOR MORE

Tektronix

Connecting Measurement to the Cloud

In 2020, Tektronix quickly adapted its test and measurement solutions to the needs of users during the global pandemic, and made the results available before the end of the year. An important Tektronix development in 2020 was TekDrive, the first native oscilloscope-to-cloud software solution to facilitate global data collaboration directly on an oscilloscope, PC, phone, or tablet. After research concluded that 70% of oscilloscope users have the need to transfer data off scope—and that was before the pandemic—TekDrive was created to enable ease and accuracy in data accessibility and collaboration at a time when people have to work remotely. TekDrive provides the ability to instantaneously share and recall data directly on an oscilloscope, eliminating the need for cumbersome data-sharing practices.

The software provides visualization and analysis capabilities that support any modern browser, including options to view, zoom, pan, measure, decode, and analyze full test-and-measurement data on any device without the need for any additional software. Data updates are instantly saved in globally accessible shared folders in which owners can manage secure access and permissions at a granular level.

TekDrive was also built with secure vendor-agnostic REST APIs for scripting, automation, and analysis. Tektronix provides software development kits (SDKs) and examples in multiple languages, such as Python, Matlab and LabVIEW. The TekCloud Developer Program also provides a secure way for third-party developers to add native TekDrive capability to their devices, instruments, and software applications. With a quick integration, any vendor of hardware or software can unleash the ecosystem of TekCloud storage, streaming, visualization, and analysis into their products.



Keithley and Initial State, both Tektronix companies, collaborated to integrate Initial State's data streaming and visualization platform with the Keithley graphical DAQ6510 and DMM6500 remote monitoring capability, real-time interactive dashboards, triggering, and the ability to share data—all without the need for a PC or external software. Users can monitor and analyze an experiment remotely in any browser—including on mobile devices.

Shared Instruments

Tektronix also launched new features to its TekScope waveform analysis PC software, allowing remote users to view and analyze data from up to 32 channels from multiple scopes at the same time, in the same interface, accelerating analysis efficiency.

This was a platform that was already under development before the pandemic, but Tektronix decided to accelerate the development of TekScope as many engineers moved to off-scope environments.

The software enables users to work with actual data files, not with screenshots. Waveform files can be taken from any location and from a wide variety of scopes, including from other brands, to improve working efficiency, save time, and eliminate unnecessary trips to the lab. And with TekScope, users can free up their scope and allow others to use it while users analyze and debug data on their own PCs. Remote users can zoom in on waveforms, move cursors, measure, and more—all while sharing with colleagues.

Unlike apparently similar software, the Tektronix Multi-Scope Analysis feature enables users to remotely control the acquisition settings on all scopes simultaneously without the need to set up each scope individually. And users asked for a solution with high resolution that can catch very fast glitches across many channels at once. The Multi-Scope Analysis feature on TekScope solves that need.

TekScope is compatible with data taken from all Tektronix scopes so users can leverage Tektronix's user interface and analysis features for all existing scopes. The software is available with a subscription model, which allows tailoring prices according to its use and type of organization without restricting its features. Tektronix is currently working to expand this type of solution in order to better support engineering education.

www.tek.com



Tektronix's TekScope PC Software offers the flexibility to perform analysis tasks outside of the lab. Users can take data from their oscilloscopes and continue the work at any place while easily sharing analysis results with their peers.

New Features and Updates for Speaker Designers and Manufacturers

The specialists in electro-acoustical transducers and audio systems test equipment at Klippel GmbH continue to update their range of products—recently expanded with the Klippel Controlled Sound (KCS) Technology and adding onto the existing features for the Klippel Analyzer System. In 2020, Klippel was extremely busy updating its dB-Lab software and the QC Software Framework, continuously improving its measurement technology for loudspeakers and audio systems.

In order to provide dedicated tools for standard-compliant audio testing, Klippel released two new software modules: Multi-tone Measurement (MTON) for lab applications and QC Multi-tone Distortion (MTD) for efficient end-of-line testing.

The Multi-Tone Measurement (MTON) is a versatile tool for the R&D engineer that helps finding the operation limits of any audio product related to mechanical and thermal compression as well as multi-tone distortion headroom. Iterative, stepped measurements protected by clear thresholds automatically pinpoint the SPLmax according to IEC 60268-21 as well as continuous max SPL (ANSI/CEA-2010-B & 2034). The well-defined multi-tone stimulus can be exported to a WAV file, and it provides standard and custom spectral weighting functions to match the characteristics of the target program material for meaningful specs that meet the requirements of a product's target application.

In addition, the new Multi-Tone Distortion Task (MTD) is an add-on for Klippel's QC software framework that can be inserted as a test step (task) in any existing QC test. It ensures overall consistency of production electrically and acoustically.

Both modules (MTON and MTD) cover multi-tone stimulus, "fingerprint" testing of speakers and audio systems (acoustically and electrically), and stimulating harmonics and intermodulation distortion. All of these features come with a batch of new test

templates using the new MTON and other long-established measurement modules such as Transfer Function Measurement (TRF), 3D Distortion Measurement (DIS), and Near Field Scanning System (NFS) that focus on taking advantage of the new IEC standard and helping users perform hassle-free compliant measurements.

New dB-Lab and QC Tools

With the dB-Lab and QC software updates, Klippel introduced new tools for testing DSP-enhanced speakers, headphones and connected audio systems, first with a hardware add-on for the SCN Scanning Vibrometer, and second with an extension for automatic testing at multiple excitation levels using the TRF module.

The Klippel Analyzer System has received another minor software update for both R&D and QC applications. This free-of-charge update for users of dB-Lab version 210 or QC 6, respectively, provides two new relevant tools for output-based testing of contemporary DSP-enhanced speakers, headphones and other audio systems according to the IEC 60268-21 standard.

This includes a hardware add-on for the Klippel Scanning Vibrometer System (SCN), allowing new powerful possibilities for acoustic testing. As the German company explains, the hardware is extended with a microphone in addition to the existing laser sensor. Together with a baffle for measuring transducers up to 10" (30 cm) in diameter or compact (smart) speakers, it provides acoustic near-field scanning technology to the SCN.

This allows comprehensive calculations of near- and far-field radiation data, such as directional characteristics and sound power as well as sound pressure output at any point in a 3D half-space. This data is extrapolated from a double shell acoustic measurement and holographic processing for direct sound separation. This is the



The Klippel Analyzer System has received new software features for both R&D and QC applications—available for users of dB-Lab major version 210 or QC 6.



Klippel's Scanning Vibrometer System (SCN) Near Field add-on

same technology used before in the Klippel NFS, now available for the much smaller SCN hardware.

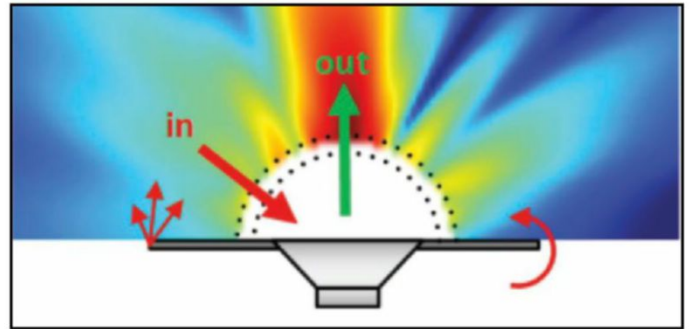
Setups using much larger and often still insufficient baffles can be replaced. The SCN equipped with the new SCN Near Field Add-On enables mechanical, electrical, and acoustical measurements in a very compact space, not requiring an anechoic room—a normal reverberant room such as a (home) office is sufficient. And the automated axis control ensures repeatable, precise, and fast positioning of microphone and laser sensors.

As Klippel anticipates, this add-on will be available in early 2021. However, beta testers are welcome to get their hands on the new test equipment and should contact Klippel if interested.

More Modules and Extensions

The Transfer Function Measurement (TRF) module also got an extension for automatic testing at multiple excitation levels. The new TRF Voltage Stepping (STEP) module (Pre-Release) works with the TRF operation, simplifying the test procedure and the graphical result display. Limits can be applied to help easily rate U_{max} and SPL_{max} in accordance with IEC 60268-21.

Furthermore, the STEP module can be used for detailed



Holographic processing and Direct Sound Separation. This is a directivity plot for an infinite baffle compensating finite dimension and edge effects.

analysis. For additional information, the measurement data can be automatically forwarded to the Time-Frequency-Analysis (TFA) module or, for perceptual analysis, to the Audio Player Module (PLAY).

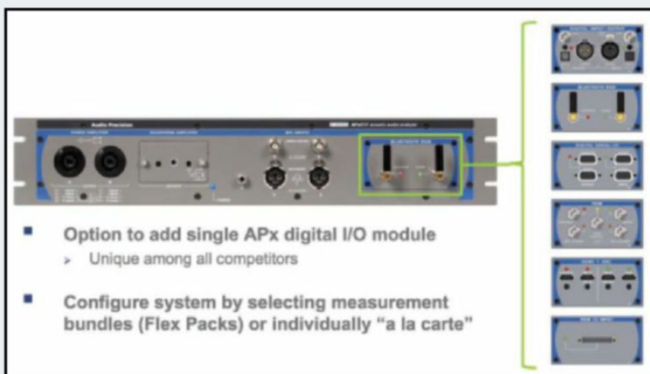
Both extensions complement Klippel's transducer and audio system diagnostic tools and are fully compliant with IEC 60268-21.

www.klippel.de

Audio Precision

New APx517B Acoustic Analyzer

Audio Precision recently announced its new APx517B acoustic analyzer, specifically designed, configured, and built for the production-line testing of speakers, microphones, headphones, headsets, and the wide range of consumer electronics incorporating speakers and microphones. With the introduction of the APx517B acoustic analyzer, manufacturers can deploy an integrated acoustic test system that brings the renowned quality, reliability, and robustness of Audio Precision's lab-oriented analyzers to their production lines.



As an integrated system—combining an audio analyzer, a power amplifier, a headphone amplifier, and microphone power supplies—with multiple optional modules, the new APx517B Acoustic Analyzer offers drastically reduced setup time and renowned AP reliability and quality to acoustic product manufacturers at a production test price.

Audio Precision's reputation for audio test and measurement, particularly for R&D duties, is undisputed and is frequently the choice for reference measurements at any stage, from scientific research to manufacturers' labs and factories. For less critical applications—such as testing low-end consumer electronics, portable use, and occasional audio measurement tasks—Audio Precision recently launched its APx500 Flex, combining the capabilities and flexibility of APx software with ASIO-capable third-party audio interfaces to create a more cost-effective solution. The APx500 Flex solution 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 the organization.

Still, in measurement scenarios where hardware performance requirements and calibration are important such as for production line test of speakers, headphones, and microphones, another type of approach was required. While the APx500 Flex project was extremely successful for Audio Precision, the Oregon-based company was already working on an optimized solution that would drastically reduce setup time, while offering the renowned AP reliability and quality—at a production test price. The result is now market-ready in the form of the new APx517B Acoustic Analyzer, engineered to the same standard as all other APx analyzers.

Measure and Repeat

On a production line, any time spent setting up a new test

system, calibrating a system for a new shift, or reconfiguring a system for a change in device under test (DUT), is effectively a “line down” situation. As an integrated system combining an audio analyzer (signal generation and acquisition), a power amplifier, a headphone amplifier, and a microphone power supply, along with any required digital interface such as Bluetooth, the APx517B has the power to drastically reduce setup and reconfiguration time by eliminating multiple individual components (from a variety of vendors) and their associated cabling. Additionally, and unlike those individual components, the APx517B is uniquely delivered with an accredited ISO/IEC 17025 calibration.

In addition to its ability to significantly impact the time component, the APx517B offers AP performance at a production test price. The base model, presented at a competitive MSRP of \$6,000 US, delivers measurement capability for the functional test of headphones, speakers, drivers, and microphones. Configuring the system for specific DUT, interface, or measurement requirements is equally affordable for manufacturers.

In its base configuration, the APx517B is a ready-to-go system for measuring analog speakers, microphones, headphones, and headsets. For digital devices, the APx517B has a module slot for the addition of a single APx digital interface module (e.g., Bluetooth, PDM or HDMI). On the software side, a standard system provides a core set of measurements and functionality to allow out-of-the-box, fundamental tests of acoustic devices: File Analysis, Sequence Mode, Input Signal Monitors (including FFT Monitor), Level & Gain, THD+N, Loudspeaker Product Test (including Rub & Buzz), Stepped Frequency Sweep, Pass/Fail, and Signal Acquisition measurement.

Since Audio Precision is part of Battery Ventures, the company that also acquired Denmark-based GRAS Sound & Vibration, the solution will also be available as an integrated acoustic analysis system with measurement microphones and other products from the prestigious manufacturer.

As the company did with the APx500 Flex system, a trio of Flex Packs is available, each offering different groups of more advanced measurements. Perceptual audio measurements—including ABC-MRT and POLQA—can likewise be used with Flex. Finally, for the user who needs the base configuration plus just one or two more

measurements, an à la carte menu of measurements is likewise available.

Software Updates

Audio Precision’s APx500 software paired with an ASIO-capable audio interface, remains an ideal test solution for very cost-sensitive analog-only test applications. For mid-range analog test and digital test applications (e.g., Bluetooth headphones and smart speakers), the APx517B acoustic analyzer, with robust, integrated hardware, digital interface options, and accredited calibration, is the optimal choice.

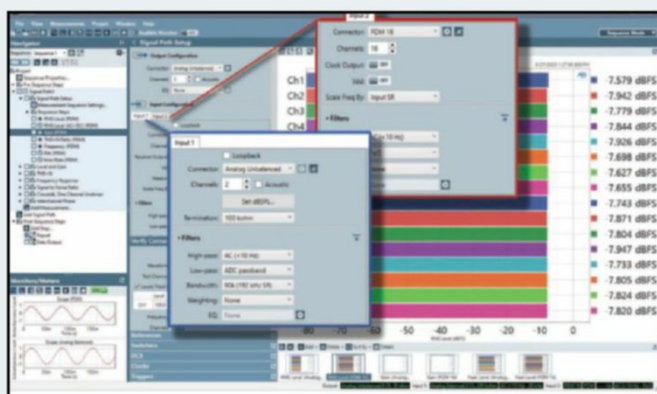
As with all other new APx analyzers, the APx517B ships with version 6.0.1 software and one year of software maintenance, effectively licensing that instrument for APx version 7.0 when it is released, as well as any minor releases that occur between versions 6.0 and 7.0.

The recently launched APx500 audio measurement software, version 6.0 supports multiple simultaneous input types, enabling the concurrent measurement of up to 16 channels of digital and eight channels of analog audio data. This new multi-input capability, paired with the highest channel count and widest range of digital I/O available, provides APx users critical cross-domain insight and the opportunity to reduce testing time.

Multiple simultaneous input types, or multi-input, is the ability to activate a second input type on any APx modular audio analyzer, or APx515 analyzer, and make measurements simultaneously on analog and digital signals. With two input types enabled (e.g., analog for one or more reference microphones and PDM 16 for an array of MEMS microphones), there will be measurement results for every enabled input channel. In addition, the sensitivity or gain function of digital microphones and other converters can be directly found via a new derived result, Sensitivity, which displays the RMS Level relationship between the analog and digital inputs.

The multi-input feature is available for use in both Sequence Mode and Bench Mode and supports all the essential measurements used for the analysis of microphones and microphone arrays. This includes Transfer Function, a dual-channel FFT analysis function added in the previous release of software. With multi-input and Transfer Function, the complex, magnitude and phase, relationship between multiple analog or digital microphones or other cross-domain devices can be found.

www.gras.dk | www.ap.com



Audio Precision APx500 software release V6.0 with multiple simultaneous input types.



The APx517B Acoustic Analyzer paired with a GRAS 45CA Headphone Test Fixture to demonstrate headphone performance measurement.

Ellisys

Addressing Bluetooth Low Energy Isochronous Channels

The 2020 release of the Bluetooth 5.2 specification introduced several major features for Bluetooth, including provisions for transporting audio over Bluetooth Low Energy (via the new LE Isochronous Physical Channel). Historically, Bluetooth audio has been delivered exclusively over “Classic” Bluetooth (BR/EDR), but with this latest update, LE Audio will soon appear in new products.

Major use-cases enabled by LE Audio include Multi-Stream Audio (connection-oriented) and Broadcast Audio for Audio Sharing (connectionless). Support is also available for assistive hearing devices. LE Audio will use Low Complexity Communications Codec (LC3)—a low-power, high-quality audio codec, ideal for achieving an optimum balance between power and quality.

The LE Isochronous Physical Channel underpins each of the new audio features. The physical channel (there are several with Bluetooth) defines low-level functionalities, such as how many channels are used, critical timings (audio anchor points), transports (e.g., connection-oriented, broadcast), PHYs, and so forth.

New terms (and a corresponding abundance of protocols and timing requirements) are introduced to define the types of traffic carried over the LE Isochronous Channel. Key terms to know include Connected Isochronous Stream (CIS) and Broadcast Isochronous Stream (BIS). A CIS is a point-to-point logical transport that uses acknowledgements to drive retransmissions. A BIS is a

connectionless logical transport; transmissions are unacknowledged. In a CIS, data can be transferred in one or both directions. In a BIS, data moves from the Broadcaster to one or more Synchronized Receivers.

A Connected Isochronous Group (CIG) consists of either a single CIS or multiple CISes (up to 31) that have similar timing characteristics. A Broadcast Isochronous Group (BIG) may contain one or more BISes.

Multi-Stream Audio involves the use of a source device that transmits independent, highly synchronized audio streams to multiple sink devices, where the audio can be rendered at the same time across the sink devices. A simple example of this is a smartphone sending two audio streams to True Wireless Stereo (TWS) earbuds. Multi-Stream audio uses the CIS and CIG approach.

Audio Sharing uses the Broadcast approach. This feature enables a single audio source (Broadcaster) to broadcast one or more audio streams to Synchronized Receivers (sinks). There is no limit on the number of receivers in this 1:M (one to many) topology. Audio Sharing use cases can be personal (I share my audio with you), or location-based (public venues, such as airport TVs that are otherwise necessarily silent or multi-language/translation services at a conference).

Information carried over an LE Isochronous link may or may not be encrypted or authenticated. Any CIS must have an association with a standard Asynchronous Connection Logical transport and largely follows existing security requirements, with some challenging differences. For BISes, a new security mode is introduced, referred to as LE Security Mode 3. This mode uses a Broadcast Code (known to users as a Bluetooth Privacy Code) as an input for authentication and encryption.

Characterizing and debugging Bluetooth LE Isochronous traffic is complex. Precision timing parameters and intricate security requirements present challenges to developers and test equipment manufacturers such as Ellisys. Ellisys engineers have been at the forefront of the development of LE Audio, having provided radio manufacturers and other early developers with pre-spec test and analysis capabilities that have helped to shape the final version of the core spec and test specs.

Ellisys qualification tools and analyzer products are designed to be reconfigured in the field to support changes to the specification, which enables early support for emerging specifications. This is key to reducing development iterations, reducing project costs, and improving overall quality.



The Ellisys Bluetooth Vanguard is an all-in-one advanced wireless protocol analysis system, fully hardware-integrated, time-synchronized, and able to support the increasingly complex tasks of Bluetooth developers. Capturing wireless traffic is a very important aspect of Bluetooth debugging, but other information is equally important. The Ellisys Bluetooth Vanguard supports one-click concurrent and tightly synchronized capture of: Bluetooth Classic BR/EDR, Bluetooth Low (BLE) Energy, Wi-Fi, WPAN (IEEE 802.15.4), raw RF spectrum, HCI, logic/GPIO, generic I²C, UART, SWD, and SPI, Audio I²S, and WCI-2.

www.ellisys.com

www.ellisys.com/products/bv1/index.php

CLIO and CLIO Pocket Analyzer Software Updates

As many of Audiomatica's users were happy to discover, the Italian audio test and measurement company had a busy year and released software updates to its CLIO family of products.

CLIO 12.5 software, supporting and driving its existing FW-02 audio interface and ready to accommodate future developments, received more Windows 10 (supporting Secure Boot) and macOS operating system support, and now offers 64-bit support.

The CLIO System with CLIO 12.5 software is a complete audio and electro-acoustic measurement instrument for labs, on-field and production-QC applications, running the CLIO 12.5 STND Standard Software or CLIO 12.5 QC Quality Control (which has also extended features not only related to Quality Control). These two parts (hardware plus software) cannot function separately. Each CLIO System is composed by the CLIO software coupled to the proprietary Audiomatica FW-02 USB Interface (24-bit at 192 kHz).

The new CLIO 12.5 software version added new functions, such as a new stereo input and output control for the Multi-Meter, Peak Meter to monitor input signal, 0.1dB output level control, output DC voltage control, new STIPA direct measurement features for rooms and auditoria characterization, as well as new Quality Control graphics management features. The new stereo multi-meter panel provides all standard measurements, from SPL (dB SPL, dB Pa, dBA, dBC) and Voltage (V, dBV, dBu, dBr), all the way to interchannel phase.

The update provided a new Graph Report Window and Windows layout functions. These new graphical functions enable users to display not only final test results, but all the associated curves in different graphs inside a dedicated graph report window. Each graph can be separately titled and formatted and, inside the same graphical container, it is also possible to

accumulate more graphs coming from different measurements. Windows layout functions allow users to define desired QC desktop settings and store them for later manual or automatic recall during testing. Finally, a new WAV2SIN processing function permits asynchronous QC test recording on multiple wave files and then analyzing them using ADB or ALSA devices.

CLIO Pocket 2.1

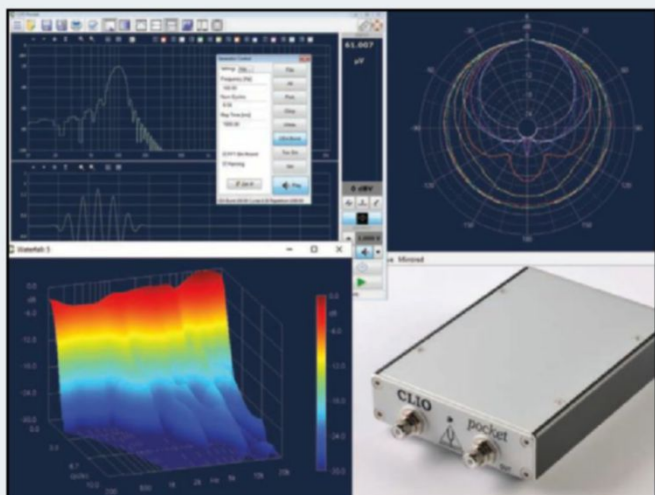
Audiomatica has also released the CLIO Pocket 2.1 software. This was mainly a bug fix release that improves stability and introduces minor updates, but at least an updated version of the Clio Pocket Manual is now available. The update to version 2.1 is free for all users already using CLIO Pocket 2 through the company's CLIO Software Upgrade Service.

Apple's introduction of macOS 10.15 Catalina, which has ended support for 32-bit applications, has sparked the need to update the CLIO Pocket software. During the first half of 2020 Audiomatica released a 64-bit beta version of the CLIO Pocket 2.12 x64 software, which runs on macOS 10.15 Catalina.

The more recent introduction of macOS 11.0 Big Sur and at the same time Apple Silicon M1-based hardware again created a new situation that will require the adaptation of the entire CLIO Pocket development process. Audiomatica is busy working on this latest development and reported that it successfully tested CLIO Pocket 2.12 x64 beta under macOS 11.0 Big Sur on Mac computers with Intel silicon.

However, tests on Mac computers with Apple M1 silicon will require further work, as with all manufacturers. Users interested to use CLIO Pocket with macOS 11.0 Big Sur with Intel silicon, can join the company's Beta Testing program.

www.audiomatica.com



Keysight

Complete USB4 Test Solutions for Standard Compliance

With the IT, mobile, and consumer electronics industries focused on the transition toward a common interface and power solutions based on the new universal USB Type-C connector, gradually developers are starting to update their designs to the latest USB4 specification. To support those efforts, Keysight Technologies announced new USB4 solutions that maximize the performance of USB designs and ensure compliance with the USB standard to deliver accuracy and high signal fidelity.

Keysight is active in next-generation USB4 specifications implemented by the USB Implementers Forum (USB-IF). A wide range of products, including smartphones, laptops, desktop computers, tablets and digital cameras, rely on next-generation high-speed digital standards. USB4, a plug and play interface for inter-digital communication, is based on the Thunderbolt protocol specification and

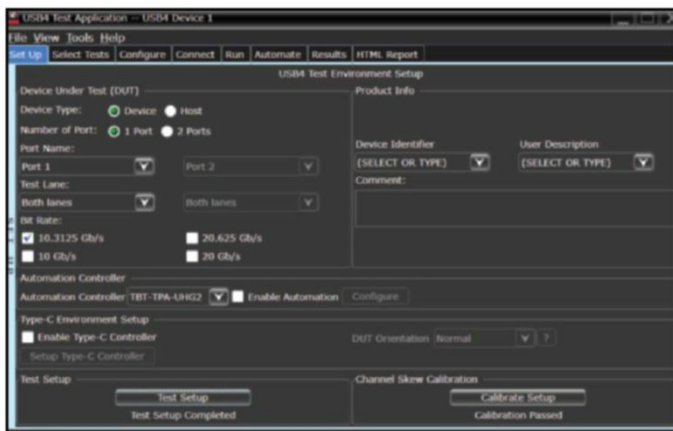
offers significant advantages in bandwidth, power and data delivery.

To enable manufacturers and brands to design debug and test, Keysight offers USB4 solutions, including test software and hardware interfaces to debug and decode signals and protocols. The Keysight D9040USBC USB4 Transmitter Test (Tx) Software is designed to rapidly test, debug and characterize complex USB4 designs, while the N5991U40A USB4 Receiver Compliance Test Software enables automation of complex stress signal calibration and receiver test procedures to ensure consistency and repeatability.

On the hardware front, Keysight offers the N7019A USB Type-C Active Link Fixture, which provides access to all Type-C signals in a live link to debug or decode acquired signals, complemented with the D9010USBP USB4 Protocol Trigger and Decode device, which can be used in conjunction with the Keysight's N7019A USB Type-C Active Link Fixture to add configurable protocol-level searches and software-based triggering to debug and decode USB4 low-speed and high-speed traffic.

In accordance with the USB standard and to ensure the interoperability of USB4 devices, Keysight also offers a comprehensive portfolio of high-speed digital instrumentation including its UXR real-time oscilloscope, which provides accuracy in waveform representation across a wide frequency range, a range of Vector Network Analyzers, which use a generated signal to analyze a network or device to deliver the accuracy needed to perform both linear and nonlinear device characterization, and highly integrated Bit Error Ratio Testers (BERTs), which identify errors introduced into the system when data is received, a critical capability in physical layer characterization, validation and compliance testing.

www.keysight.com



Synopsys

DesignWare USB4 IP Solution

To support all features in the latest USB4 specification, Synopsys introduced the industry's first complete DesignWare USB4 IP solution consisting of controllers, routers, PHYs, and verification IP. The DesignWare USB4 IP operates at up to 40Gbps, which is twice the maximum data rate of USB 3.2 and is backward compatible with USB 3.x and USB 2.0 systems. The DesignWare USB4 IP supports multiple high-speed interface protocols, including USB4, DisplayPort 1.4a TX, PCI Express, and Thunderbolt for efficient data transfer and simultaneous delivery of data, power, and high-resolution video through a single USB Type-C cable.

Synopsys has already achieved successful test chip tapeout of its USB4 IP in an advanced 5nm FinFET process, demonstrating the robustness of the IP across process, voltage, and temperature

variations. The new DesignWare USB4 IP is designed to meet the functionality, power, performance, and area requirements of a broad range of storage, PC, and tablet SoC designs as well as software development debug and easy deployment of artificial intelligence (AI) applications at the edge. As the leading provider of USB IP for nearly two decades, Synopsys is enabling designers to lower the risk and adoption barrier of integrating USB4 functionality into their SoCs.



www.synopsys.com/designware

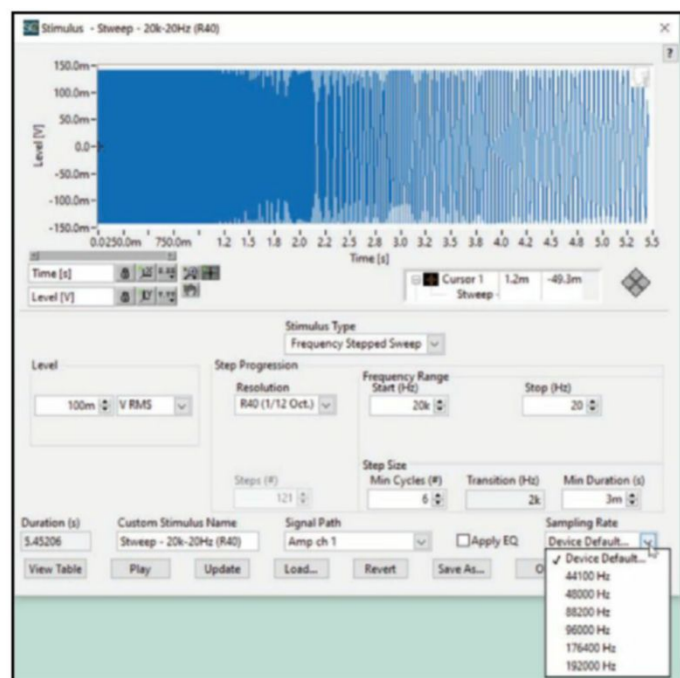
Listen, Inc.

Expanded Integration with SoundCheck 18

SoundCheck 18, the latest release of Listen Inc.'s flagship audio test and measurement system, contains many new features focused on automation and simplicity. Some of these are specifically designed for large-scale integration in factory environments, while others are designed to accelerate R&D testing, particularly where multiple channels are required and/or voice-activated devices are being tested.

Test configuration data, such as limits, test levels, and test signals, can now be passed into SoundCheck from external programs using TCP/IP commands. By externally storing parameters, a single sequence can be used for multiple products, simplifying the number of sequences that a large organization needs to maintain, and reducing test configuration time. This functionality is also useful for applications where a sequence needs to be run many times with different parameters, for example, testing voice recognition with a range of voices or test levels, or where SoundCheck tests are run as part of a larger automated test framework controlled using Python, C#, Visual Basic, LabVIEW, etc.

Setup time is reduced with true plug'n'play hardware configuration. Listen and Portland Tool & Die interfaces are simply connected via the USB cable, and the software automatically creates input and output signal paths and populates parameters such as sampling rate and calibration values. Furthermore, when



Many audio interfaces have a latency that is dependent on the sampling rate, and SoundCheck 18 features improved sampling rate management to allow improved communication with these devices.



substituting devices of the same type, SoundCheck will re-use the configuration previously set up, either manually or during a sequence, so that signal paths and sequence configurations do not need to be updated. This offers significant time-savings when deploying a sequence over multiple stations accelerating set-up time for large-scale operations.

Listen also introduced IEEE 1451 Transducer Electronic Data Sheet (TEDS) support for manufacturer-related information. TEDS support (with compatible Listen hardware) enables automatic identification, configuration, and calibration of TEDS microphones and accelerometers, also saving time on initial hardware setup or whenever hardware is changed.

The development and optimization of test sequences has also been simplified with improvements to gain control selection. Similarly, sampling rate management has been streamlined with automatic latency rate adjustment for devices where latency varies with the sampling rate, and the ability to set the sampling rate in the stimulus editor. The sampling rate, when using WAV files, is also now automatically adjusted, making it simpler to make measurement with wave file signals such as speech or music.

For MEMS microphone tests, the Portland Tool & Die MEMS interfaces can now be controlled from within SoundCheck for faster setup as well as greater measurement reliability as the device settings can be built into the test sequence. Sequences using the R&D grade DCC-1448 interface will also work seamlessly with the production-grade PQC-1448, making it much simpler to transition sequences from the R&D lab to the production line.

Additional new features include WASAPI driver support for improved multichannel capabilities, and enhanced digital I/O control for easy conversion of sequences between hardware options.

The latest SoundCheck 18.1 update includes several improvements and introduces control for the upcoming AmpConnect 621 interface, an all-in-one multichannel interface (6-in, 2-out) that is exclusively controlled via the software for seamless integration with test sequences and virtual instruments.

www.listeninc.com

Brüel & Kjær

New Pressure-Field Measurement Microphone and Cartridge

To help engineers and acousticians achieve accurate measurements in everyday tasks, Brüel & Kjær has created a robust and reliable 1/2" Constant Current Line Drive (CCLD) pressure-field microphone—Type 4971-H-041. This new microphone was designed for measuring close to hard reflective surfaces, close to sound ports of audio devices, or in flush mountings. The new Brüel microphone is also suitable for random-incidence measurements and free-field measurements with 90° incidence.

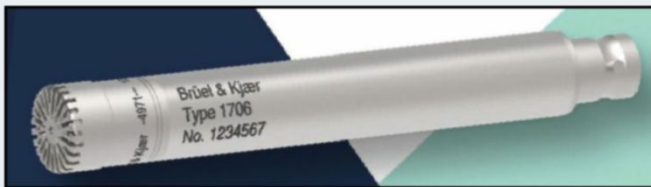
This new version is a factory-assembled combination of a 1/2" pre-polarized pressure-field microphone cartridge (Type 4971) and a high temperature 1/2" CCLD preamplifier (Type 1706 with a BNC connector), which connects to CCLD input conditioning, thus

ensuring all measurements can be conducted using a general data acquisition system. The microphone cartridge and preamplifier are sealed together and are considered as one unit with a single type and serial number stored in the TEDS chip.

The microphone's wide frequency range (5Hz to 20kHz, ± 2 dB), broad dynamic range (20dB(A) to 146dB) and immunity to harsh, unpredictable environments also makes it suitable for component testing and for pressure-field measurements.

The microphone is supplied with the Brüel&Kjær Microphone Viewer, an application that users can install on a PC. The application contains frequency correction curves for different conditions and other useful information, such as the influence of different accessories on the frequency response. The correction data can be uploaded to PULSE LabShop or BK Connect, where it can be used to correct the frequency response in real time according to the actual conditions during use, or exported as a Microsoft Excel file for use in applications such as MATLAB. This data is also available online.

www.bksv.com | www.hbkworld.com



ZUMI Systems

A New Platform for Audio Test and Measurement

If you are an audio engineer, you probably have to deal with a cumbersome chain of hardware and software systems to configure and maintain in order to get the measurement results needed. ZUMI Systems, a company from Sendai, Japan, approached this problem with what it calls "Swiss Army Knife for Audio Testing," in the form of a complete measurement hardware and software system on a single board.

Targeting all needs—from audio engineers doing research to product managers and production quality control—the ZUMI Systems' solution was designed to be easy to use, allowing the configuration of precise, bespoke audio test systems at a competitive price.

The ZUMI CS-A86 Platform

With the ZUMI CS-A86 board at the core of every system sold by the company, ZUMI provides a system that is simple enough for production line transducer testing and advanced enough to meet R&D engineering needs without sacrificing performance.

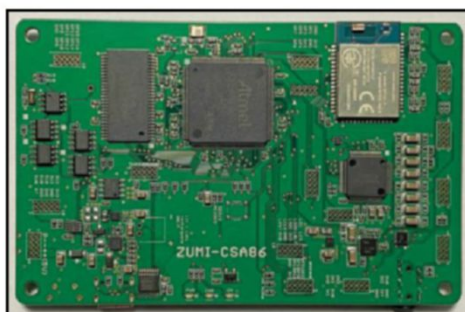
After five years of development, leveraging its 15 years of experience in headphone production test system design with tier one headphone factories in China, ZUMI's CEO Simon Weston is proud of the

results achieved. "The CS-A86 is a single board audio spectrum measurement instrument. It operates autonomously to execute advanced measurement sequences, applies pass/fail criteria and logs all the data."

ZUMI's standalone test and measurement systems are based on the core CS-A86 module and are customizable for any application. The CS-A86 is aimed at implementation into automated test equipment for production and development. The primary function is to obtain frequency response and frequency spectrum measurements for DUTs that require testing of signal paths especially involving filter structures or other signal conditioning elements.

"The ZUMI devices are as comfortable operating independently as a host for Automated Test Equipment (ATEs) as they are in the role of a plugin slave device in a larger system driver over UART or SPI by a host controller," adds Weston.

The test capabilities are extensive and go well beyond frequency response, and they include active noise cancellation (ANC) curves with production trimming, channel balance with the ability to add criteria limits to all measurements so operator error is delimited. "The ability to fool-proof test is a small secret behind the ZUMI devices. I



spent years solving simple problems at factories in China due to complex test equipment,” notes, Weston.

This is why from the factory floor to product thinking all products from ZUMI are based on real application and use cases, while being built on the fundamentals of acoustic engineering.

The ZUMI CS-A86 platform is also very fast. For reference, the platform can load a DUT via Bluetooth, perform sweeps and load the ANC tuning in under 30 seconds. This means saving precious minutes when compared with other systems, and translates into greater production output, at a lower cost.

The built-in capability to configure DUT's via serial interfaces that allows a comprehensive set of quality control tests to be

completed without user input is another differentiating feature. The CS-A86 provides six analog input channels and two hard-wired output sense channels. All channels are sampled simultaneously at up to 100kHz with 18-bit resolution.

The CS-A86 package also includes an LCD display based UI board for displaying measurement results but there is also a simple interface to output to a larger display.

In addition to the CS-A86 platform, Zumi also offers other complete systems in a number of configurations including test systems for Bluetooth ANC Headphones (HP), Bluetooth Earphones (EP), and Drivers (DT). All available as plug-and-play solutions, ready to be deployed.

www.zumisystems.com

Loudsoft

FINE R+D 2020 Acoustic Audio Analyzer System

Loudsoft's FINE R+D is an acoustic measurement system for research and development of loudspeaker, headphone and microspeaker transducers and systems. The Danish company founded by Peter Larsen continues to expand and improve the possibilities of this Fast Fourier Transform (FFT) analyzer packaged with a 1U rack hardware front-end and popular for both R&D and Q&C applications. The updated FINE R+D 2020 solution now supports multi-microphone measurements with three microphones.

One of the main features which made FINE R+D stand out was the possibility to perform anechoic measurements to be done in a normal room, since the reflections from walls and floor/ceiling would be excluded by the FINE R+D software. The 2020 update introduces compensation for room resonances as defined by the National Research Council (NRC) (Canada) standards. The updated software also allows export of normalized curves in LAB and TXT formats.

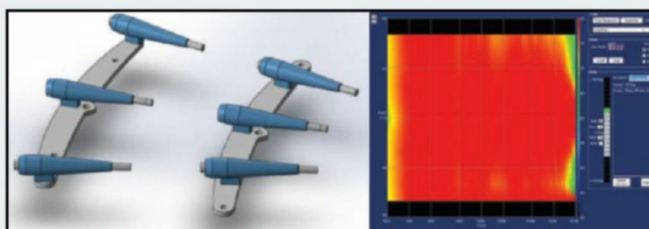
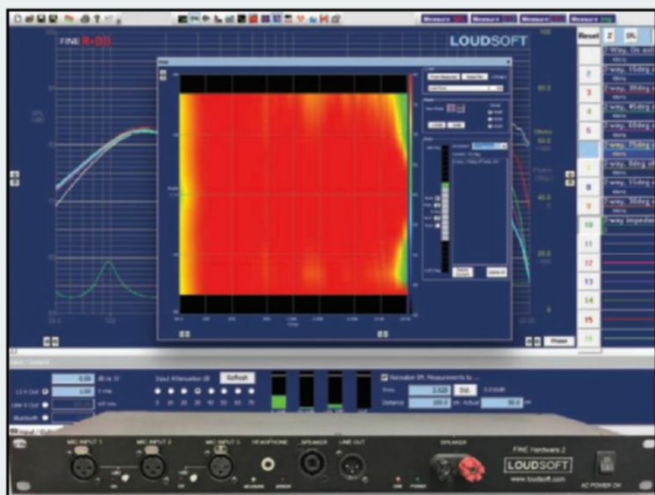
Since the 2019 updates, the Loudsoft FINE R+D solution is able to support sample rates at 48kHz, 96kHz, and 192kHz, perform Bluetooth measurements with delay compensation,

and measure SPL and impedance up to ~100 kHz in normal rooms. The software also now supports direct measurements for applications in smart speakers, amplifiers, microphones, and crossover systems. It performs high-accuracy impedance measurements in LO (Low current) and HI (High current) modes. Also polar plots, contour plots, and automatic conversion from measured frequency responses are included. The new Normalization Mode is ideal for microphone measurements and relative SPL responses.

With the FINE R+D 2020 update, Loudsoft introduced new multi-microphone measurements, now offering a special “Boomerang” 3 Loudsoft microphone array with fixed angles and distances. The new Multimicrophone Array module for FINE R+D 2020 greatly helps making multiple microphone measurements. As an example, users can simultaneously measure SPL from three microphones with one click. This technique is extremely useful for off-axis measurements and multiple measurements of loudspeakers as well as large professional arrays. All curves are easily dragged into the polar/contour display for viewing the dispersion or sound power.

In addition, Loudsoft also offers the new three-microphone array, combining three GRAS microphones into a pre-arranged pattern for different distances and angles. These microphones can be individually configured as for example one 100kHz GRAS 46BE + two microphones, plus two less expensive 30kHz 1/4” microphones.

www.loudsoft.com



Loudsoft's two-way multiple microphone measurements plus contour plot

MiniDSP

New UMIK Measurement Solutions

The hard-working minds at miniDSP introduced two new advanced products in its Acoustic Measurement product line—including the UMIK-2, miniDSP's second-generation omnidirectional acoustic measurement microphone, offering a significant upgrade in performance and functionality compared to the popular UMIK-1. And for multichannel measurements, there's the new UMIK-X USB Multichannel microphone array, offering up to 16-channel audio recording, A²B wiring, and REW integration.

The new miniDSP UMIK-2 has everything that made the first-generation UMIK-1 one of the company's best-selling products, including a convenient plug-and-play USB connection and a unique calibration file for each microphone to ensure accuracy. But the UMIK-2 is also a substantial upgrade, featuring a larger 1/2" microphone capsule that provides lower noise and lower distortion, which is preserved by the 32-bit analog-to-digital converter (ADC) built into the microphone shell. Furthermore, sample rates are selectable from 44.1kHz to 192kHz for compatibility with the widest possible range of measurement and recording software.

As a standard USB Audio Class 2 microphone, UMIK-2 is compatible with any software. But as with previous miniDSP efforts, the UMIK-2 is fully compatible with REW (Room EQ Wizard) and Dirac Live (version 3 and later). Driverless operation with macOS, Linux, Android and iOS, and a supplied ASIO driver for Windows ensure simple integration into many other measurement and recording applications.

And miniDSP once again stretches the limits of affordable acoustic measurement equipment with the UMIK-2. Priced at \$195 US, this new reference measurement microphone represents an incredible value for an all-in-one low noise, high SPL, high bit rate microphone.

A MEMS Multichannel Solution

The next product in miniDSP's Acoustic Measurement product line is a much more ambitious development. The new UMIK-X



The miniDSP UMIK-X multichannel solution was designed for tuning large rooms or challenging acoustic environments, performing multiple measurements at multiple positions.

was designed as a solution for tuning large rooms or challenging acoustic environments (e.g., a vehicle cabin), performing multiple measurements at multiple positions. In order to offer a more efficient and low-cost way to achieve those complex and time-consuming tasks, miniDSP leveraged the latest MEMS microphones from Knowles to create the miniDSP UMIK-X.

This is a distributed microphone array containing up to 16 Knowles MEMS microphone capsules for multichannel measurement. Four microphone devices with a low noise buffer and high-performance modulator are mounted onto each UMA4 microphone module, and up to four UMA4 microphone modules can be daisy-chained with a single twisted-pair cable using Analog Devices' A²B protocol for multichannel audio transport.

With this technology, a measurement chain can be up to 40 meters long, so more than enough for a home theater and even worship centers and auditoriums. The UMA4 microphone modules are interfaced to the computer with the included USB-A²B interface, which provides 16 channels at 24-bit/44.1kHz or 48kHz. The USB-A²B is a UAC-2 compliant interface that is also USB powered, allowing the use of any standard USB adapter in any part of the world.

Other applications for the UMIK-X include environmental monitoring and distributed soundfield measurement. The distributed omnidirectional microphones in the array allow quick measurement and analysis in applications such as room and hall equalization and distributed subwoofer tuning.

Furthermore, the UMIK-X is compatible with Room EQ Wizard's latest multi-channel measurement capability. In fact, miniDSP partnered with REW and offers the cost of a full REW Multichannel measurement software license with each UMIK-X purchase, so users receive a complete software and hardware package to start their projects.

Starting at \$550 US, the UMIK X solution is available in a basic package and complemented with a range of accessories. The complete measurement kit includes everything needed to get started: USB interface, mic arrays, REW multichannel license and adjustable 1.2m stands. The UMIK-X is now available on miniDSP's webshop.

www.minidsp.com



The new miniDSP UMIK-2 USB measurement microphone is a substantial upgrade.



Fast-Track Your LE Audio Development

Ellisys offers an array of protocol analysis solutions that support the new Isochronous Channels feature for Bluetooth® Low Energy.



Bluetooth Vanguard™

Advanced Wireless
Protocol Analysis System

BT + BLE + Wi-Fi 3x + WPAN

www.ellisys.com/bv1



Bluetooth Explorer™

Dual-Mode Bluetooth
Protocol Analysis System

BT + BLE + Wi-Fi 2x

www.ellisys.com/bex400



Bluetooth Tracker™

Ultra-Portable Bluetooth Low
Energy Protocol Analyzer

BLE + Wi-Fi 1x

www.ellisys.com/btr1

QuantAsylum

A Real-World View of Test and Measurement

Test & Measurement Focus

By
David Logvin
(United States)



Photo 1: Matt Taylor, President and founder of QuantAsylum (insert). In the main photo we can see the company's area used for product development and prototyping.

Following the review of two QuantAsylum's products—the QA401 Acoustic Analyzer and the QA451 programmable load—featured in the December 2020 edition of *audioXpress*, there was still a lot of engaging information about the company and its activities to be told. This article provides an overview of QuantAsylum, its history, philosophy, and product roadmap.

QuantAsylum is a privately held test and measurement company, with a solid catalog of product offerings for the automated testing of audio products for both design and manufacturing environments. Based in the Seattle, WA area, the company was founded by Matt Taylor, a veteran technologist with a breadth of design and management experience—including work at HTC, Microsoft, Motorola, and most recently as the CTO of Gogoro, a Taiwan-based manufacturer of electric scooters.

Not too long ago, Matt Taylor, President and founder of QuantAsylum, agreed to share an inside look and virtual factory tour of the operations at his 2500 ft² facility in Snoqualmie, WA (**Photo 1**). Inside this dedicated building is a 1000 ft² prototype area.

The blank PCBs are sourced from China,

components are purchased from US distributors, and the boards are assembled in Seattle (**Photo 2**). The boards are then installed in cases and the final test is performed on all units at the QuantAsylum corporate headquarters in Snoqualmie.

The top and bottom aluminum cases (**Photo 3**) are custom fabricated as blank panels in China. Cleverly, the size and footprint is universal across all QuantAsylum products, so the casework can have full-color graphics printed on demand using their in-house UV printer. The UV printer is used to print artwork on the top, front, and back panels of every case (**Photo 4**). The UV printing process isn't quite as good as a traditional silk screened panel, but it provides unrivaled flexibility for a just in time US-based final assembly process.

The assembled hardware is tested at one of two test stations, each equipped with a fanless

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The ZUMI CS-A86 is a single board audio spectrum measurement instrument.

The CS-A86 operates autonomously to execute advanced measurement sequences, apply pass-fail criteria and log all data.

The device is aimed at implementation in automated test equipment for production and development. The primary function is to obtain frequency response function measurements for devices under test (DUT) that require testing of signal paths especially involving filter structures or other signal conditioning elements. The built-in capability to configure DUT's via serial interfaces allows a comprehensive set of quality control tests to be completed without user input.

Features

- Standalone audio spectrum measurement instrument
- 6 input channels
- 2 x 2ch analogue output
- Bluetooth A2DP source, HFP AG, SPP
- Digital interfacing via USART, UART, SPI, I2C, SPP, GPIO

Applications

- Custom test fixtures for drivers and mics
- Automated production measurement for audio related PCBAs
- ANC filter testing
- Quality control systems with remote data reporting



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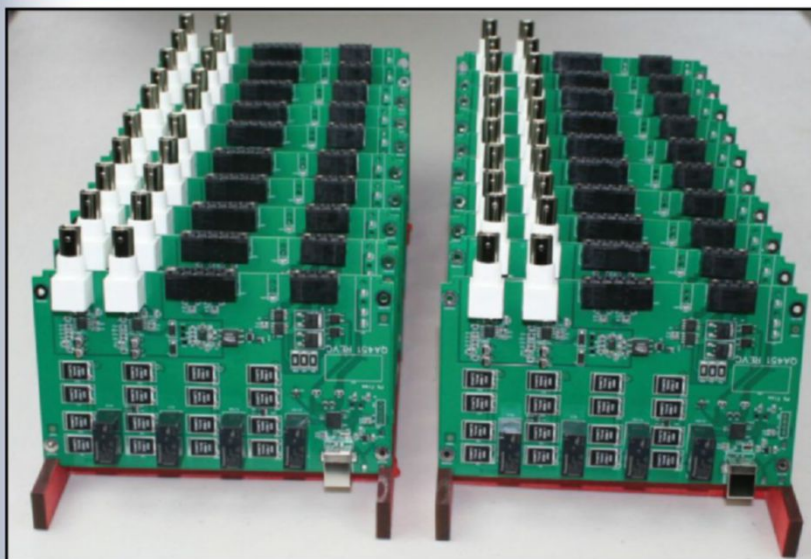


Photo 2: This image shows 20 QA451 (programmable amplifier loads) PCBs ready to be installed into sheet metal cases.



Photo 3: The QA401 panels have been fully printed and are ready for assembly with the PCBs.



Photo 4: Three top cases are printed in parallel with a UV color printer. After the white layer is complete, the color layer is printed next.

and low-cost micro PC mounted on the back of the touchscreen interface (**Photo 5** and **Photo 6**).

Finally, Matt Taylor shared a preview of a new, not yet shipping product, the QA440 relay box (**Photo 7**). This innovative accessory enables flexible and seamless routing of audio channels for production testing of a 5.1 channel receiver via the QA401 analyzer. With this relay box between the DUT multichannel receiver and the QA401, the receiver can be tested under full-load on all channels, without operator intervention (e.g., cable connection and disconnection), thanks to onboard integrated relay control.

Given the circumstances not allowing for in-person visits, Taylor was kind enough to provide a behind-the-scenes interview to *audioXpress* and preview what's in store for QuantAsylum.

David Logvin: QuantAsylum launched its first general-purpose audio product, the QA400 Acoustic Analyzer, in 2012. Can you tell us more about the early start-up years and how you grew this business?

Matt Taylor: The first audio product (the QA400) in 2012 gave us an initial glimpse into what was needed from an audio perspective. There were a lot of companies coming to my long-time friend and business partner (based in Shenzhen, China) and those early requests helped shape the longer-term portfolio. That was an especially important time for understanding the business. Coming from Microsoft, the easy answer to meeting the needs was "Just tell them to write a program to control it!" But many of these companies wanted to test simpler things like piezo buzzers and test tones coming out of integrated circuits. And they needed to test them very, very quickly. They had the skills to build sophisticated material handling machines, but the automation software for audio testing needed to be more turnkey. In short, we weren't there yet.

The QA401 analyzer was introduced in 2015, and then the automation solution arrived in 2018 with a software product called Tractor. Tractor enables engineers to quickly build sequences of tests without writing any code. Tests results can be stored locally or in the cloud, and operators need only interact with a touchscreen and barcode scanner to verify the operation of the product. So, in retrospect, it took about six years to settle on the recipe and understand what the market was going to be. And there were products along the way as the recipe was being figured out (DDS

generators, oscilloscopes, logic analyzers) that we probably won't revisit because the larger market has served that segment well.

Moving forward, continuing to build out this ecosystem is the focus over the next few years. Multichannel voltage measurements, current measurement, relay control boxes are all coming. The aim is to broaden the appeal to engineers at mid- and small-size companies that are creating products that need to be tested at manufacturing, warranty service, and more—quickly and with a cloud-based audit trail if needed.

David Logvin: Can you share an overview of the current manufacturing capability and share some details about the production team?

Matt Taylor: The production "team" is all contractors, which can expand and contract with demand. Top and bottom cases and PCBs are made in China, and local assembly houses populate the boards in Renton and Bellevue (both suburbs of Seattle, WA) from parts purchased from US distributors. From there, the assembled

boards go into cases, and then we test them and ship them. The time required to take a populated QA401 board, assemble it into the case, test it,



Photo 5: Outgoing hardware is shown undergoing final tests.

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box it, label it, and have it ready for UPS pickup is under 10 minutes. I do most of the design, with some occasional help with mechanical, FPGA, and software from folks around the world.

David Logvin: In many ways, QuantAsylum has avoided a lot of traditional business models that the test and measurement world favors. Conventional wisdom has often pushed lower-cost measurement systems toward a sound card-based approach. Is calibration the primary reason you decided that the QA400 (and the QA401) would be dedicated hardware systems?

Matt Taylor: I think sound cards fall short for several reasons. First, I don't think common USB audio interfaces are suitable for factories—ASIO maybe. But there are too many unknowns with background sample rate conversions, bit depths changing seemingly at random, other apps trying to use the card. Anyone that has ever tested HiFi DACs will tell you it is an involved process to be certain the DAC that says it's operating at 24-bit/48kHz is actually operating at that rate end-to-end. Second, sound cards aren't usually galvanically isolated from the PC. And finally, there's the issue of calibration and gain. A gain knob in a factory is a big problem. Gains need to be fixed and tightly controlled. A QA401 in loopback will typically exhibit gain error standard deviation of under 0.01dB from one unit to the

next. That means the QA401 you buy tomorrow will have very good agreement with the QA401 you buy next year.

David Logvin: You take the support of small-scale manufacturers and start-ups very seriously. You have made customized front panels for rack mounts and low MOQ specialized test gear with souped-up specs for unique applications (e.g., the QA480). How are you able to deliver these customized products without overloading your manufacturing and operations teams?

Matt Taylor: Tools and the Internet are the enabler here. If someone wants four products in a single 2U rack space, a contract ME can create the front panel DXF in under an hour, then we'll laser cut it out of acrylic in-house to verify the fit, and then that goes to be laser cut out of aluminum in Nevada or Utah, then we get it back in a week and get it powder coated locally. And then the graphics can be printed using the UV printer. These steps are all part of the normal product flow, so a custom front panel just tags along with the other material. And usually, the requests for custom front panels are from manufacturers that are doing an install into a rack. I think it's a smart thing to support because if they need one they usually need more. For the QA480 (a low distortion 1kHz oscillator + notch), several of the subsections on the PCB had already shipped on other products and the case artwork can be printed on demand with the UV printer. So it wasn't a lot of effort to go from something that was originally for internal use (the QA480 was created for testing a next-generation analyzer) to something that was useful to others too.

David Logvin: When I worked for small consumer electronics manufacturing companies, my mantra was to not have more SKUs (stock keeping units) than employees—to keep the focus on innovation and the core products. Do you think that is still practical today?

Matt Taylor: Yes, I think it's still a good rule. I've managed big pieces of large engineering efforts with hundreds of engineers focused on a single product, and also startup efforts where I was a co-founder and first engineer. And almost universally there's a push to get a product out the door, take the support pain by building a large support organization, learn, refine, repeat. The early days of smartphones at Microsoft were so

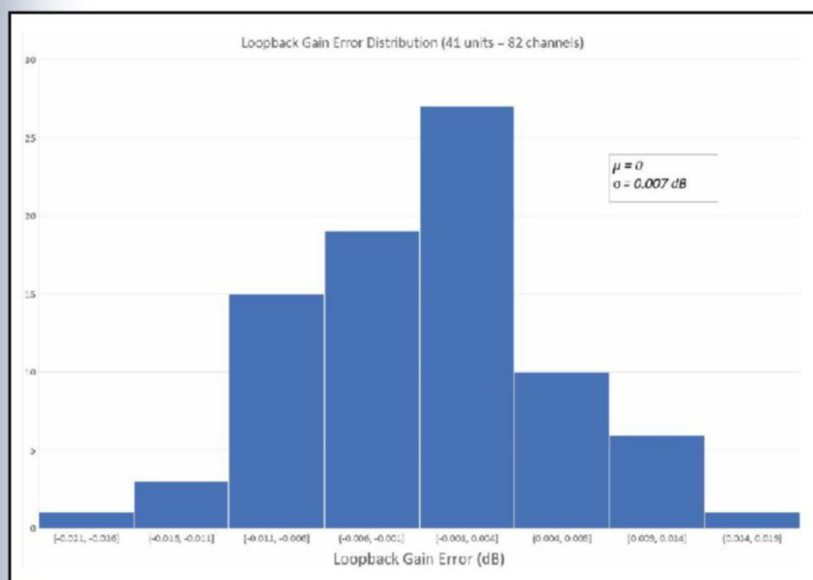


Photo 6: A histogram plot displays the loopback gain error measured at final test on 41 units of the QA401. There are 82 channels (left and right for each unit) shown in this plot, with the impressive statistical result of 0dB gain error and a sigma of 0.007dB.

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complicated that we'd see return rates of 30% to 40% in some cases. The products weren't broken—the consumer just didn't want to cope with the rough edges and minutia of getting everything set up. But the support effort was large and the support organizations were large.

With QuantAsylum, the focus has been more on managing growth toward a particular support effort and return rate. That means focusing on a core set of scenarios and doing those really

well, rather than doing everything. As an example, things like THD vs. power out—these are the bread-and-butter measurements you have to do well. I like to think of our tools as your trusty digital volt meter (DVM)—the first place you turn for your fundamental measurements. We don't do waterfall plots or esoteric distortion perception metrics. But if you want to know the output impedance of your amp vs. frequency, that's a click away. The result is a return rate under 1% and a support effort that is a very small part of the day in spite of thousands of products in the field. The business doesn't work otherwise.

David Logvin: Looking at your product road map, are there plans to integrate more instruments into a single chassis, or do you plan on staying with the modular-based approach that you have leveraged to date?

Matt Taylor: All of the products share a common custom mechanical design. This makes it easy to order cases without having to forecast; you just know they will be needed. Moving forward that stays the same: stick with the common chassis with custom front-panel for each product. For companies that would like the functionality in a rack, we can create a new faceplate and bolt two or four boxes to that faceplate. For example, for mic testing you might have two QA471 mic pre-amps, a QA460 transducer driver, and a QA401 audio analyzer. Those four boxes can be mounted to a single 2U rack faceplate. That gives you the ability to drive a speaker with the QA460, handle two mics (a reference and a DUT) each with 10dB of gain with the QA471, and then measure the signals with the QA401, take the difference and compare that to a pass/fail mask.

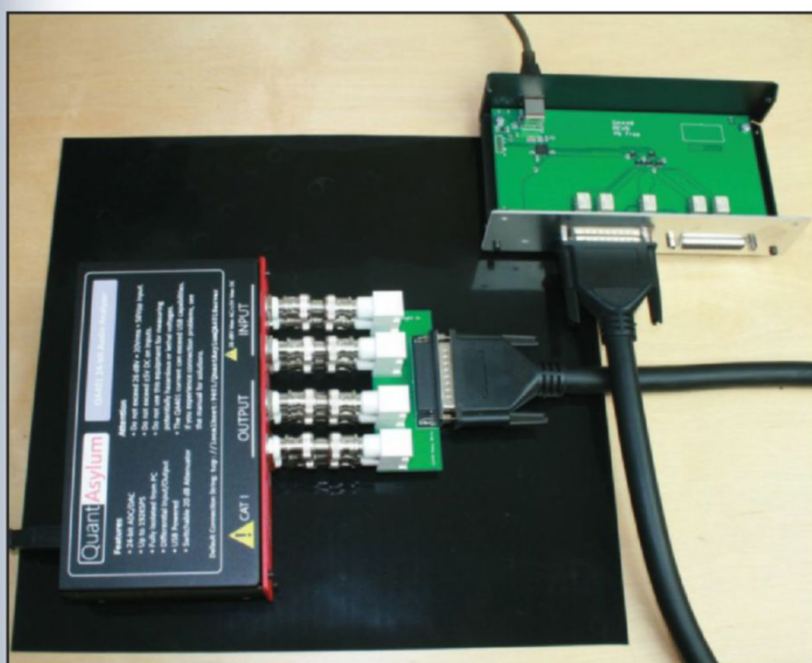


Photo 7: The QA400 relay box (shown without a cover and with an attached breakout board) is also a boon for the final manufacturing testing of the QA401 itself, allowing the loopback test to be performed in a single operation, with no cable changes whatsoever. All inputs and outputs go into the QA440 over a common DB25 cable. The routing box then routes outputs to inputs and shorts inputs, as required by the test sequence.

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. Logvin 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, Logvin 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.



David Logvin: Are there any discontinued products that you plan on revisiting and updating in the near future?

Matt Taylor: Yes, there's an omnipresent need for tools that permit very detailed power analysis. Expect to see something in 2021, hopefully that will make it easy to log current consumption at a very high rate and over several decades—from nA to amps. There was an early product called the QA150 that had some of that functionality.

David Logvin: Another area of key differentiation for QuantAsylum is that the software is free and open source. Most test and measurement

companies charge for their software and then charge upgrade fees for R&D/quality control modes or institute annual fees or maintenance and software service plans. Is the rebellious approach you have instituted something that can be maintained as QuantAsylum grows its business?

Matt Taylor: Yes, the software and upgrades should always be free. We can't support every product forever, but the QA401 is five years old and still seeing around 10 software releases per year. Open source is very important for software like Tractor, because the manufacturing engineer may need the flexibility to do something we couldn't anticipate.

I think another big shift here has been getting up on Github this year and migrating to REST as a control means for the product. Github means the developers have a shared common workflow. They can log bugs, submit feature requests, and Github works very well for technical writing; sharing plots and data is very fast. It's a great way to interact with other developers and users. REST is a design philosophy for controlling and querying the things via web API. Instead of needing an engineer that knows how to write GPIB or LabVIEW—a fairly specialized set of skills—you can use an engineer who knows how to code web apps. And how cool is that!

An IT person who knows how to use Python to query an employee's salary from the web can also make a THD reading from the QA401 using the same knowledge and same tools. The REST software interface first appeared in the QA350 DVM, and it's been a good move. A user just shared an application on Github that they wrote in Javascript where the audio analyzer was running in a web page with really impressive performance. That is the power of REST for test and measurement—you leverage the know-how of millions of developers.

David Logvin: With more folks working from home these days, do see your role expanding in making home-based measurement systems both more useful and affordable?

Matt Taylor: Yes, I think so. Second quarter of 2020 showed more than two times growth vs. the same quarter of the previous year, and I suspect a big part of that was a shift to home-based labs. The affordability is tightly tied to performance. At a twentieth of the price, the QA401 won't match the performance of a \$10,000 analyzer. But most products shipping today don't need a \$10,000 analyzer to confirm that their output is 0dBV $\pm$ 0.2dB. Nor do they need a \$10,000 analyzer to confirm the THD at 100W is -90dB. There's no question the very high-performance analyzers are needed in many R&D settings. The bigger question is, "now that you need to build out five test bays for your manufacturing line, does the QA401 product have enough margin to verify your product?" And I think for most products, the answer is yes. 🚀



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The Lab Bench of 2021: Virtins Multi Instrument Measurement Software

The Research

By
Stuart Yaniger
Photos by Cynthia Wenslow



Photo 1: The stack of the RME-2 FS R and the Linear Audio Autoranger II doesn't look intimidating, but "though she be but little, she is fierce."

In 2015, Stuart Yaniger wrote a series of *audioXpress* articles discussing how to adapt home computers and sound cards for audio test and measurement. In this article, he experiments with 2021 technology to see what sort of measurement capability he can now achieve and updates his findings and recommendations for our readers.

As with anything related to computers and measurement, the world has moved quickly. The availability of higher-powered computing, much more capable sound cards and external audio interfaces, and far more sophisticated software has vastly expanded the capabilities of small labs and hobbyists to measure and characterize acoustic and electronic devices. In 2015, the appearance of measurements in audio discussions was an exception—these days, it seems like nearly every enthusiast is waving around measurement microphones to generate data for digital room correction, acoustic treatments, and loudspeaker placement.

From a personal standpoint, I've also spent several years using top professional gear (e.g., Audio Precision products) on a daily basis, and that has certainly recalibrated my standards. Of course, pro equipment of that level carries a significant price tag, which is not a big impediment in larger companies, but can be crippling for small labs and individuals who have not won the lottery.

So, all of that said, I thought it would be interesting to see what sort of measurement capability one can achieve using 2021 technology and update my findings and recommendations from the Stone Ages of 2015. Although there's a lot of options in the low-cost range, I want to

demonstrate what a moderate outlay can achieve. So the setup I'm using here, while not cheap, can in many respects outperform the professional gear. I'm very much in the position of a high school violinist who has been practicing on a Stradivarius—the limitations will almost certainly be the user, not the equipment!

Evaluation Hardware

First, a brief mention of the hardware I used in this evaluation. These days, desktop computers are far rarer than 6 years ago, and options for actual sound cards are likewise much more limited. However, external interfaces abound, usually set up as USB devices, and the available performance is astounding. One example is the RME ADI-2 Pro (see Resources), now updated to the FS R model.

The ADI-2 Pro FS R (which I'll just call ADI-2 for brevity) is capable of a 768kHz sample rate and 24-bit operation. After filtering, the upper frequency limit for the inputs is 180kHz. This is somewhat more restricted than, say, the Audio Precision APx525 that I use for most of my reviews, which samples up to nearly 2.5MHz and has a 1MHz high-frequency limit, but more than generally needed for audio use.

The ADI-2 has balanced inputs that can accept up to

12V signals before overload, and balanced outputs that can output up to 12V. By using the balanced headphone outputs, signals up to 19V can be generated. It also has digital I/O, specifically SPDIF and AES/EBU through coax, XLR, and Toslink.

That said, there's a lot of other interface choices out there, and the intrepid experimenter can probably find something less expensive that will still offer top performance. I chose the ADI-2 because of its feature set for my uses beyond audio measurement, as well as the raw performance. Oh and also, because it was a generous gift from *audioXpress* technical editor Jan Didden!

To extend the input voltage range and to speed calibration, I used the Linear Audio Autoranger MkII Measurement Interface (ARII, see Resources). This adds single-ended I/O, voltage gain and attenuation, and calibrated AC voltage measurement to the ADI-2. The combination of these two devices is compact and accessible (**Photo 1**).

Next, the software, which is really the centerpiece here. Most of the software choices I outlined in my earlier articles are long obsolete. When you see plots from them posted at online forums and blogs, it has that same feel as seeing someone with an AOL or Compuserve email address. But a few have survived and evolved.

For example, Room Equalization Wizard (REW) is an extremely popular freeware package, with a remarkable feature set, constant upgrades, and enthusiastic support from the developer and the users. It's very easy to use and does a great job with both basic audio measurements (frequency response, distortion) and some fairly sophisticated room and acoustic analysis capabilities. I use it semi-regularly, and manage to slide back into a comfort zone after an hour or so. It's not quite as intuitive or versatile as the APx software running Audio Precision's APx-series analyzers, but it's remarkably good, and the graphics are much more attractive.

ARTA is also still alive, well, and living in my computer. Along with its companion software (STEPS and LIMP), it will handle almost all routine audio measurement (distortion, frequency response, noise, etc.), and is quite intuitive and easy to use. It's popular and justifiably so.

But if you want something that's truly professional software in the manner of LabView or MathCAD, Virtins Technologies' Multi Instrument (MI) offers much broader capability, extending well beyond audio into nearly any kind of electronic instrumentation.

The User Manual is 468 pages of rather terse prose, giving a hint as to the software's versatility and feature set. I previously wrote about an

earlier version of this software, so some of this will necessarily be a bit of a rehash, but the earlier article has a still-valid broad overview.

In brief, with the MI package, the user gets an oscilloscope, a signal generator (arbitrary waveform), a spectrum analyzer, a frequency counter, a XY recorder, a signal recorder, an LCR meter, a correlation analyzer, a data logger, a logic analyzer, and more—you get the idea.

The user can select any combination of these instruments to run for an experiment. For example, you can use an oscilloscope, a signal generator, and a spectrum analyzer simultaneously when doing distortion measurements.

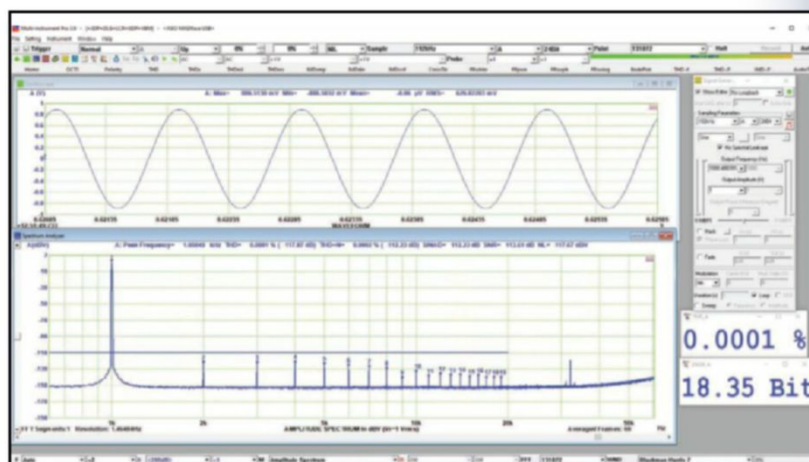


Figure 1: Multi Instrument (MI) is just that: The user can run any number of instruments simultaneously. This is a loopback measurement of the RME-1/ARII measurement hardware.

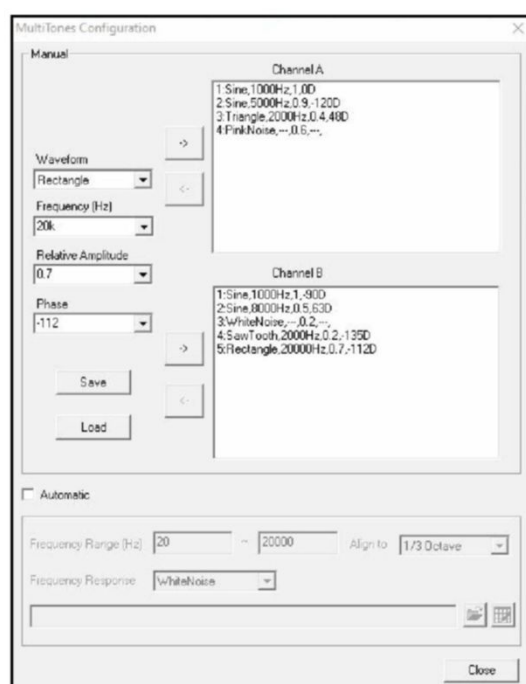


Figure 2: MI's Signal Generator allows remarkably flexible creation of composite waveforms.

Figure 1 shows a loopback measurement at 1V of the ADI-2 and the ARII, where you can see the automatic measure of THD, THD+N, SINAD, total

noise, and signal-to-noise on top of the distortion spectrum.

There are multiple versions and options: I used the fully-equipped MI Pro, and given the relatively low cost of the entire package, I would strongly recommend simplifying your life by shooting for the top. You'll find that it's much more than you think you need, but you can grow into the feature set. It's expensive (\$200 to \$500) compared to most hobby-level software, but much less expensive than professional measurement software—the price tags of which always contain a comma.

A software key activates the license on a single computer. For a nominal fee, you can get a dongle allowing use on multiple computers (but not simultaneously). Just try not to lose it... You also have several choices of skins, and I still can't decide if I like white or black backgrounds better, so I'll be cheerfully inconsistent.

The Instruments (well, some of them)

The least superficially impressive looking instrument in the bunch is probably the Signal Generator. But its seemingly simple appearance belies the deep functionality of the virtual instrument. For example, the multitone generator enables the user to pile in any collection of waveforms one could think of, with varying frequencies, wave shapes, and amplitudes. For periodic functions (sine, rectangular, sawtooth, triangle), the relative phase can also be varied, and composite signals can include various noise functions as well.

Figure 2 shows a screenshot of a constructed waveform that I like to call the "Kitchen Sink." It's not exactly useful, but it's a nice encapsulation of the versatility of waveform construction. What can be useful, however, is the ability to add a small amount of white noise to a test signal for dithering. This can allow determination and removal of any quantization distortion in low level measurements. A simpler version was generated with three sine waves of varying frequencies and phases. **Figure 3** shows the time and frequency domain of that signal fed into an inexpensive, small amplifier I had on hand. The varying phases can be seen by looking at the envelope of the periodic signal and noting their change from cycle to cycle.

What I didn't show here, but mentioned in my earlier article, is the innocent-looking "mask" option. The mask (in its simplest form) sets all points within it to zero, so that signal bursts can be generated. The mask can be phase locked to the signal or independent of it. The mask can also be realized as a window

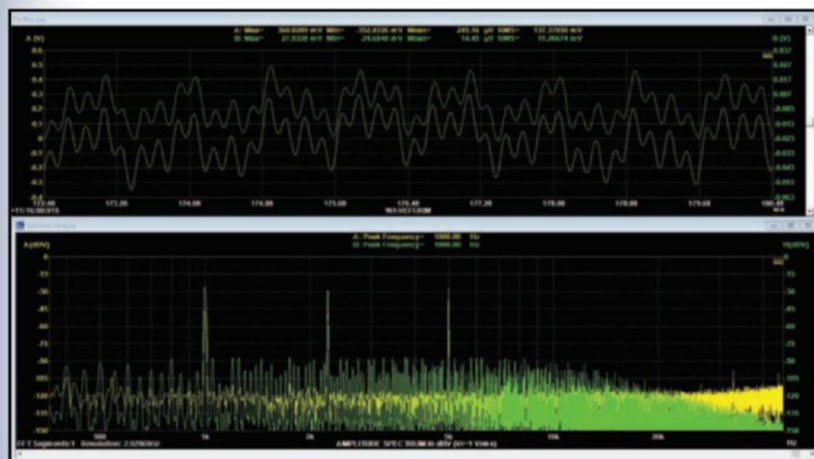


Figure 3: Adding together three sine waves of different frequencies and phases creates a time domain wave/envelope, which deconstructs to a relatively simple frequency magnitude spectrum.

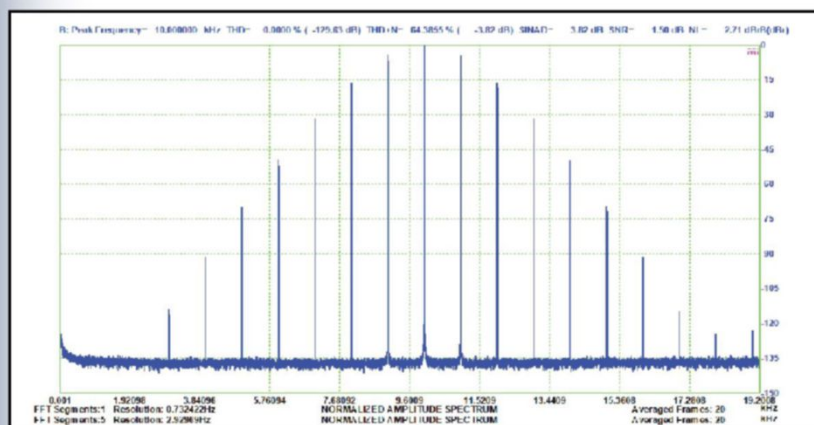


Figure 4: MI can also frequency modulate a carrier as a test signal.

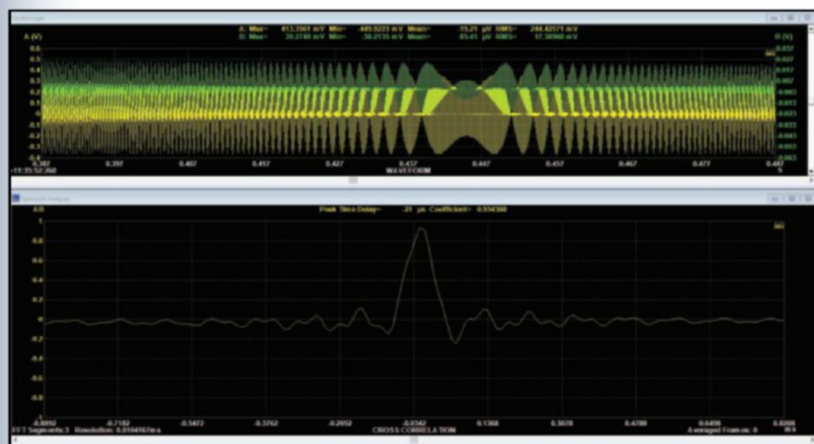


Figure 5: The cross-correlation mode enables rapid and unambiguous determination of time delays.

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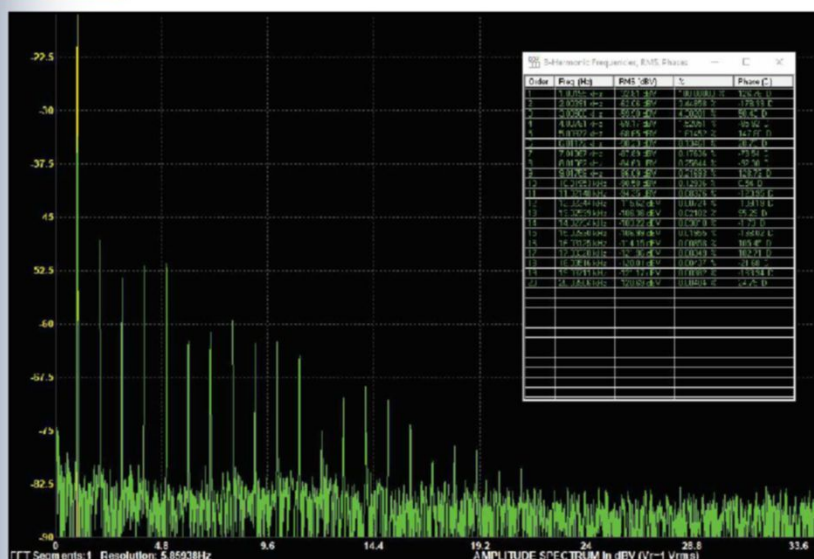


Figure 6: The Spectrum Analyzer can capture both magnitude and phase of distortion harmonics.

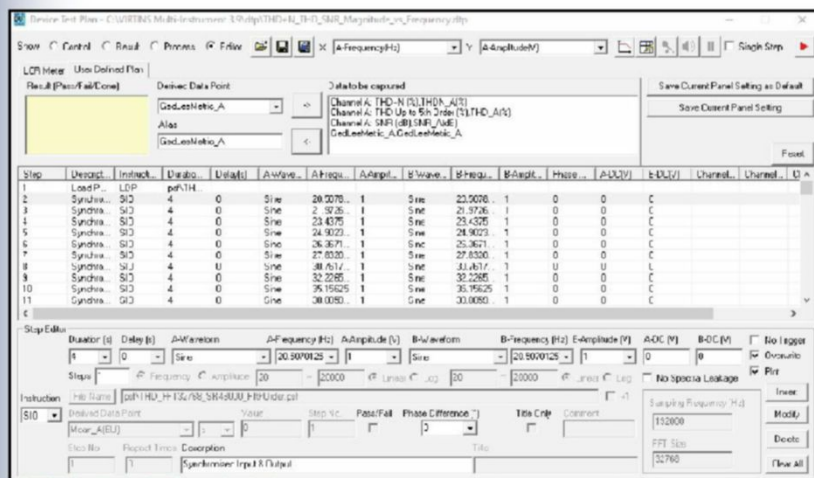


Figure 7: The Device Test Plan enables measurements to be sequenced, for example this stepped distortion vs. frequency measurement.

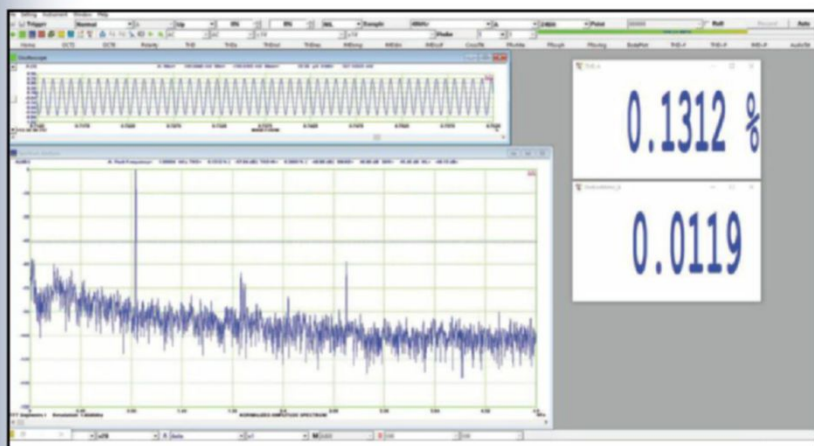


Figure 8: A 1kHz sine wave at 90dB SPL shows low distortion, has a low GedLee Metric, and subjectively sounds clean.

function, that is, a defined increase and decrease in amplitude to shape the envelope of the burst.

The signal can also be swept in frequency or amplitude, linear or logarithmic. A nice touch is the ability to have the signal sweep both forward and in reverse. Oh, and did I mention that you can generate FM signals (**Figure 4**)?

The Spectrum Analyzer is similarly feature-rich. Besides having a ridiculously large number of available windowing functions and being able to do fast Fourier transforms (FFTs) up to 4.2 million points, the spectrum analyzer includes cross-correlation and auto-correlation, as well as measurements like coherence that are derived from the correlation functions.

Figure 5 shows a cross-correlation derived from a frequency sweep (seen in the upper oscilloscope window) between signals measured by the two channels of the ADI-2 in a voltage-current setup described later in this article. The cross-correlation peaks at about 21μs, which is, not coincidentally, one sampling period ($f_s = 48\text{kHz}$), neatly measuring the time delay between channels with a complex signal.

A nice feature of this instrument is the ability to measure not only the magnitude but also the phase of the harmonics. **Figure 6** shows the distortion spectrum of an inexpensive lab amplifier along with an MI-generated table of harmonic amplitudes and phases.

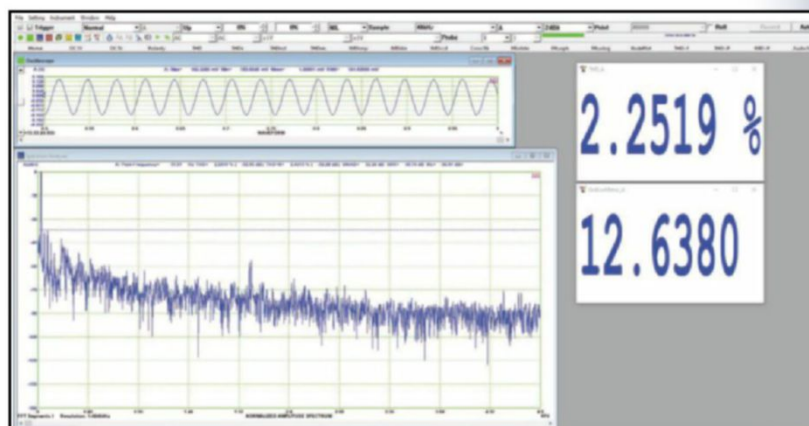
As they say on late night TV commercials, “But that’s not all!” MI has a function called Derived Data Points (DDP), which display a numerical value derived from a frame of data; of course, it can be set to update with each acquired data frame. There are more than 200 different options, depending on which instrument is active. You can see examples of the DDPs (in this case, Effective Number of Bits and SINAD) in Figure 1. Further, upper and lower limits on the value can be defined to either set off a visual alarm in the window, sound an alert, or start another process to bring the measured value back into range. For those who want to use MI for statistical process control, the alerts can be set for both control limits and specification limits (in the MI language, “Low-Low, Low, High, High-High”).

MI also enables the user to create User Defined Data Points (UDDP), where the definition contains acquired data, constants, and mathematical functions set up in combination by the user (as the name implies) much in the manner of a spreadsheet. So any sort of customized measurement can be assembled using the building blocks of acquired data. This is an amazingly powerful tool, and as users come up with new needs for measurement

metrics, Virtins will often incorporate these needs into the options in succeeding versions.

An example of this is the Total Distortion metric, which using a complex signal as a stimulus, subtracts the scaled input from the output of the device under test (DUT). So if you create a multitone test signal (e.g., the 42 tone signal I use in many tests) and measure the response to it, MI will report back the total harmonic, intermodulation, and anharmonic components of the measured signal. This addition was in response to a user request on the company's support forum (see Resources).

Virtins MI also has a feature called Device Test Plan (DTP), where a sequence of actions (including UDDPs and conditional statements) can be programmed. An example of a DTP being created is shown in **Figure 7**. The user initiates the Plan and it runs through whatever steps are included. For this specific example, a constant amplitude signal is stepped through varying frequencies to create four charts, THD+N vs. frequency, THD vs. frequency, signal-to-noise vs. frequency, and GedLee Metric vs. frequency. It's easy to see that the signal amplitude and phase could be varied as



well. There are 25 instruction options provided, and up to eight charts can be displayed simultaneously.

A Few Example Measurements

GedLee Metric: For a very long time, audio engineers have been aware that human ears have differing sensitivity to different harmonics, and this varies again with distribution or harmonics and fundamental frequency in a rather complex manner. The upshot is that THD alone is a relatively poor way to characterize the audibility of distortion. Various weighting schemes have been developed over the years to improve the correlation between measured distortion, that is, deviation from a linear transfer function, and perception.

Figure 9: From the same speaker, reducing the frequency to 31.5Hz creates an unpleasant sound, reflected more in the GedLee Metric than in the THD.

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Figure 10: Wow and Flutter measurements can be displayed unweighted (left) or weighted (right), and are resolved into both separate and combined wow & flutter.

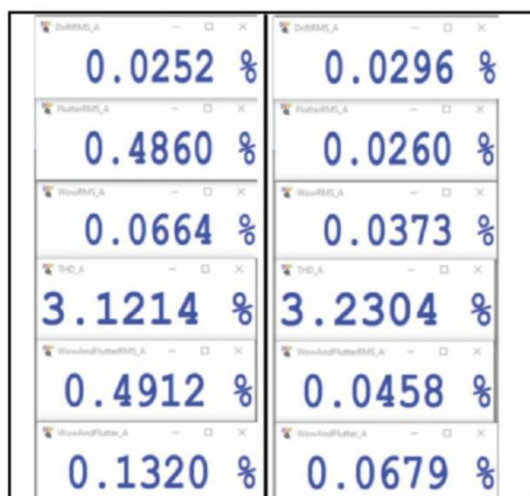


Figure 11: This interface circuit allows the simultaneous measurement of voltage and current through a transducer.

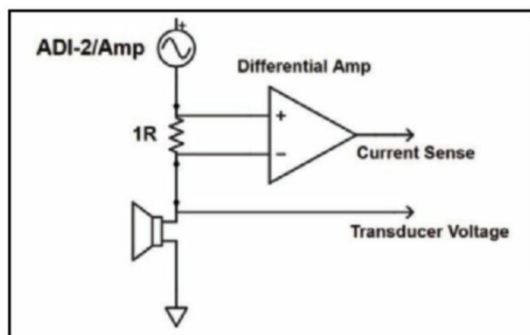
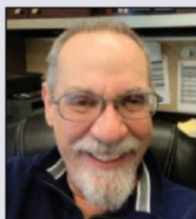


Photo 2: Taking measurements is not always pretty. This is a kluged together setup to simultaneously measure current and voltage when driving a small loudspeaker.

About the Author

Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently works as the R&D director for a construction products company. His professional research interests have spanned theoretical physics, electronics, chemistry, spectroscopy, aerospace, biology, and sensory science. One day, he will figure out what he would like to be when he grows up.



About 20 years ago, the team of Earl Geddes and Lidia Lee developed a rather sophisticated weighting system that took into account the distribution of harmonics, their relative phases (which were not accounted for in earlier weighting schemes), and the overall curvature of the transfer function. This became known as the GedLee Metric (see Resources). The GedLee Metric, G_m , is defined as:

$$G_m = \sqrt{\int_{-1}^1 \left(\cos\left(\frac{x\pi}{2}\right) \right)^2 \left(\frac{d^2}{dx^2} T(x) \right)^2 dx}$$

where $T(x)$ is the normalized system transfer function. In this formula, the squared second derivative weights higher-order harmonics more heavily than lower-order harmonics, with squaring used to make any curvature positive independent of direction. The squared cosine term weights deviations near the zero crossing more heavily than at higher values, since cosine is at a maximum at zero and goes to zero at $\pm\pi/2$, much in the manner of a windowing function.

Geddes and Lee's data suggested that $G_m < 1$ would be imperceptible, and $1 < G_m < 3$ would be relatively tolerable. There are assumptions in their model of the system being time invariant and non-hysteretic, which is not strictly true for real-world transducers. Nonetheless, the model seems to work well in predicting the audibility of distortion.

To measure the G_m of some speakers I had handy (Vanatoo Transparent 1E powered mini-monitors), I used a PCB Piezotronics 376A32 1/2" condenser lab mic attached to the RME-2's input via a 48V phantom power supply. The mic was set up about 1 meter from the speaker, and the output level was adjusted to 90dB SPL with pink noise.

We can get a flavor of what G_m can tell us by comparing it to harmonic distortion measurements at different levels and frequencies. For example, **Figure 8** shows the spectrum of a 1kHz sine wave, with a total harmonic distortion (THD) of 0.13%. Subjectively, it sounded quite clean, and the GedLee Metric is indeed well below the detection threshold of 1. Conversely, if we turn the speaker's limiters off and drive it hard in the bass (31.5Hz), the sound is quite buzzy and unpleasant. **Figure 9** shows the THD starting to rise to a bit more than 2%, not a particularly large number at those frequencies. But the GedLee Metric is 12.6, well into the "annoying" range and more accurately describes the perceived level of distortion.

Wow and Flutter

Though the golden days of vinyl are behind us, there's certainly a strong hobbyist and anachrophile

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attraction to the medium. Part of the attraction is the technical challenge of extracting high performance from dragging a rock through a piece of plastic. Old timers like me remember the days of wow and flutter (W&F) meters. And in a fit of delicious irony, we can now do W&F measurements using digital technology. Virtins has built this capability into MI with a vengeance—one can measure wow, flutter, wow and flutter, and drift, in both weighted and unweighted modes (see Resources). **Figure 10** shows unweighted (left) and weighted (right) measurements for a Technics SP10 Mk3 playing a JIS test tone (3kHz). Of course, you can view frequency and distortion spectra as well. Using the Chart Recorder option, one can even plot the change in frequency over time to determine rotational inconsistencies.

Loudspeaker Drive Measurements

I've been playing on and off with some ways to

characterize amplifier needs for driving speakers to a desired volume. Most speakers look nothing like a pure resistance, and even as reactances, there's some "wobble" seen under various drive conditions. I've been gathering some data on using real-world signals (music, voices) rather than test tones to characterize this and see, for a given application, what the amplifier will be asked to deliver in terms of both voltage and current at each point in the signal. With the Audio Precision APx1701 transducer interface, this is relatively easy to do, since that box contains the power amplifiers and a series resistance for current sensing.

To duplicate this capability, I set up a circuit shown in **Figure 11**. Voltage across the series sensing resistor is measured using a differential amplifier to remove the main signal across the speaker, which appears as common mode. This can then be converted to current. I chose a 1R resistor to make the math easier ($1A = 1V$). For a differential amplifier, I used a TI OPA1637EVL board because of the high common mode rejection (CMR) and low noise and distortion. The board was powered via a floating battery and the Linear Audio Silent Switcher power supply. I had some small speaker drivers from Stetron intended for smart speakers, so I used one of them as the test mule. The speaker was driven by an NAD M10 integrated amp, which was overkill, but that amp is small and portable, and I happened to have one nearby. **Photo 2** shows the test setup—it is not as elegant as the Audio Precision unit but, putting aside the unnecessarily expensive driving amplifier, it's much less expensive!

The voltage across the speaker was sent to Channel A, the output voltage of the differential amp (proportional to current) was sent to Channel B. I set up the MI in an X-Y recorder mode (Lissajous) with voltage on the X-axis and current on the Y-axis ($1mV = 1mA$); you can change the Y-axis label to milliamps (mA) directly if seeing "mV" bothers you and scale for the sensing resistor size.

Using white noise as a test signal, I got the trace shown in **Figure 12**, which has a rather galactic look to it. DDPs show the RMS voltage and current, and their ratio, which I termed "effective current." But still, you can see an area on the I-V plane that an amp is asked to supply, and this can be a useful measure for loudspeakers. Things get even more interesting when using voice and music signals, which I'll talk about further in coming months.

With MI's impedance measurement setup, a transducer's impedance vs. frequency is measured in a two-step process. With a setup like the one shown in Figure 11, the impedance

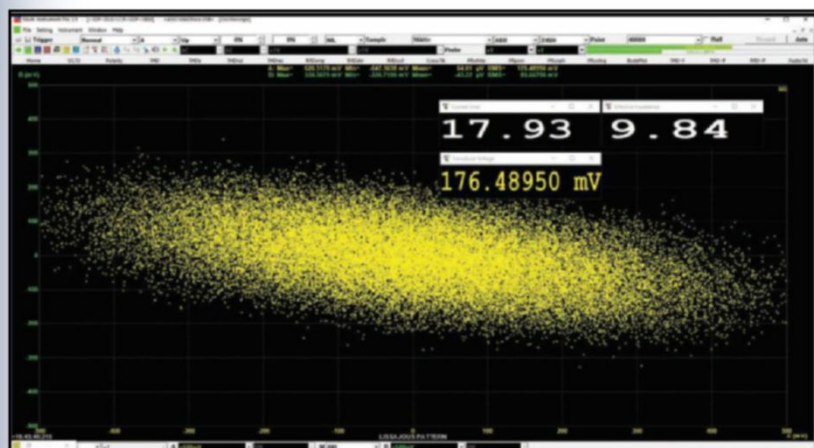


Figure 12: When the speaker is driven by a white noise signal, the synchronous current and voltage can be mapped out to determine amplifier needs.

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Virtins User Support Forum, <https://www.virtins.com/forum/index.php>

can be measured in one go using a Device Test Plan (**Figure 13**).

Final Thoughts

Everything technological gets smaller, faster, and less expensive. And measurement gear is no exception. What I have on my bench is about \$2,600 worth of equipment and software having noise and distortion capabilities that I could only dream of a few years ago. And that's being extravagant on the hardware side. I could get 95% of the performance and all of the versatility for half the cost.

As with my last article about the Virtins MI software, I feel that I've only scratched the surface with these basic measurement examples. I haven't mentioned the spectrogram, waterfall, speech intelligibility, cumulative spectral display, digital filtering, and a few dozen more things.

MI admittedly is complex and has a very steep learning curve. It can also do just about anything that you can conceive of for audio test and measurement. While the combination of MI and an interface is far fussier and inconvenient than, say, an Audio Precision analyzer, one can get similar or better results at the

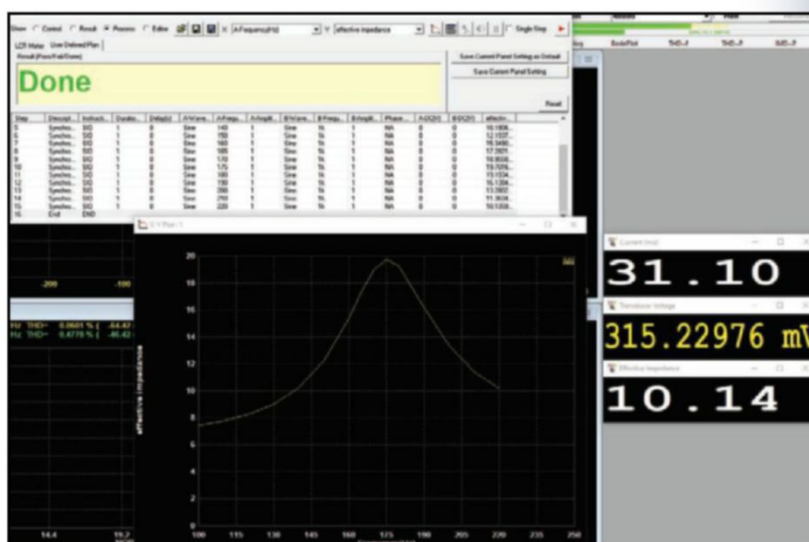


Figure 13: With simultaneous measurement of current and voltage, the impedance curve of a transducer can be obtained in a single step using a Device Test Plan.

expense of efficiency. Whatever it doesn't do out of the box can be programmed by the user, and as I've hinted, you'll see some interesting uses in my next couple of loudspeaker reviews. 

Author Acknowledgements: Besides the kind help and support of the folks at Virtins, I'd like to offer a hat-tip to John Jones and Raoul Jasselette for hints, goads, and generous sharing of data.

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Very Low Distortion, Greater Than 400mA Output Composite Amplifier Headphone Driver

By
Michael Steffes

In this article, Michael Steffes explains how a standalone amplifier, which already has very low audio band distortion, combined with a very low noise audio preamplifier, can lower this even further in a composite amplifier structure.

Several very high output current, dual current feedback amplifier (CFA) options have been available from the early ADSL line driver days. One of those (the THS6012, Reference 1) has been re-branded as the audio TPA6120A2 driver (Reference 2), with a much smaller package option (the 14 pin VQFN, RGY) made available at a lower cost. While using this as a standalone amplifier already provides very low audio band distortion, combining this with a recent very low noise audio preamplifier (the OPA1612, Reference 3) can lower this even further in a composite amplifier structure.

The need for very high current output differential twisted pair line drivers emerged in the late 1990s as the xDSL (Digital Subscriber Line) standards emerged. The earliest Central Office (CO) devices were higher voltage, independent, dual-channel, CFA designs like the THS6012. This core >100MHz CFA device can also find wide application as a high-power audio headphone driver—subsequently re-introduced in 2004 as the TPA6120A2. This later device is relatively sparse in specifications, but those can be mapped from the more thorough (and accurate) THS6012 device datasheet. Similarly, as there was no separate simulation model for the audio version, we can use the 2011 updated transistor level THS6012 model.

Inverting Composite Amplifier Design with an Imbedded TPA6120A2 Gain Stage

As is always the case when imbedding a power stage inside the loop, we would like that stage to be much faster than the targeted overall loop bandwidth for stability reasons. To reduce the swing requirements (slew rate and output current) from the core amplifier, target a gain of +5V/V in the power stage. Since this is a CFA-type amplifier, the bandwidth can always be

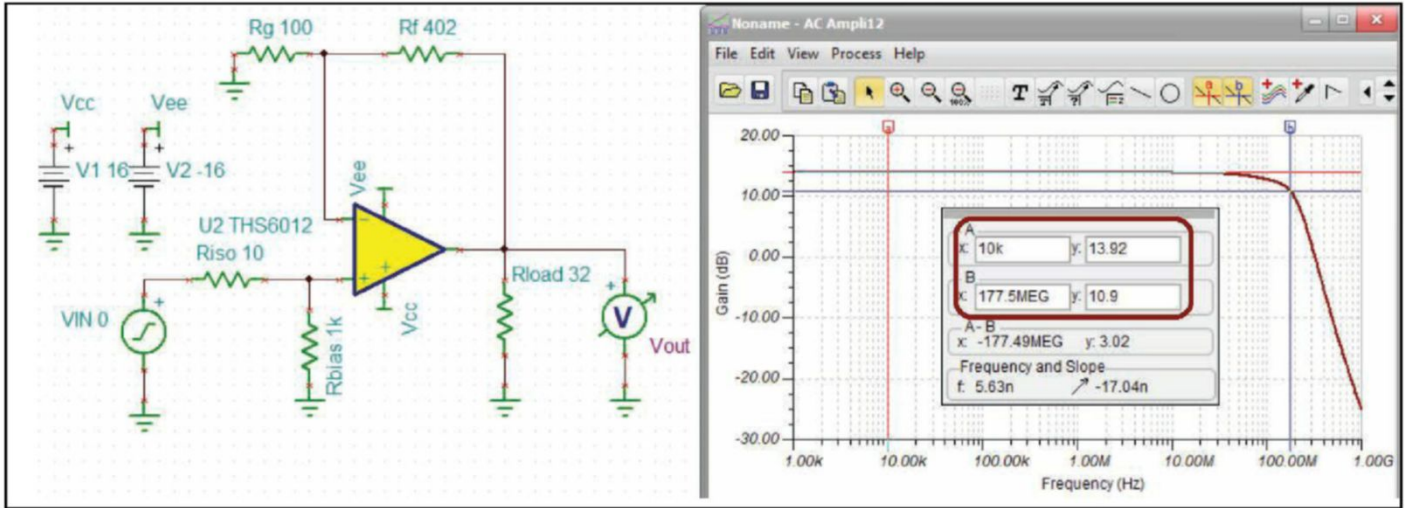
extended at higher gains by reducing the feedback resistor value. **Figure 1** shows the small signal bandwidth (SSBW) for the intended gain of +5V/V power stage using the THS6012 model. The ±16V supplies shown here are the maximum for this device where a ±12V swing is then available even into a 32Ω load (±375mA peak, here using the ±400mA minimum specification from the THS6012 vs. the ±700mA typical only specification from the TPA6120A2).

This very capable power stage can then be imbedded inside the loop of many different core op-amps. To get the lowest noise and distortion, use the dual OPA1612 “SoundPlus” device (Reference 3). Targeting an overall inverting gain of -10V/V, and including an estimated 0.6nsec package propagation delay for the power amplifier, gives the closed-loop response of **Figure 2**.

Imbedding this gain of +5V/V stage inside the loop for this simulated 92MHz gain bandwidth product (GBP) OPA1612 has effectively created a 5X92MHz = 460MHz decoupled voltage feedback amplifier (VFA).

This combination would not be stable at unity gain, but the inverting gain of -10V/V (noise gain = 11V/V) shows excellent stability with only 1.9dB peaking and 47MHz SSBW. No added compensation is apparently required where a loop gain simulation shows a LG=0dB crossover (Xover) at 24MHz with 48° phase margin. Theoretically, this 48° phase margin should produce 1.9dB peaking (Reference 4, Figure 2) exactly matching Figure 2.

This remarkable 47MHz SSBW at a gain of -10V/V suggests wider application at frequencies far exceeding the audio range. At some point, either the OPA1612 or TPA6120A2 will slew limit constraining the frequency range for a full scale ±12V output swing. Solving for that limit using the 1300V/μsec TPA6120A2 slew rate specification, and a 12V_{PEAK} maximum output, gives a maximum operating frequency of 17MHz. This



will never be reached as the much lower 27V/ μ sec OPA1612 slew rate will limit the maximum swing range to 1.8MHz using its lower maximum required output of 2.4V_{PEAK}. Lower swings of course can operate to higher frequencies or an alternate, higher slew rate, input op amp can be selected.

The OPA1612 is not only extremely low noise and distortion itself, but has a unique open loop response that includes a phase lead at higher frequencies. This feature (Reference 3, Figure 5) is very useful for overall stability in this composite structure. Recall, the transmission line delay in the simulation circuit of Figure 2 is not added externally—you will get that along with the package as an approximate delay. A 0.6nsec delay at the 24MHz LG Xover adds 5° phase shift in the loop. If the layout adds significantly more delay, the phase margin will decrease rapidly.

Compensation Options for this Inverting Composite Amplifier

A common next step for applying the composite circuit of Figure 2 would be to add a feedback bandlimiting capacitor. This will immediately create two sources of instability. The most obvious is that this capacitor will be shaping the noise gain to 1V/V at higher frequencies, putting the overall loop

(for this essentially decompensated op-amp) into oscillations. Fixing this first issue with a capacitor to ground on the inverting node, a second instability appears from those two series capacitors acting as a load to the output CFA stage. Both issues can be handled easily as shown in the 100kHz bandlimited modification of **Figure 3**.

The capacitor to ground on the inverting node essentially shapes the noise gain to be constant over frequency using capacitor ratios equal to the resistor ratios. This approach is a common inverting VFA design technique to apply decompensated op-amps over a wide range of gains (Reference 5). The small series resistor in series with the feedback capacitor acts to isolate the closed-loop TPA6120A2 power stage from what now looks like a capacitive load in the feedback path.

This resistor was only effective in a fairly narrow range of values. Going lower than 5.9 Ω brings in the output stage instability while going higher decreases the effectiveness of the overall noise gain shaping with the capacitor to ground on the inverting node. The design of Figure 3 gives an overall loop phase margin of 32° using the values shown with a closed loop 97kHz F-3dB. Again, the original composite circuit of Figure 2 appears stable

Figure 1: THS6012's gain of +5V/V with extended 178MHz SSBW using a reduced RF = 402 Ω

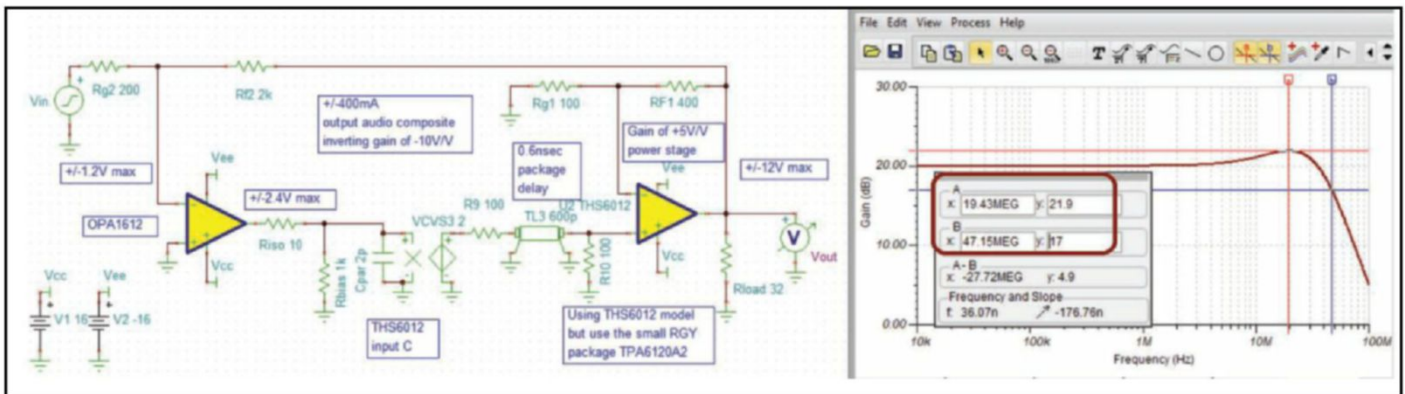


Figure 2: Inverting gain of -10V/V composite power amplifier

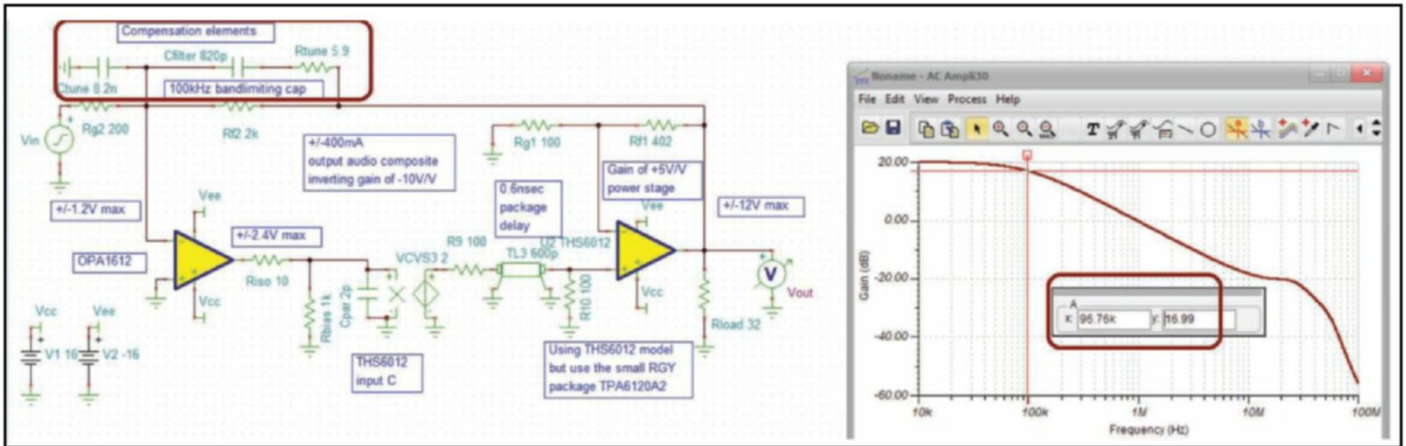


Figure 3: Bandlimited design to a 100kHz F-3dB with compensation elements

with no compensation—trying to bandlimit it with a typical feedback capacitor creates this potential for instability.

Expected Harmonic Distortion Improvement Using the Composite Structure

The TPA6120A2's already remarkably low distortion driving a high swing into a heavy load should be improved by the overall loop gain in this composite structure. However, it is very difficult to extract a useful large signal harmonic distortion number from the two datasheets that might apply for the output power stage design shown here. For instance, using the lower 402Ω feedback resistor for this CFA power amplifier will increase its loop gain and hence lower the output stage distortion—but no data

is shown for that condition. These exact operating conditions do not appear in either datasheet with the closest perhaps showing a worst case second harmonic of -70dBc at 1MHz for a 20V_{pp} swing into a 25Ω load (Reference 2, Figure 30). Estimating a 20dB loop gain improvement going from this 1MHz specification down to 100kHz would suggest the output CFA by itself can deliver <-90dBc distortion terms for this 20V_{pp} swing into a 25Ω load at 100kHz. From this already exceptional intrinsic output stage linearity, the composite loop gain should further reduce the actual harmonic distortion at the load. At 100kHz, the loop gain for Figure 2 or Figure 3 is 52dB suggesting this circuit might provide <-140dBc at 100kHz 20Vpp into a 32Ω load—even lower at audio frequencies since the composite LG continues up to 72dB at 10kHz.

Where the source can drive the relatively low input impedance for this inverting design, extremely low harmonic distortion driving high powers into low audio loads can be delivered—combining the best of recent audio amplifiers with the high current output legacy ADSL current feedback line drivers. The inverting design lends itself to simple compensation techniques simply using a feedback capacitor and a capacitor to ground to shape the noise gain at higher frequencies.

About the Author

Michael Steffes has worked since 1982 in high-speed signal path IC development spanning design roles for Comlinear Corp. (the original current feedback originator) with product definition and launch roles through National Semiconductor, BurrBrown, Texas Instruments, and Intersil. He continues to focus on High-Performance ADC interface solutions, DSL/PLC line driver design, and signal path optimization tools. Providing new product data sheet and applications collateral for more than 150 industry leading amplifier products has led to hundreds of contributed articles and application notes since 1988.



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The Science and Physics of High-End Mic-Preamplifier Design (Part 2)

Minimizing Noise and Distortion While Maximizing Headroom

By
Mike Meechan

In the first part of this article series, Mike Meechan considered the fundamental physics that play a primary role in microphone pre-amplifier design. Now he discusses its role in the real practical world and what the mic pre-amp must contend with to retain the fidelity of the mic signal along its path from transducer to output.

In live environments, retaining the fidelity of the mic signal can be an onerous task. Even with the prevalence now of intelligent lighting and LEDs to replace the industry standard PAR56 and 64 incandescent bulbs, powerful dimmers still emit RFI/EMI, with the situation worsened where unshielded cabling is used in the rig. It may also be the case that small-signal cables to and from the stage are run alongside speaker cables carrying heavy currents.

A Balanced Approach

Professional low-Z microphones use a three-wire connection—via an XLR connector or a jack plug—which carries ground and anti-phase signals from the source. Ideally then, (contaminating) magnetic fields induce equal voltages on both wires—and theoretically, these can then be nulled when applied to a transformer or differential amplifier (**Figure 1**). This can prove much more difficult than might at first be appreciated.

Using Transistors to Provide an Active Solution

The microphone's necessarily low Z_{SOURCE} —to minimize noise pickup—presents something of an engineering conundrum. Plus ça change. For optimum noise performance, it is imperative that R_{SOURCE} and the characteristic R_{NOISE} of the op-amp match optimally. However, the Optimum Source Impedance (OSI) of most op-amps is considerably higher than

the $2\text{k}\Omega$ load impedance required by the microphone capsule. (Industry practice is for the microphone capsule to hit a Z_{IN} at mid-band frequencies some 5 to 10 times its own. As such, it behaves as an unloaded voltage generator with little high-frequency roll-off). Likewise, realizing an op-amp's best noise performance occurs with R_{SOURCE} higher by an order of magnitude—to the extent that using an op-amp directly seriously compromises front-end noise performance by $\approx 10\text{dB}$ to 15dB . Thus, professional designs eschew this topology.

Two solutions find common usage: the first employs an input transformer (such as the acclaimed Jensen JT-16-A), which matches $Z_{\text{MICROPHONE}}$ to device OSI and provides galvanic isolation; the second (active) solution capitalizes on the better noise performance at low R_{SOURCE} of the transistor. A transistor/op-amp hybrid, with accurately matched transistors, and laser-trimmed resistor arrays, can offer common mode rejection ratios (CMRRs) at DC around 100dB (dependent on transistor matching and careful physical layout). With transistors, e_n and i_n alter in magnitude and in ratio to one another: lower I_C causes lower i_n —not inconceivably—since i_n is due to minor random discontinuities in device currents.

The ratio between the two—"noise impedance" (or OSI)—thus can be altered. Figure 3 from Part 1 of this article series shows the various plots of noise figure, i_n and e_n , against I_C for the BC560 low-noise transistor, a popular low-cost device. Others

including the 2SB737, the ZTX214C, the BC461, the MPSA55, or the 2N4401, find favor in many designs. Given identical measuring conditions, resistors of different values produce the same noise power—only the e_n and i_n ratios change, with the optimum value being the OSI—the resistance value at which the device is optimally quiet for audio purposes.

A further transistor noise component to consider is base i_n —for finite Z_{SOURCE} , this must be considered as a quadrature addition to e_n . (With a super-matched pair, base i_n is a very well-defined function of I_C). Fortunately, while i_n unlike e_n , increases proportionally to the square root of stage current, high i_n is of lesser importance when dealing with low-impedance sources such as microphones—making optimization unnecessary.

With bipolar transistors, the theoretical value for emitter-base e_n is a function only of absolute temperature and I_C :

$$e_n = KT \sqrt{\frac{2}{qI_C}} \quad \text{Equation 1}$$

Equation 1 indicates that voltage noise e_n can be reduced by increasing collector current, I_C . In practice, this is indeed borne out— I_C can be increased until parasitic transistor noise limits further reduction. This noise floor is usually created by, and modeled as, an equivalent resistor—the so-called “base-spreading resistance” (r_{bb})—in series with the base of the transistor. Consequently, low parasitic transistor noise is an important factor in ultra-low noise applications (e.g., microphone pre-amplifiers)—driving values of I_C are typically higher than normal. Even small values of series resistance unacceptably worsen noise performance.

Note: for very low R_{SOURCE} (e.g., a microphone), the r_{bb} of the transistor must be added to R_{SOURCE} in the calculation. However, OSI can be further lowered by reducing the ratio of the amplifier-inherent voltage and current noise. This is achieved by paralleling identical input devices, which maintains the same noise voltage, but alters the noise current. Proportionately, this changes the noise impedance. r_{bb} is reduced by a factor dependent upon the number of parallel devices. Noise reduces by the square root of the “paralleling” factor (e.g., paralleling three devices would reduce noise by $\sqrt{3}$). To achieve an equivalent reduction in voltage noise, similar arrangements can be applied to op-amps. Consider that while e_n is inversely proportional to the square root of stage current, shot (Schottky) noise contribution, conversely, can be reduced by maintaining high I_C reducing dynamic emitter resistance, r_e .

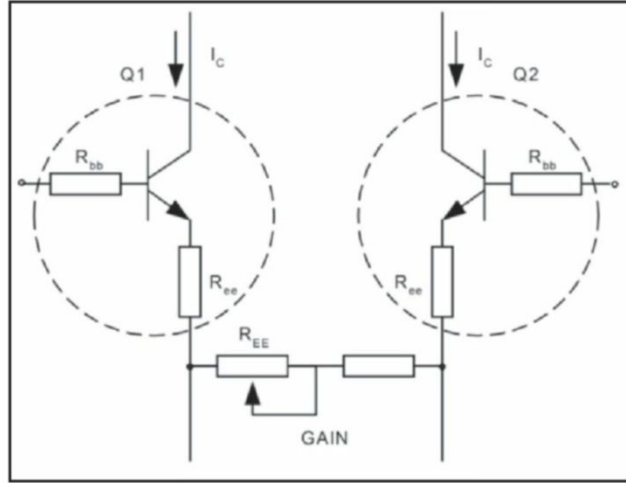


Figure 1: Real-world transistor model showing r_{ee} and r_{bb} (Source: Bateson)

This trend (specifically, e_n dropping and in rising as I_C increases) provides a (theoretically-simple) method of optimizing transistor noise for any given operating Z_{SOURCE} . Equation 1 illustrated that a “noiseless” source has an irreducible source of Johnson noise from the value of its R_{SOURCE} —any amplifier adds its own noise. And sadly there’s no getting around that.

Consequently, the amplifier’s noise voltage is added to the input signal while its i_n generates an e_n across Z_{SOURCE} . Since as far as audio frequencies are concerned the noise is uncorrelated, the square of the values are added. Good practice dictates picking I_C from the graph of e_n and i_n (vs. I_C): the $[e_n + (i_n R_s)]$ term thus is minimized for a given R_{SOURCE} . Here, with a $1k\Omega$ resistance, an I_C around $0.1mA$ appears to minimize the sum of voltage terms. So... to achieve a lower NF, prudence might suggest using a slightly lower I_C —after all, i_n is reducing at a greater rate than e_n is increasing. However, with a small-value I_{SOURCE} (e.g., a 150Ω to 200Ω microphone), low e_n is considerably more important.

In practice, therefore, we operate the transistor at a higher I_C than may superficially first seem optimal: $2mA$ proves to give the best all round performance for this device. Where r_{bb} is known, e_n for device operation at a particular I_C can be calculated as follows:

$$e_n = \sqrt{4KT r_{bb} + \frac{2(kT)^2}{qI_C} V \sqrt{Hz}} \quad \text{Equation 2}$$

The graph depicted in Figure 3 from Part 1, which shows narrowband NF vs. I_C , and NF vs. frequency, bears a value around $1.5dB$. This is acceptable: the enticing diktat for high I_C ($1mA$ to $5mA$) is one many discrete and integrated designs follow, the rationale being that in can be sacrificed somewhat so that lower e_n , which as shown earlier is particularly important with low- Z impedances (microphones!) can be achieved.

However, most mic preamplifiers—certainly those designed using discrete components—are optimized for noise, not distortion performance. Just as r_{bb} adds itself to the value of R_{SOURCE} and worsens noise performance, so the other resistance

inherent in the transistor makeup, R_e , the intrinsic emitter resistance, worsens distortion performance, and R_e is a function of collector current, being quantified as follows:

$$R_e = 1/25I_C \quad \text{Equation 3}$$

(where I_C is the collector current in milliamps)

Varying Topologies

As always, it is a case of compromising and conflicting operating conditions, with the designer presented with something of a dilemma. Low I_C reduces this intrinsic emitter resistor-induced distortion but worsens noise performance. High I_C reduces noise but worsens distortion. Practically, it's simply a question of carefully choosing the active devices (and compromising the noise/distortion performance for a given set of conditions).

Figure 2 shows a typical example of this: Q1/Q2 form a super-matched beta pair operated in differential, long-tailed pair configuration, which provides Z_{SOURCE} matching/initial amplification of the mic signal. The super-matched pair, close-tolerance resistors, and a differential-style output to the op-amp, meaning that noise and CMR are excellent. The op-amp provides the remainder of the amplification necessary (and a single-ended output).

Topologies like this—given consideration to careful design—can achieve excellent performance and 0dB to 70dB gain range controlled via a single (anti-log) potentiometer, which is preferred by some people in a concert-type environment (or better still, a multipole switch with precision resistor values connected to each pole, cascaded to a fine-tuning ± 10 dB ultra-low-noise op-amp outputs stage).

Each transistor acts as a voltage-controlled current sink, or “transconductance stage,” transformed into a voltage amplifier by the addition of the collector resistor. The op-amp performs as a current-to-voltage converter, and since it holds each of the collector voltages constant, both behave as true transconductance devices. Since $I_{E1} = I_{E2}$, we control I_C with this emitter resistor. Collector resistor values are chosen such that the V_{ce} maximum voltage isn't exceeded, and also that the transistor operates as near 0V as possible—making a full rail-to-rail output swing available. An important caveat is that high level common-mode signals are amplified fully by the transistors before being nulled by the following stage—consequently, the maximum V_{ce} rating of the transistor can be exceeded (and the device sometimes destroyed or operating headroom seriously eroded, should high CM signals be present. This applies equally to integrated designs).

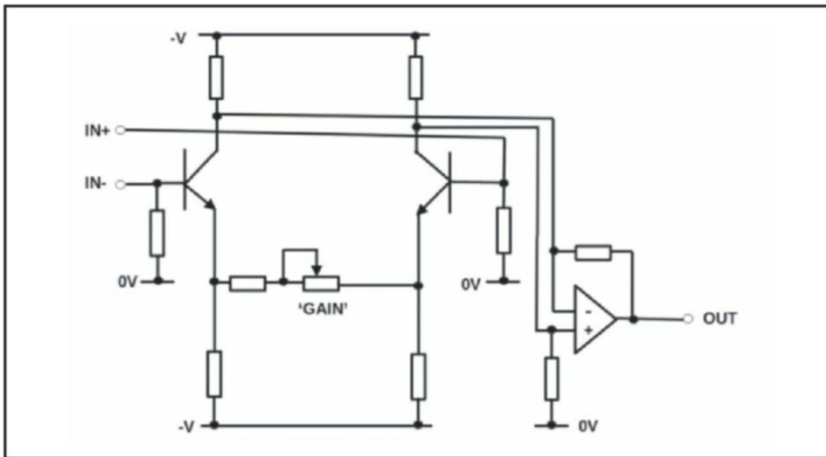


Figure 2: Microphone differential preamplifier with transistor front end

Device	LM194/394	SSM2212	MAT03 (PNP)	MAT320 (PNP)	THAT320 (PNP)
Noise Voltage Density	1.8nV/√Hz	0.85nV/√Hz	1nV/√Hz	0.9nV/√Hz	0.8nV/√Hz
Gain Bandwidth Product	Unspecified	200MHz	190MHz	325MHz	325MHz
Bulk Resistance R_{ee}	0.4Ω	≈0.3Ω	0.3Ω	2Ω	≈2Ω
Base Spreading Resistance R_{bb}	25Ω	≈28Ω	25Ω	32Ω	32Ω

Table 1: Typical specifications of various super-matched low-noise transistor pairs

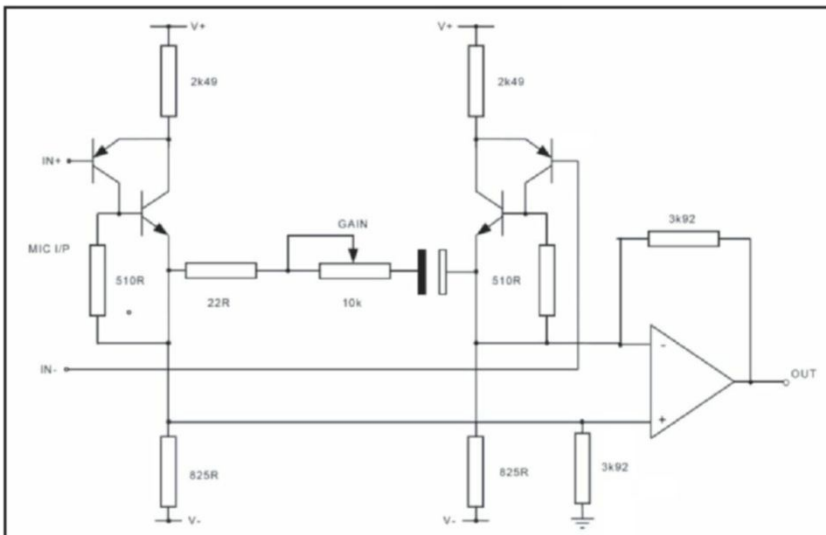
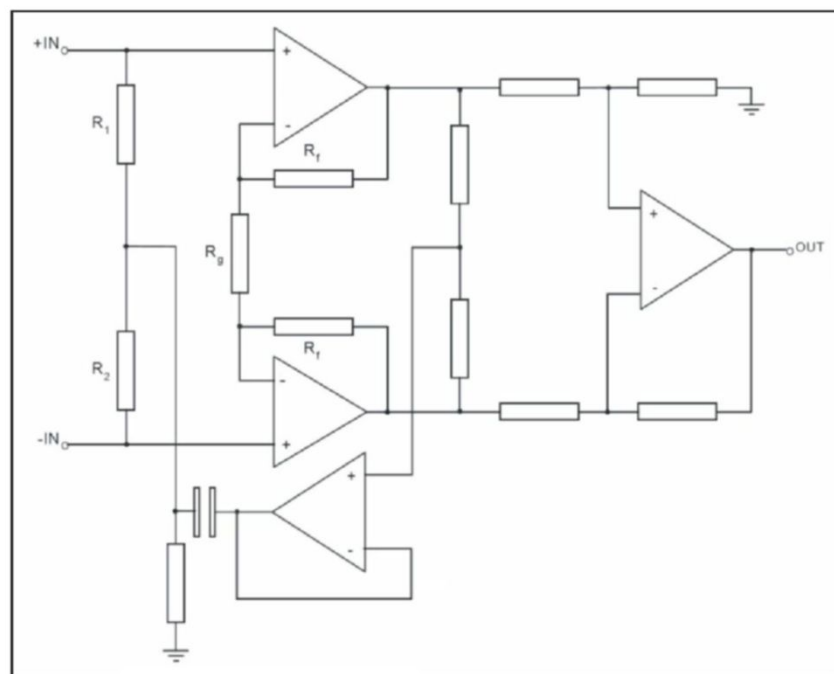


Figure 3: Sziklai pair (as employed in the legacy Mackie 1202VLZ4 mixer) used here to reduce distortion (Source: Bateson)

Where there exists the cost justification of implementing more expensive components, the dramatic increase in performance to be had from dedicated, low-noise super-matched transistor pairs (e.g., Texas Instruments' LM394/194, Analog Devices' SSM2212, or THAT Corp.'s super-matched NPN low-noise audio transistor pairs) more than compensates for the (slight) cost penalty incurred.

Additionally, a complementary package—the THAT-320—is also available as a PNP-type device (as is Analog Devices' MAT-03). PNP types are often preferred because of their better low-frequency performance figures and typical lower r_{bb} figures compared to NPN devices. A further refinement of the basic topology might use the input-device-paralleling topology outlined earlier. **Table 1** shows the typical specifications for various super-matched transistor pairs.

Discrete transistor designs use virtually-identical configurations, but regrettably have an electrolytic in series with R_{EE} , the gain-controlling resistor between the two emitters (super-matched pairs don't suffer from this problem, so no capacitor is required). Large capacitance values are necessary so that gain at DC—not Audio Frequencies (AF)—is



rolled off (small characteristic inherent mismatches of the discrete transistors generate DC offsets, which would otherwise be amplified up by the full gain of the configuration).

Unfortunately, transistor mismatch can occur in one of two ways—consequently, there is always a small and degrading voltage across the electrolytic,

Figure 4: Bootstrapping input resistors improves CMR performance (Source: Whitlock)

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which is in the signal path. Using non-polarized electrolytics, (a back-to-back pair from the Nichicon MUSE audio range is a good choice)—complete with R_{BIAS} —effects cure.

A modification to the more-conventional two-transistor design appeared first in Dove's very comprehensive treatment (1991) of pro-audio

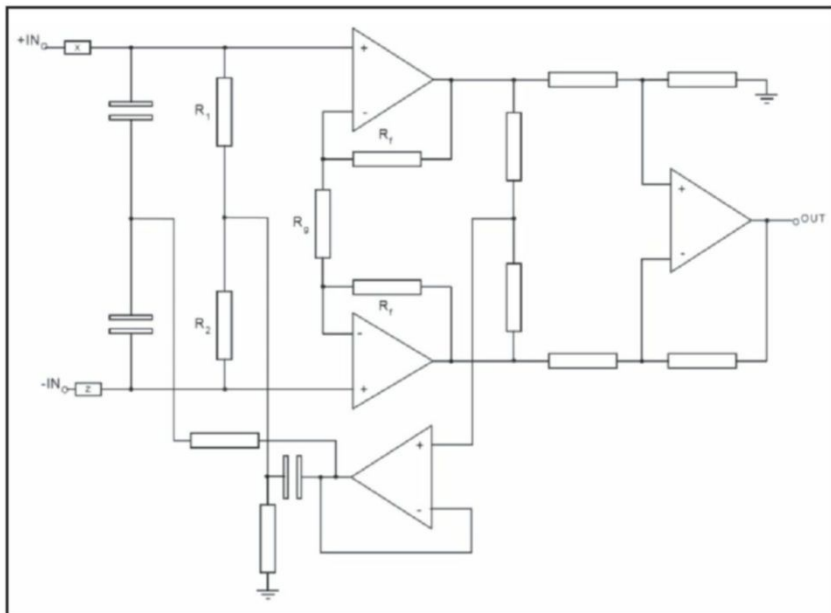


Figure 5: Bootstrapping RFI filtering capacitors improves RFI performance (Source: Whitlock)

About the Author

Mike Meechan studied electrical/electronic engineering in Glasgow and has both BS and MS degrees. Until 1996, he worked as senior engineer at the BBC's World Service, Bush House, London, then at BBC Monitoring. A period as a freelance designer/author in the professional audio field followed, and he founded Audio Elixir, (an audio design house specializing in low-noise pro-audio analog solutions). From 1998-2002, he was a documentation manager for a large organization providing end-to-end digital broadcast solutions, to BSkyB, London, UPC EuroHITS, Amsterdam and Prague, and DirecTV, Los Angeles. For 10 years he was chief architect and led the Motorsport Knowledge Exchange project for Oxford Brookes University (School of Technology). Since 2011, he's been providing audio engineering consultancy to the industry. Over a period of 30 years, Meechan has authored 80 articles/papers on pro-audio/broadcast-related matters, and designed, built, and published numerous acclaimed pro-audio products. His research interests include low-noise audio design, particularly pre-amps, EQ, and dynamics processing. He previously held memberships to the Audio Engineering Society, the Institute of Engineers & Technologists, and PLASA.



mixer design. His topology used four transistors—as opposed to the more-usual two—with each transistor pair configured in a complementary Darlington (or Sziklai) configuration, which behaved as a super-gain transistor (**Figure 3**).

In a novel manifestation, NPN transistors carry most of the current. Input stage current is set virtually-constant by the 510Ω resistors and the V_{BE} of the two NPNs. Constant current through the input devices means that r_e isn't modulated at high gains and signal levels. Most stage current is carried by the NPN transistors through R_{EE} and into the op-amp—distortion consequently is almost eliminated. In contemporary designs, the THAT 340 quad low-noise NPN/PNP transistor array would lend itself readily to this configuration.

Figure 4 shows an interesting solution to the problem of CMR. Rather than have the input impedance-setting resistors tied to ground, their mid-point is bootstrapped by a further op-amp driven as shown by the common-mode voltage outputted from the first stages ($R3/R4$ node). To AC common-mode signals, the circuit's input impedances are ≈ 1000 or more times the value of $R1$ and $R2$. However, to differential signals, they present their normal values. Furthermore, the technique can be extended to improve RFI suppression (**Figure 5**).

These patented modifications (U.S. Patent 5,568,561) elevate the CMR properties of the active input stage to be almost that of the transformer, (typically 65dB CMRR at 20kHz, even with unmatched source impedances), and while complete isolation remains a domain in which the transformer is as yet unchallenged, some commentators dispute this. RFI can be further suppressed using common-mode chokes (typically bifilar-wound pulse transformers) which block RFI but pass audio high-frequency as necessary. Bootstrapping a floating power supply with the CM signal, and then using the floating power supply to power the two input devices, can improve CMR. And I know that I have conveniently glossed over transformer mic preamps—but that will have to be left for a (future) article.

Next month, we look at building some of the (theoretically) perfect active mic preamp. Seeking the Philosopher's Stone indeed.

Resources

S. Bateson, "Design for Mixing: The Mackie 1202 Mixer," *Audio Media*, pp. 78-82, AM Publishing Ltd., June 1993.

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M. Meechan, "Microphone Preamps – Performance Needn't be Sacrificed for Economy," *Audio Media*, pp. 80-91, AM Publishing Ltd., February 1996.

B. Whitlock, "A New Balanced Audio Input Circuit for Maximum Common-mode Rejection in Real-world Environments," Proceedings of the 101st Convention of the Audio Engineering Society, New York, 1996.

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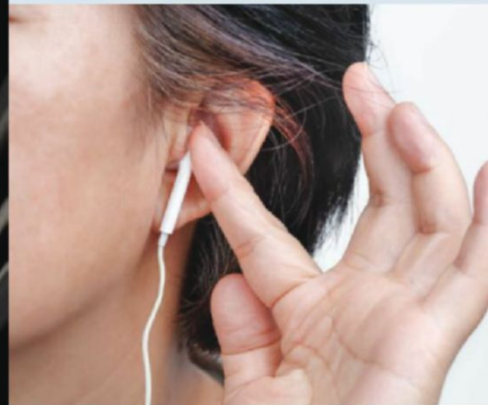
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Clever Engineering

One of the problems when designing large audio devices, (e.g., mixing consoles) is the enormous amount of wiring that's needed. Wire is expensive, and so is hand soldering all those connections. In this article, Ethan Winer shows how electronic engineers can reduce the number of wires using rotary encoders, serial communications and other solutions.

By
Ethan Winer

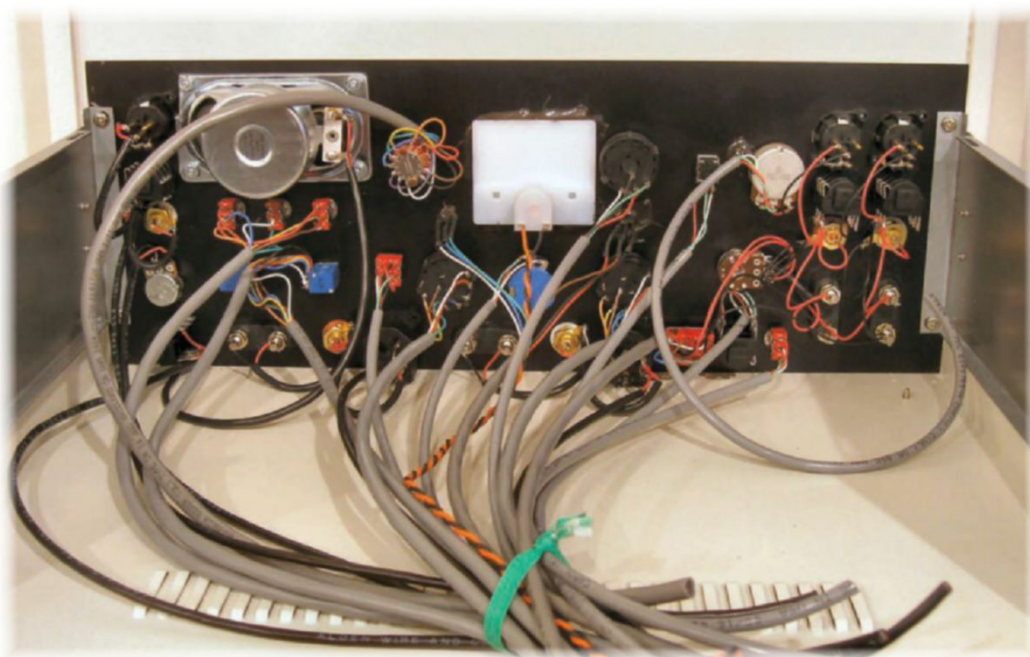


Photo 1: It took many hours to hand-wire all of these switches, potentiometers, and other components!

When it comes to wiring techniques, first up is the BCD rotary switch, where BCD stands for Binary Coded Decimal. **Photo 1** shows the rear of an audio test device that I built in 2018, and mentioned in the June 2019 issue of *audioXpress* magazine. The rotary switch to the left of the white VU meter in the middle of the panel has 11 positions to adjust amplifier gain from 0 through 100dB. So a total of 12 wires are needed, including the common connection. If I had used a BCD switch, only five wires would be needed.

BCD Switches

Photo 2 shows the construction of a conventional three-position rotary switch. In this rear view, the terminal at left is called the Common, and it connects via the metal slider one at a time to the other three terminals. Here, the switch is in the middle #2 position. In contrast, a BCD switch outputs a binary number that identifies the current switch position. Using binary numbers accommodates up to 16 positions, using only four output terminals plus the common. **Table 1** shows the 16 values 0-15 that can be represented by four binary digits.

So instead of a common contact that connects alternately to terminals 1, 2, 3, and so forth, the common contact in a BCD switch is typically connected to the power supply. The supply voltage

appears in 16 unique combinations at the switch's four output terminals, and an inexpensive logic IC then decodes the output number and sends it on to the host device to process. So using a BCD switch when appropriate can reduce the high cost of human labor to solder the switch terminals and additional wire connectors. It can also save on the cost of expensive copper when the wire runs are long. One example of long wire runs is physically large recording consoles that can have as many as 80 inputs or even more.

Figure 1 shows an exploded view of a BCD rotary switch, and **Figure 2** shows the equivalent schematic. In Figure 2, the switch drawing is viewed from the front and is shown in switch position number 1, which is decimal number 0 and BCD number 0. The solid circles have a terminal present, where the hollow circles do not.

Serial Communications

As useful as BCD coding is for switches, an even more efficient method to minimize wiring is to use serial communications. With this method, digital data bits are sent one after the other down a single pair of wires. In the early days of personal computers, parallel printers were connected using 25-pin connectors having separate wires for each

of the 8 bits of data being sent. Other wires served such tasks as handshaking, so the printer could tell the computer to pause sending until the printer was ready to receive more data. Back then memory was very expensive so printers could only hold a page or two of text in their receive buffer.

Likewise, computer sound cards back then were plugged into multi-pin connectors on the motherboard to transfer 16 bits of data on 16 separate connections. Today printers and sound cards, and many other devices (e.g., scanners and external hard drives) connect using USB, which is also a serial protocol. Of course, serial communication requires a much faster transmission rate than parallel because the bits are sent one after the other rather than all at once. Plus, the devices at both ends need additional circuitry to encode and decode the serial data stream. So BCD is still useful for simple passive components (e.g., switches).

Rotary and Up/Down Encoders

From the user’s perspective, a rotary encoder is similar to a rotary switch. But instead of a fixed number of positions, the knob continuously rotates. An incremental rotary encoder merely outputs a series of pulses as the knob is turned, where the absolute type of encoder outputs the current knob position. There are many sub-types within these broad categories, and also several construction materials (e.g., mechanical, magnetic, and optical). So I’ll just cover the basics. The main feature of rotary encoders is that they can accommodate dozens or even hundreds of positions efficiently, and with only a few connecting wires. Up/Down encoders serve a similar function, but use a pair of push button switches instead of a rotating knob.

The only thing an incremental encoder needs to do is output a signal to show that the knob is being turned, and also indicate the direction of rotation. Incremental encoders are common in applications where there are too many positions for a conventional rotary switch to handle. For example, most modern hi-fi and home theater receivers control the volume digitally, typically with 90 or more steps in 1dB increments.

Years ago, I owned a Yamaha SY-77 synthesizer containing more than 100 musical instrument sounds. That too used a rotary encoder to scroll through all of the patch names on a small LCD display. It’s not practical to build a switch with that many positions, so the knob of a rotary encoder will complete several rotations as you adjust the volume over a large range. But unlike a rotary switch, when the power is turned off then on, the previous setting is not available. So it’s up to the receiver or



Photo 2: This conventional rotary switch has three positions. The Common terminal at left connects alternately to the other three contacts. Note the small detents in the switch’s metal frame at bottom. The springy metal attached to the center knob shaft clicks into place for each fixed position.

Decimal	Binary
0	0000
1	0001
2	0010
3	0011
4	0100
5	0101
6	0110
7	0111
8	1000
9	1001
10	1010
11	1011
12	1100
13	1101
14	1110
15	1111

Table 1: Binary math uses only two digits, so 1 comes after 0, but then 10 comes after 1, followed by 11, then 100, and so forth as shown.

other device to remember the last volume setting. Likewise for Up/Down switches. In contrast, the current position of an absolute encoder is always available. Like the BCD switches described earlier, absolute rotary encoders also output a binary number to indicate a large number of positions with fewer connections.

When many positions are used, the host software can optionally be written to detect if you’re quickly spinning the dial, for example to increment by tens instead of single digits. Up/Down switches can be handled the same way when there are many possible steps. To set the Clock on my kitchen oven I press Clock, then the Up or Down buttons that normally set the oven temperature. Pressing the Up button once increments the time by one minute. Pressing and holding the button for one second advances the time fairly quickly. Continuing to press for two

more seconds advances the time much more rapidly. I realize this is pretty basic stuff, but writing the software that reads these encoders and buttons and changes the timing as described is fairly complex.

Matrix Addressing

Another clever example that reduces the cost of long wire runs is found inside pinball machines. Most games have a playfield containing many dozens of lights, and just as many switches, all which must be controlled and read independently by the computer that manages play. Each bulb or switch has two contacts, and with conventional wiring all of the common grounds are connected. So one ground wire can snake through all the light bulb sockets connecting their ground side, and another ground wire can serve all the switches. But that still leaves a lot of separate wires needed for all of the switch and light bulb “hot” terminals. Even more relevant, computer chips don’t have 100 or more extra output terminals to control that many lights and switches individually. With matrix addressing, as many as

64 light bulbs can be controlled by only 16 signal wires, and up to 64 switches can be read the same way.

Instead of running separate wires from the computer controller to every switch and light bulb, a matrix system of Rows and Columns is used. **Figure 3** shows a portion of the schematic for my 1980s vintage Fire Power pinball machine. This section details the matrix that controls all of the light bulbs, and another similar section (not shown) reads all of the switches. To address the light bulbs individually, the rows and columns are strobed (activated in sequence) with only one row and one column active for a few milliseconds at a time. When several bulbs are to illuminate, each is activated in sequence, but this happens so quickly that all the bulbs appear to be illuminated at once. Diodes are placed in series with each bulb to avoid interaction between the bulbs.

Reading a matrix of switches is similar, where the CPU continuously polls each switch very quickly in sequence. Even with all the additional electronics



Figure 1: This exploded view shows the construction of a 16-position BCD rotary switch. The ball bearings click into fixed positions in the slots inside the tube at bottom left. (Source: Grayhill, Inc.)

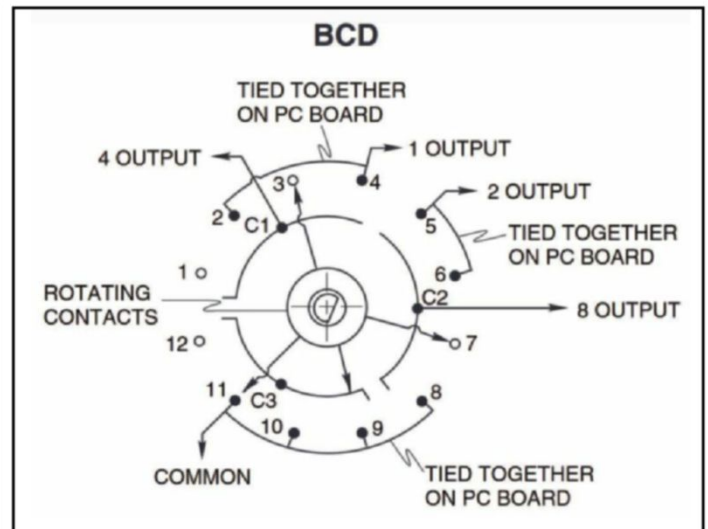
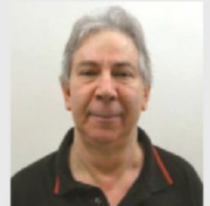


Figure 2: This schematic details the internal wiring of the BCD switch shown in Figure 1. (Source: Grayhill, Inc.)

About the Author

Ethan Winer has been an audio engineer and professional musician for more than 50 years. His Cello Rondo music video has received nearly 2 million views on YouTube and other websites, and his book *The Audio Expert* published by Focal Press and now in its second edition, is available at: www.amazon.com and on Ethan's own website ethanwiner.com. Ethan is also a principal at RealTraps [<http://realtraps.com>], a US manufacturer of high-quality acoustic treatment.



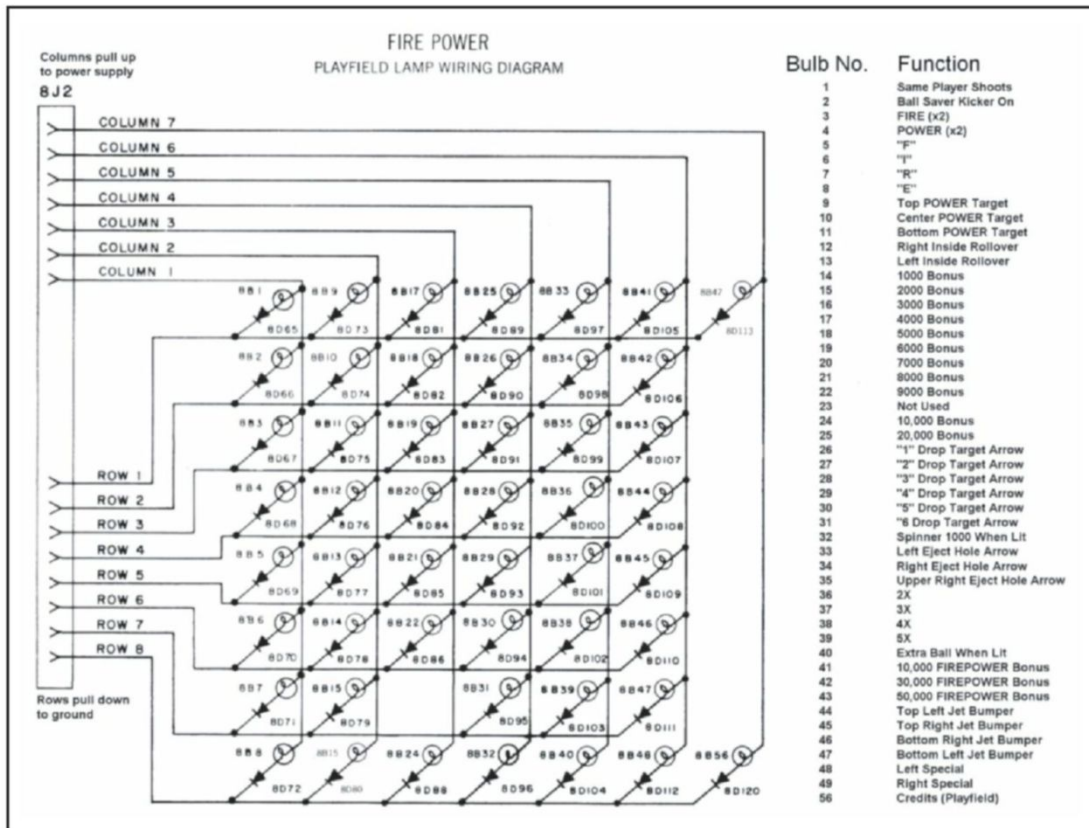


Figure 3: The Fire Power pinball light bulb matrix can control up to 64 bulbs using only 16 wires. Since the active bulbs are illuminated only briefly, in sequence, they're powered by 18V instead of the 6V for which they're normally rated. So the average power dissipation is still safely within their normal operating range.

needed to implement matrix addressing, it's still less expensive to manufacture than using the additional wire, connectors, hand labor, and computer controller chips. Two of my pinball machines are 40 years old, and sometimes the circuit board connectors become flaky. One side benefit of the matrix is that when one entire row or column goes out on my machines, it's easy to tell which circuit board connector pin needs to be pushed in more tightly.

Photo 3 shows the lower cabinet interior of Fire Power, with the playfield lifted to see its underside. The wire bundles at the bottom of the photo are coming down from the power supply and driver board inside the upper cabinet behind the painted glass. These wires connect to all the light bulbs and switches on the playfield, and to other electromechanical components such as the flippers and ball kickers. Without matrix addressing even more wires would be needed, and those wire bundles would be three times thicker. 

Resource

E. Winer, "Building a Null Tester Device," *audioXpress*, June 2019, <https://audioxpress.com/article/you-can-diy-building-a-null-tester-device>

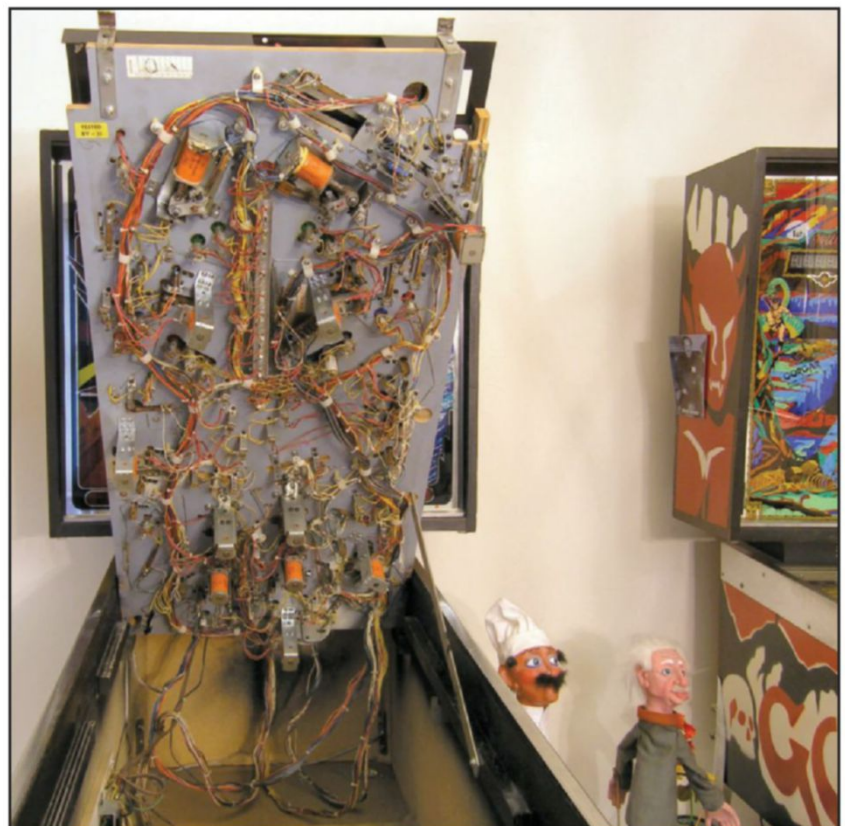


Photo 3: Fire Power pinball machine with the "hood" lifted to show its internal wiring.

X-Altra Moving Coil/Moving Magnet RIAA EQ Preamplifier (Part 2)

MM Preamp, System Gain Amplifier, Filter, PSU, and Measurements



In the second part of this article, the author details the Moving Magnet EQ stage, system gain amplifier and filters, along with the system's power supply unit for the X-Altra design. The article concludes with some measurements of the X-Altra preamplifier.

By
Andrew C. Russell

For the second half of this article, we will begin with the Moving Magnet (MM) EQ amplifier design (**Figure 1**). The junction field effect transistor (JFET) plus op-amp topology used in the X-Altra MM preamp stage borrows heavily from Denis Colin's excellent LP797 design published in *audioXpress* in 2007 but in this implementation, all active EQ is applied around the first stage rather than Colin's active/passive split EQ architecture.

The key gain element is U21, a dual ultra-low noise N-channel JET from Linear Systems, the LSK389B featuring 0.9nV/√Hz typical (1.9nV/√Hz maximum) input noise voltage. By paralleling the two devices, the equivalent input noise of the JFETs is reduced by 1/√2 to 0.63nV/√Hz typical. In the LP797, the JFET source resistor was set to 10Ω, which contributed just 410 pV/√Hz noise (57nV RMS over the audio band). In an all active EQ arrangement such as the X-Altra, if the gain resistor is set much below 100Ω (191nV RMS wideband noise contribution), the feedback equalization components quickly become large and unwieldy. At least 1% tolerance capacitors will be required for acceptable

EQ accuracy, which are expensive at high values such as 0.47μF or 0.1μF. These are likely to be wound film types, so they will be large and prone to noise pickup. With the gain resistor set to 110 Ω, the EQ capacitors are readily available in 1% tolerance NPO/COG surface mount device (SMD) types and they are inexpensive.

By using a JFET front end in this MM amplifier, the current noise $\times R_{\text{SOURCE}}$ term discussed in Part I (*audioXpress*, February 2021) is effectively removed (Part 1, Figure 2) and results in a >3dB SNR improvement using the standard cartridge compared to a NE5534A op-amp.

The MM signal is fed from the source selection switch U25. R79 and R81 in conjunction with U22 DIP switch (left moving magnet load or LMML) enable the cartridge loading resistance to be set to either 26kΩ, 31kΩ, 35kΩ, or 47kΩ allowing the high-frequency response to be tailored to minimize response peaking due to the interaction of cable capacitance and cartridge inductance, and separately, tip mechanical resonance.

Note, to use this feature successfully, you will

ideally need a test record and an oscilloscope. Failing this, the best solution is to simply set the loading by ear for the best sound. R80 provides a 47k Ω cartridge load (U22 switches open) and the bias for the JFET input pair. R74 and R75 at 110 Ω combined resistance sets the self-bias JFET I_d at nominally 1.5mA per device, 3mA in total, though I_{ds} vs. V_{gs} spreads will mean this figure can vary by as much as 30% to 40%.

The output signal is developed across R82 (1k Ω) and the drain coupled to the input of one half of U26, an LM4562, via a 2.2 μ F film capacitor C90. The LM4562 provides the bulk of the open-loop gain and drives the EQ feedback network R71-R73 and C83-C85. R76 and C86 along with R77 and C87 provide loop compensation. The output secondary post filter (R58 and C63) -3dB bandwidth is set to 220kHz and provides just enough correction at 20kHz to pull the simulated EQ conformance to within ± 0.15 dB over the audio band.

To arrive at the specific values, I used the formulas from Stanley Lipshitz's 1979 "On RIAA Equalization Networks" paper in a spreadsheet and then optimized the secondary post filter in LTspice to ensure accurate conformance. R73 (220 Ω) in the feedback network is also determined during the LTspice optimization and helps to flatten any residual RIAA conformance deviation in the mid band.

As with the MC front end preamplifier discussed in Part 1 of the article, the MM front end stage power supply rejection ratio (PSRR) is essentially zero. To remedy this, the other half of U26 is used to implement a very low noise 10V regulator, the reference input on pin 3 being derived from the resistor divider chain used in the MC voltage regulator circuit (Part 1, Figure 5). R78 and C88 provide further noise filtering prior to feeding the front end JFET drain load resistor R82, where the regulators simulated noise output is 50pV/ $\sqrt{\text{Hz}}$ (1kHz) or about 7nV RMS over the audio band.

An LTspice model used to investigate the phono amp's performance showed that the 1kHz input referred spot noise with the input shorted was just under 1.6nV/ $\sqrt{\text{Hz}}$, most of this attributable to the 110 Ω source resistor. By way of contrast, Colin's LP797 input referred noise floor was under 1nV/ $\sqrt{\text{Hz}}$. However, with the standard cartridge connected to the input, the simulated noise figure was 5.4nV/ $\sqrt{\text{Hz}}$, just 0.2nV higher than the LP797 despite the 11x higher source resistor used on the X-Altra.

The overload margin, measured by driving the input at 1kHz to the onset of clipping at the output of the preamplifier (U15), is 189mV peak at 1kHz or, reference a 5mV input, 31.5dB with the system gain set to 0dB. Since this is an all-active feedback

design, the 31.5dB figure holds over the whole audio band from < 20Hz through to > 50kHz. The DC offset at the output of U26 pin 7 is as much 70mV due to U26's bias current flowing through R76, necessitating DC blocking capacitor C62.

System Gain Amplifier and High-Pass Filter

The system gain amplifier is configured around U27, an OPA1641 JFET input op-amp. The gain may be adjusted to 0dB, 6dB and 9.5dB via DIP switch U16 left gain system (LSG). This stage provides additional signal chain gain flexibility in the case of lower output cartridges—both MC and MM. Note that in this position, and in U7, U17, and U18, a lower-cost NE5534A op-amp can be used, in which case a 33pF NPO/COG 0805 capacitor should be fitted (C95, C96 for the left channel and C5 and C6 for the right channel).

The system gain stage feeds into a conventional Sallen-Key high-pass filter constructed around U17. The 20Hz high-pass filter is permanently enabled, which is the case when J2 (45Hz high-pass filter) is closed and completely removes the problem of bass speaker "flap" due to record warp. When the switch, located on the rear panel, is opened, the turnover frequency moves up to 45Hz as the total filter capacitance (C64-C67 and C72 and C73) is reduced from 141nF to 47nF. Many older classical recordings (vinyl and their CD re-releases) suffer from an annoying "acoustic" rumble, which is particularly apparent if a sub bass is used. It has been suggested the cause is air conditioning and/or low-frequency air current noise being picked up by the sensitive microphones in large recording spaces. The 45Hz filter reduces this noise significantly without really affecting the music's bass "weight." With organ music, you might lose notes well below 45Hz, but if this is a concern, you can simply switch the filter out.

The output from the filter stage is routed through C61, a 22 μ F bipolar capacitor, R57 (47 Ω) to U15, the preamplifier output connector. A 3.3nF capacitor provides additional radio frequency interference (RFI) noise protection.

About the Author

Andrew C. Russell worked for the first half his career in the electronics industry in product development, qualifications and marketing. During the second half of his career, Russell was in business management and business development in the semiconductor industry working for a global MNC, spending the last 10 years based in Japan, Taiwan, and China. He struck out on his own in 2015, founding Ovation High Fidelity and doing freelance audio design consulting and contracting with an emphasis on power amplifiers and preamplifiers. When not working in his lab, he can be found walking his dog or enjoying his vinyl collection.



System's Power Supply

The power supply (**Figure 2**) uses a 3.5VA 2× 18VAC secondary Nuvotem Talema PCB mount toroid (marketed in the US under the “Acme” and now “Amgis” brand names from Digi-Key). Figure 7

from Part 1 shows the MC performance with $R_{gen} = 47\Omega$ and despite the power supply unit (PSU) being co-located with the amplifier circuit, and the very high gains involved, mains noise measured at the preamp output remains below -90dBV. Each

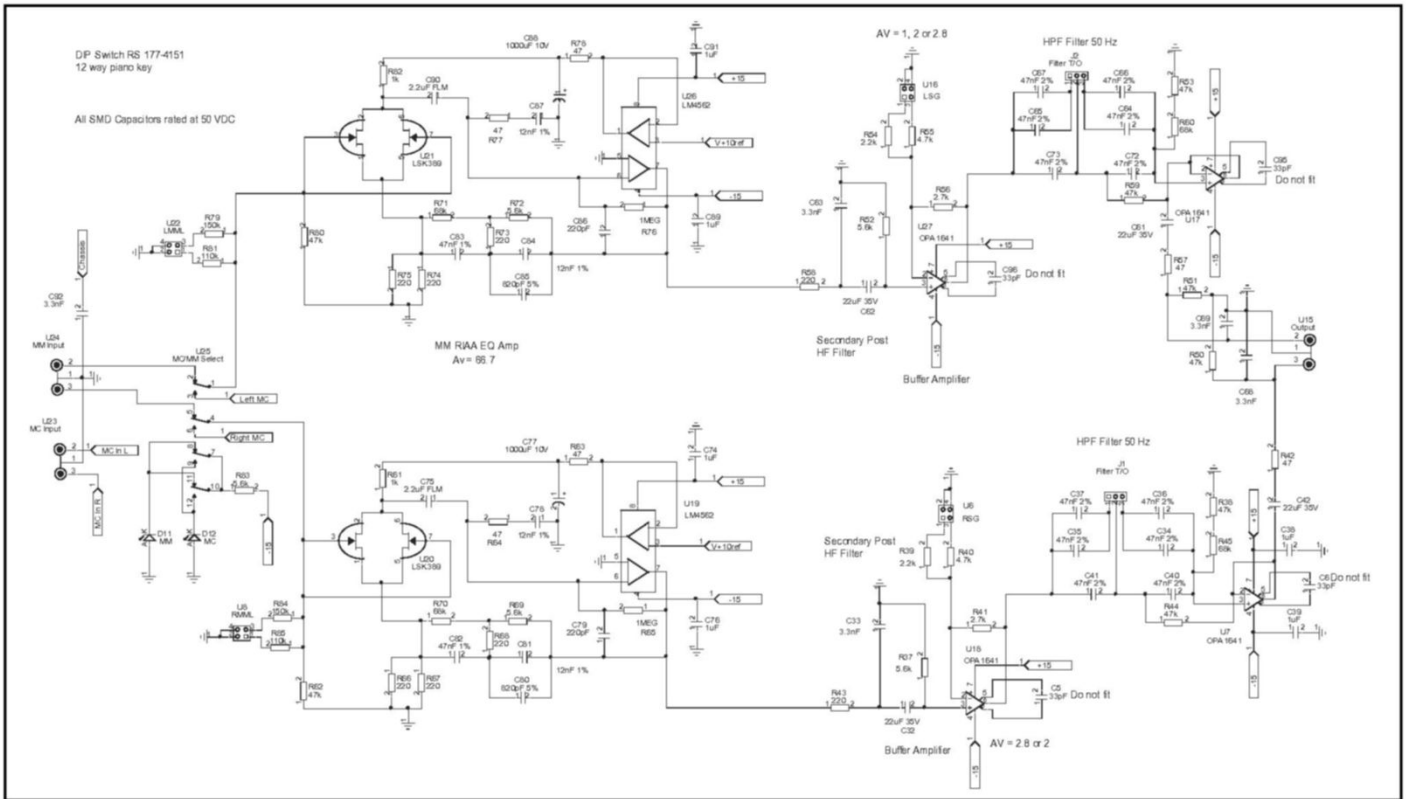


Figure 1: This schematic shows the design for the moving magnet phono amp, system gain amp, and the rumble filters.

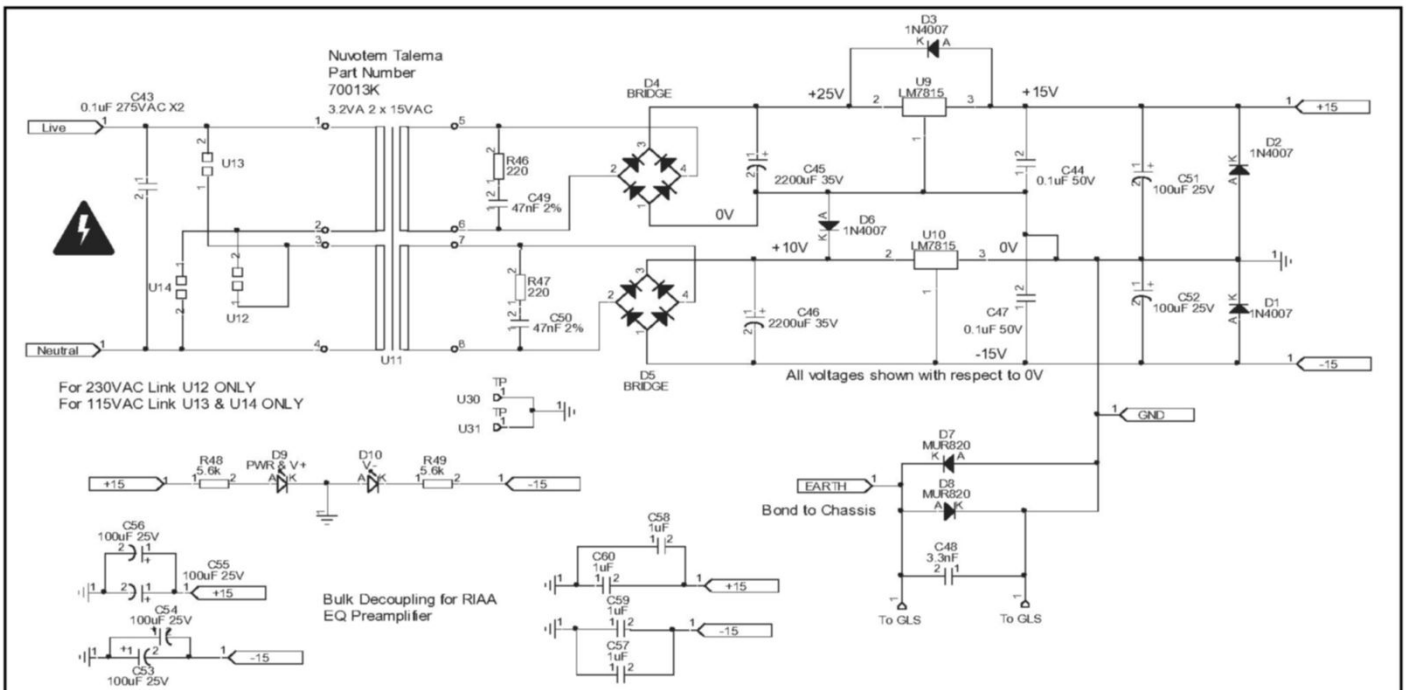


Figure 2: This is the schematic for the system's power supply unit and ground lifter.

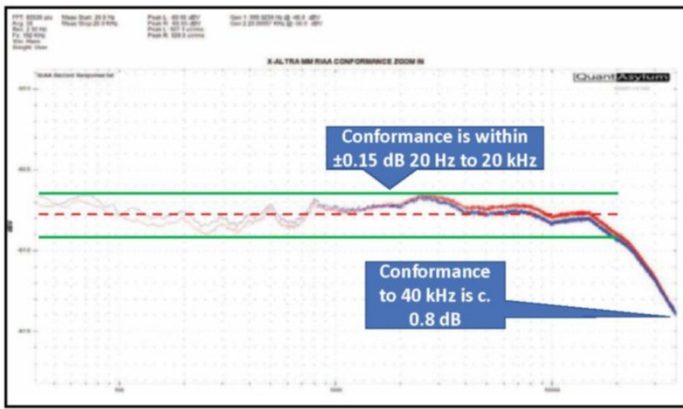


Figure 3: X-Altra RIAA's measured conformance 20Hz to 40kHz with the rumble filter switched out

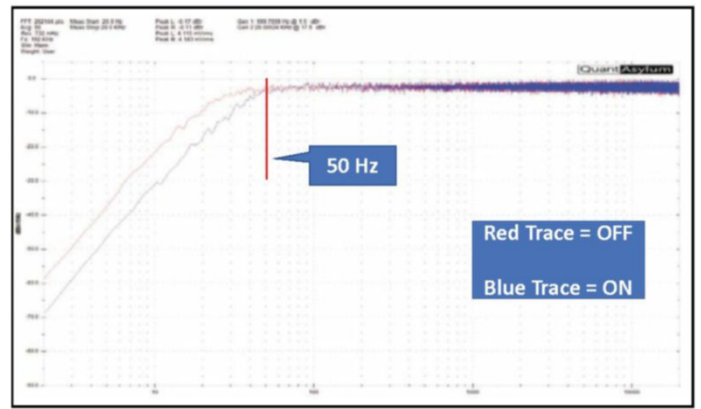


Figure 4: Acoustic rumble filter performance

secondary is separately rectified and smoothed (D7, D8, and C49 and C50). A snubber consisting of a 220Ω and 47nF capacitor is used across each of the bridge rectifiers to minimize RFI caused by diodes switching as they come out of conduction.

The smoothed DC appearing across the filter capacitors is fed to standard TO220 LM7815 linear regulators (U15 and U16) after which the two single rail supplies are combined to provide a -15V, 0V, +15V output. Output decoupling is provided by C44, C47 and C51 and C52. D1 and D2 ensure during power up or down, the rails are clamped and any polarity inversion greater than 0.6V is avoided—if not, potential latch-up may occur (see Douglas Self's *Small Signal Amplifier Design* for a write-up on this). Additional decoupling and local reservoir capacitance is placed around the

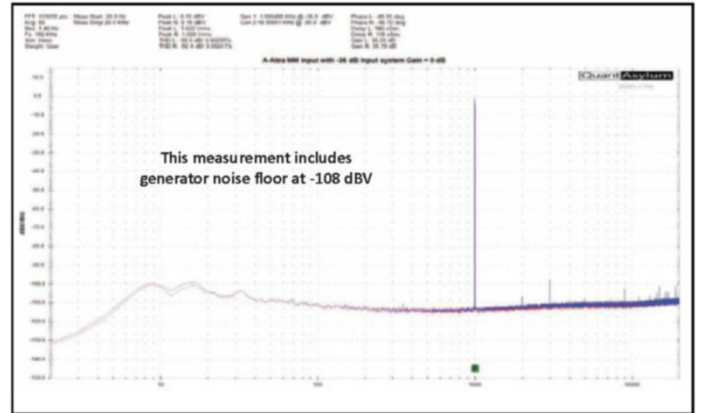


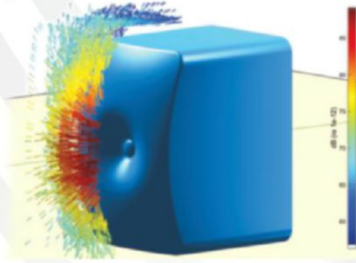
Figure 5: MM distortion at 15mV input (X scale 2Hz to 20KHz)

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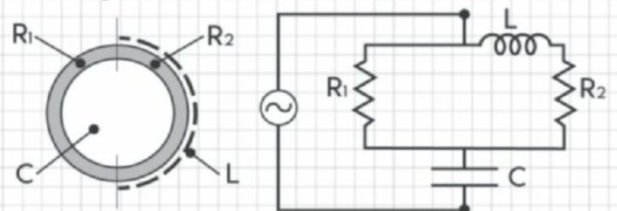
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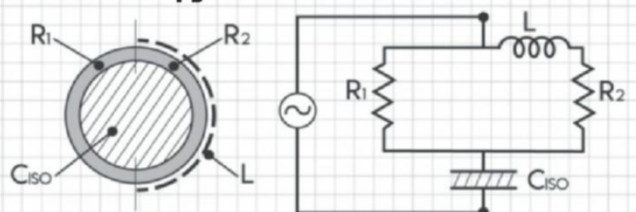
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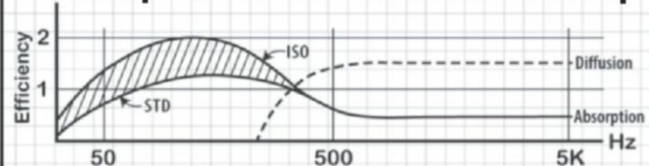
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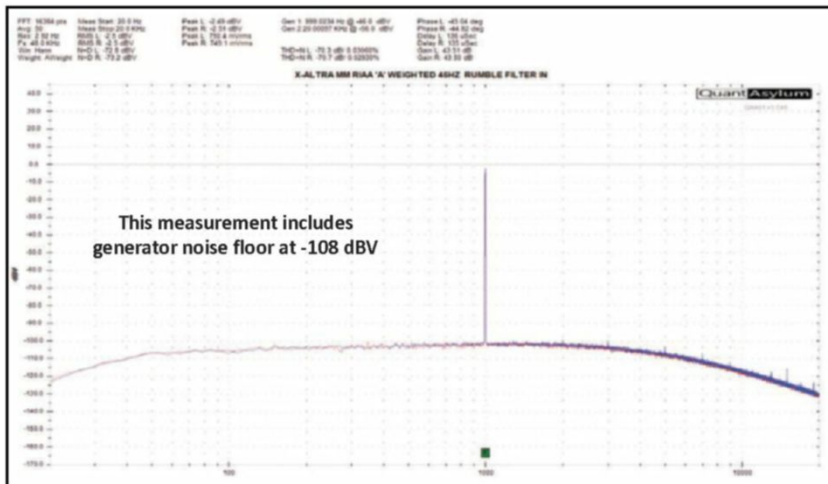


Figure 6: "A" weighted MM noise and distortion with acoustic rumble filter switched in

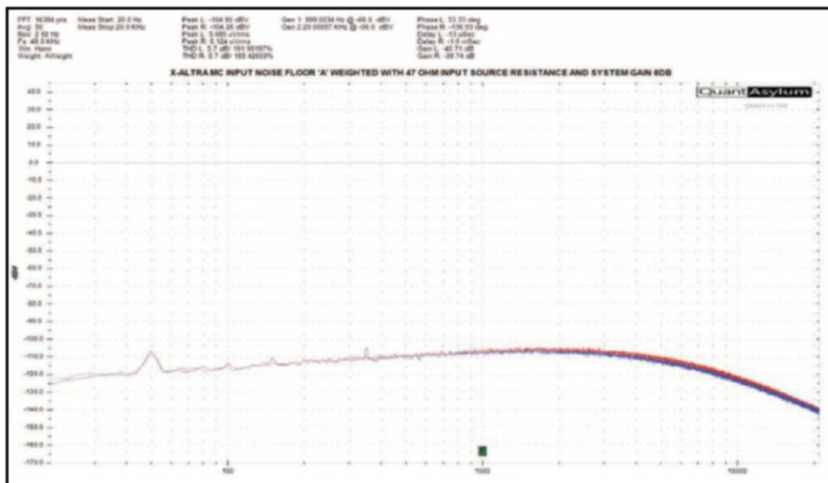


Figure 7: "A" weighted MC noise floor with 47Ω source resistance and acoustic rumble filter switched in

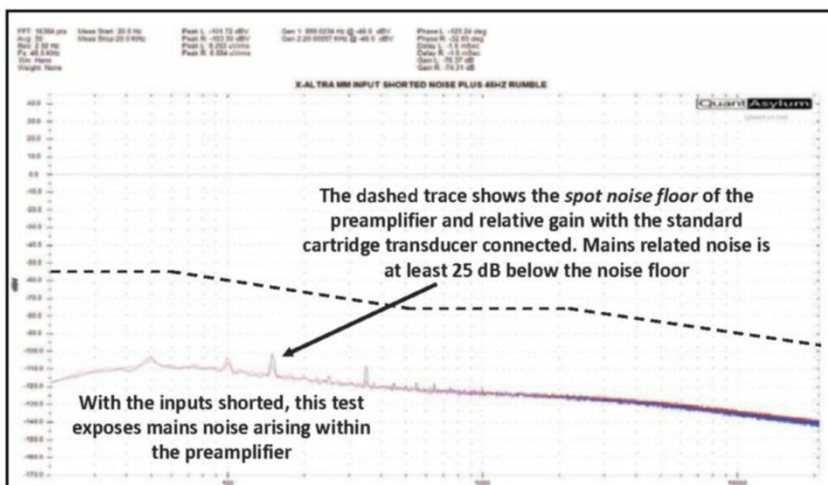


Figure 8: MM noise floor, input shorted, and acoustic rumble filter switched in, no weighting

PCB near the amplifiers, of which C53-C60 form part (the others are shown on the various amplifier section schematics). D3 and D6 clamp the reverse voltage across the LM7815 devices should the main filter capacitors discharge more quickly than the output filter capacitors.

A ground lifter consisting of two anti-parallel connected MUR820 200A peak surge capability diodes (D7 and D8) ensure that at audio frequencies the amplifier electronics float with respect to the housing and assume the ground potential of the preamplifier to which the X-Altra connects. At RF, C48 (3.3nF) effectively bonds the chassis to the power supply 0V, thus providing RFI screening through the metal housing. The ground lifter can be defeated by the "GL" switch located on the rear panel. C43, a 0.1μF 275 VAC "X" cap, is connected across the mains supply right next to the transformer and provides some filtering of any series mode line conducted noise, while U12, U13 and U14 allow the PSU to be configured for 115VAC or 230VAC mains input. For 230VAC operation, link U12 and leave U13 and U14 open, and for 115VAC operation leave U12 open, and link U13 and U14.

All of the PCB tracks around the rectifiers and reservoir capacitors are short and where possible all return paths are laid adjacent to each other on opposite sides of the PCB to minimize loop areas and thus any radiated noise.

Measurements

All the measurements (**Figures 3–8**) were taken with the QuantAsylum 401 Audio Analyzer, using the provided RIAA pre-emphasis curves. It is important to note, where applicable, that the input to the DUT is taken directly from the QA401 set at -46dBV for MM and -66dBV for MC and will include the DAC noise floor at -108dBV. In the noise floor measurements where the DUT input is shorted, no DAC noise is present, and the figures more accurately represent the preamplifier's noise floor.

Conclusion

The X-Altra RIAA EQ preamplifier's noise performance is achieved using input devices and circuit topologies best suited to the different requirements of MM and MC generator sources. On both inputs types, given the simplicity of the design, the thermal noise floor approaches state of the art, and particularly so in the case of the MC input preamplifier stage. On the MM input, noise performance is ultimately limited by the pickup cartridge L and R, but nevertheless, the SNR with the standard cartridge is > 3dB better than a NE5534A based circuit at >75 dB reference 5 mV, and just 0.2dB lower than Colin's LP797 from 2007

Project Files

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

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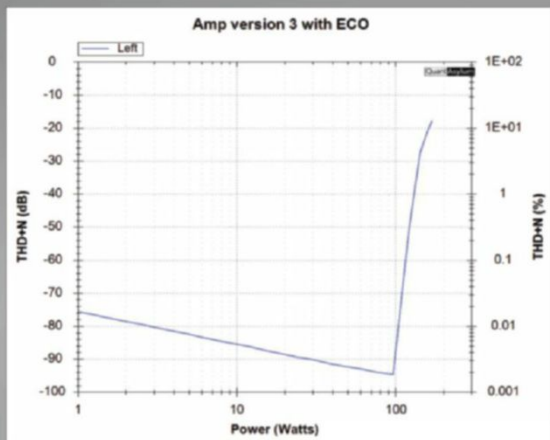
despite an 11x difference in the gain setting resistor. On both MC and MM, overload margin is > 31dB ref 500µV and 5mV, respectively.

The compact PCB layout was facilitated by the extensive use of SMD components, allowing the PSU to be co-located with the amplifier in the same

housing. The mains-related noise measured below -100dBV on MM inputs and -90dBV on MC inputs (both unweighted measurements).

For readers wishing to build the X-Altra preamplifier, PCB sets can be ordered at <http://hifisonix.com/x-altra-phono-eq-preamp>. 

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Souped-Up Practice Amps

After highlighting four of his favorite practice amplifiers, Richard Honeycutt details some modern-day modifications that he would make if he were going to build a clone of one of them—the Silvertone “case amp.”

By
Richard Honeycutt
(United States)



Photo 1: Long a popular small guitar amp, the Fender Champ was available for a time as model 5C1, using a 6SJ7 pentode preamp tube.

By far, most hollow-state guitar amplifiers use a very predictable set of tubes: 12AX7/7015/ECC83 preamps, 12AT7 phase splitters, and 6L6GT or EL34 output tubes (or 6V6GT or EL84 in lower-power amps). If a hollow-state rectifier is used, there is some variation among tube types. But in the late 1970s, guitarists discovered a new choice of amps that still garners a small following today. Originally called “practice amps,” they were small, having a single 8” or smaller speaker, and had a unique tone that was well adapted to blues guitar styles.

Flashback: In the 1950s, when rock and roll burst on the scene, a market for “beginner” guitars and amps suddenly appeared. Although many of these used triode preamp stages and a single-ended pentode or beam-power-tube output stage, there were a few that took the less-common path of using a pentode preamp. Four of these amps were the Fender Champ model 5C1, the Silvertone models 1430 and 1449/1457 (same amp, different model numbers), and the Vox AC4.

The Fender Champ

The Fender Champ was introduced as the Champion 800, and has gone through many design changes over the years. Originally

intended as a “starter” amp, it came to be used as a practice amp and as a recording amp in the 1960s. The original model number was 5C1 (see **Photo 1**). All three tubes are octal-based. The 6SJ7 preamp is a sharp cut-off pentode whose typical operating characteristics are shown in **Table 1**.

The 6V6 (or more the more commonly used 6V6GT) is a beam power tube having a maximum rated power output of 5.5W when used with a 315V B+ supply. The Champion 800 was rated at 4W output, running a 350V B+. The Champ went through a number of changes, of which the first major one was the change from a 6SJ7 pentode preamp to a 12AX7 and the output tube to an EL84/6BQ5. (See Figure 2.) The Champ model 5F1 (“’57 tweed”) was used as the studio amp when Derek and the Dominoes recorded “Layla” in late 1970. Fender has introduced 17 or 18 models of the Champ over the years.

The Silvertone Model 1430

The Silvertone model 1430 (see **Photo 2**) was a hot-chassis amplifier, meaning that the B+ supply as well as the filaments of the 35W4 rectifier and 50C5 output tube, were connected directly to the AC line with no intervening power transformer (see **Figure 1**).

6SJ7 Operating Characteristics

Heater Voltage (AC or DC)	6.3V
Heater Current	0.3A
Direct Interelectrode Capacitances:	
Grid to Plate	0.005pf
Input	6pf
Output	7pf
Maximum Overall Length	2 5/8”
Maximum Diameter	1 5/16”
Base	Small Wafer Octal 8-pin

Amplifier—Class A		
Operating Conditions and Characteristics:		
Heater Voltage	6.3V 6.3V	
Plate Voltage	100V	250V maximum
Screen Voltage	100V	100V maximum
Grid Voltage	-3V -3V	
Suppressor	Connected to cathode at socket	
Plate Current	2.9mA	3mA
Screen Current	0.9mA	0.8mA
Amplification Factor (approximate)	1100	2500
Plate Resistance (approximate)	0.7MΩ	1.5MΩ
Transconductance	1575μS	1650μS

Table 1: The 6SJ7’s operating characteristics



Photo 2: The Silvertone model 1430 “widowmaker” also used a pentode preamp.

This meant that lethal voltages could appear on the ground side of the input jack; hence the nickname “widowmaker.” This amp used a 12AU6 sharp-cut-off pentode as the preamp. Operating at 100V on plate and screen, this tube has a transconductance of $3900\mu\text{S}$ and a plate resistance of $0.5\text{M}\Omega$, making the gain of the Silvertone’s preamp stage a bit different from that of the Champ. The 50C5 was rated at 1.2W maximum output, about 5.3dB less than the Champ would put out. Actually, this means that the Silvertone would sound over half as loud as the Champ, if the speaker efficiencies were the same.

The Silvertone Model 1448/1457

The Silvertone 1448/1457 used essentially the same circuit as the 1430, but with the addition of a transformer to isolate the B+ and circuit ground from the power line.

This amp was strictly a beginner’s amp—it was actually built into the guitar case (**Photo 3**), but it produced a surprisingly good blues tone.

However, during the amp’s manufacturing life, distorted blues guitar sounds were not in vogue, so this was not considered a very desirable amplifier. Harmony, Wards Airline, and Supro all built amps that were substantially similar to the Silvertone “case amp,” although they were not built to double as guitar cases.

The Vox AC4

The Vox AC4 used an EF86 pentode as the preamp tube, a 12AX7 in the tone control circuit, and a single-ended EL84 as the output tube (**Photo 4**). The single tone control functioned as an improved “losser” control: the actual loser control provided only treble cut, but the AC4’s control used section 1 of the 12AX7 as a high-passed signal source, and section 2 as a low-passed source. Tone control P2 faded the signal fed to the EL84 between the high- and low-passed sources, providing true treble and bass boost options. The Vox also included a tremolo circuit using a neon-lamp relaxation oscillator that acted by varying the bias of the first 12AX7 stage (**Figure 2**).

The EF86 has a transconductance of about $2000\mu\text{S}$, which is between those of the 6SJ7 and the 12AU6. The EL84 is very similar to a 6BQ5, and puts out about 6W in a single-ended Class A configuration with a 250V B+.

The standard wisdom among tube enthusiasts is that amps with triode preamp stages have a smoother and more desirable distortion than amps with pentode preamps, because of the larger percentage of lower-order harmonics produced by overdriven triodes. However, bearing in mind that many artists have used “Power Soak” devices to allow them to overdrive their pentode or beam power output stages, the “triode sound” may not be the ultimate goal for everybody’s taste. In particular, I’ve found that the Silvertone 1449 “case amp” sounds just fine for Southern Rock.

Proposed Amp Modifications

If I were going to build a clone of this amp, I’d use a 6AU6 preamp rather than the original 12AU6, to keep all filament voltages the same. I’d change to either a 6AQ5A (discontinued, but available NOS) or a 6V6 output tube. These put out about 2W with a 180V B+. I’d move to a 6AX4GTB half-wave hollow-state rectifier, since the soft regulation they provide gives a nice feel for blues, and again, to keep filament voltages at 6.3V. Rather than using a simple isolation transformer, I’d change to a power transformer that would give me a higher B+ for the output tube,

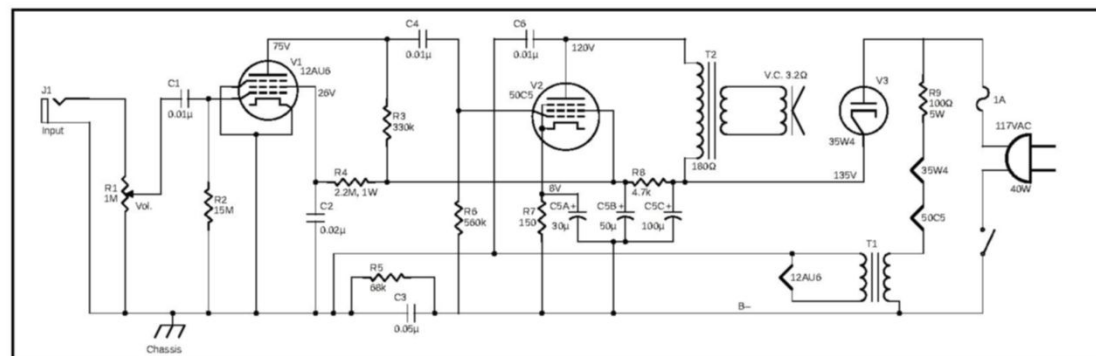


Figure 1: The Silvertone Model 1430 was a hot-chassis amp.



Photo 3: The Silvertone model 1449 (later sold as the model 1457) was another pentode-preamp model.

such as a Hammond 162G6. And the output transformer would need a 4,000Ω primary and an 8Ω secondary. A Hammond 125BSE would work, although its 5,000Ω primary impedance (with an 8Ω load) would slightly reduce output power.

A schematic of the modified “case amp” is shown in **Figure 3**.

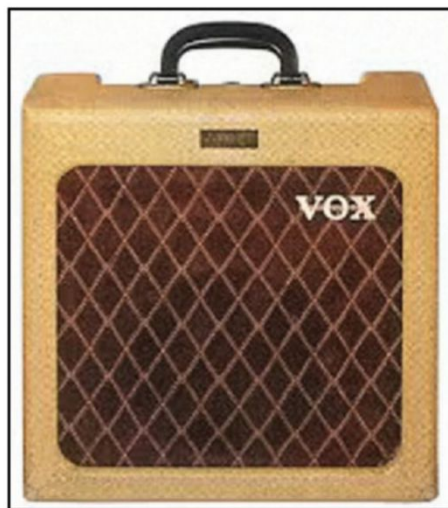


Photo 4: A pentode-preamp model from Europe, the Vox AC4 used circuitry similar to that of the Champ and the two Silvertones mentioned in the article.

Other than the changes mentioned above, some component values have been changed because the different power transformer and rectifier provide a bit higher B+ to the output stage, and I wanted to keep the voltages at the input stage unchanged. The triple electrolytic capacitor shown as a 30/8/100μF at 25/200/200V could be replaced by single capacitors. The LTspice simulation software did not run correctly with the 0.01μF capacitor from the 6V6 control grid to ground, as shown in the Silvertone Model 1430 schematic, so I omitted it. In building this amplifier, one might find that such a capacitor is necessary to prevent oscillation. The value may have to be determined by experimentation.

Photo 5 shows the output wave from the first stage when a 100mV_p, 1kHz sine wave is fed into it. The amplitude is 14.89V, and the stage is at the verge of clipping. **Photo 6** shows the Fast Fourier Transform (FFT) of this wave. Second-harmonic distortion is about 12%; third-harmonic is about 0.9%. So in contrast to standard wisdom about pentodes producing mainly odd-order harmonics, it appears that when grid-leak-biased, a 6AU6 or 12AU6 stage produces mainly second-order distortion.

Looking at the output stage, we find that it clips at about 1.45W with an

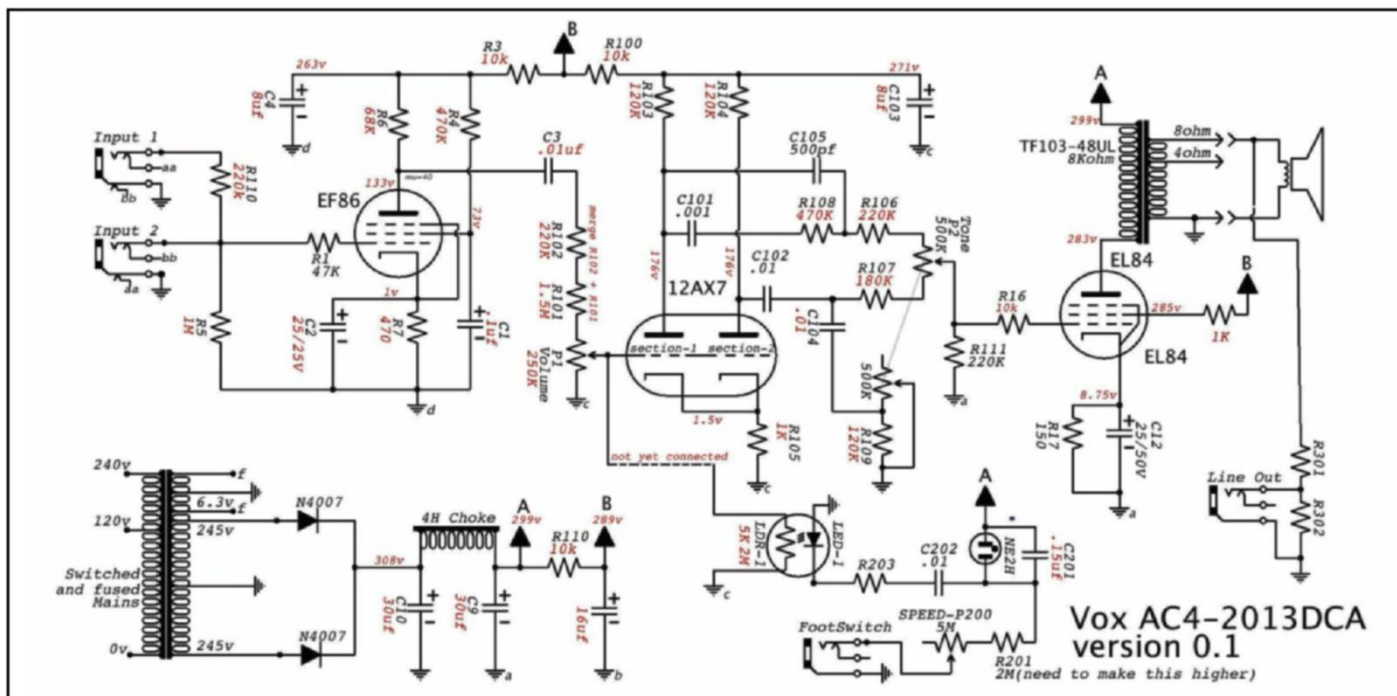


Figure 2: The Vox AC4 used a 12AX7 in an unusual tone control circuit.



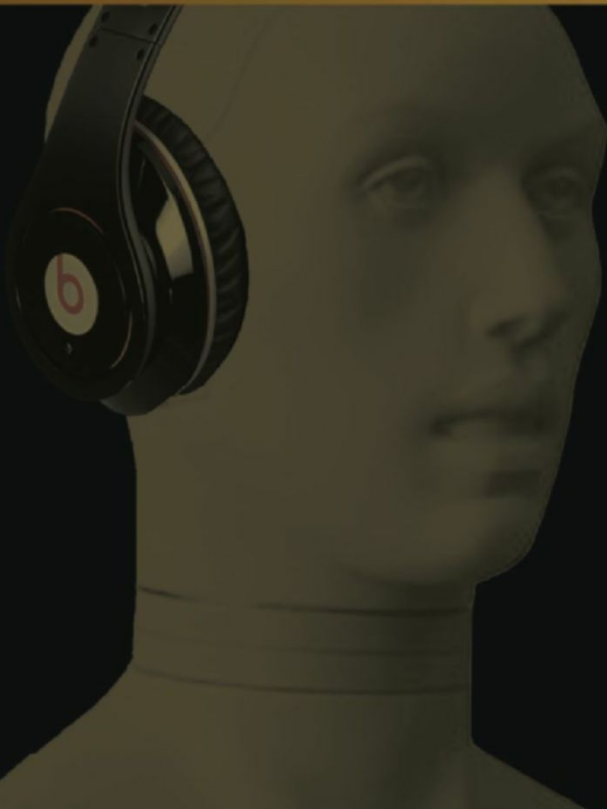
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Figure 3: This schematic shows the modified “case amp.”

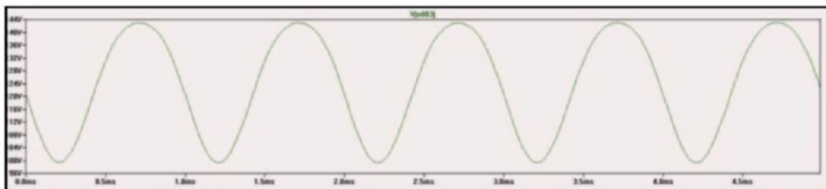
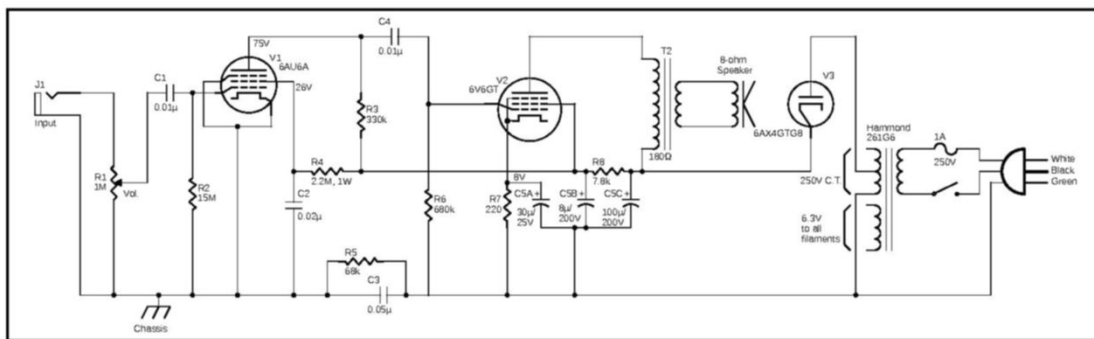


Photo 5: The output from the first stage is just shy of clipping.

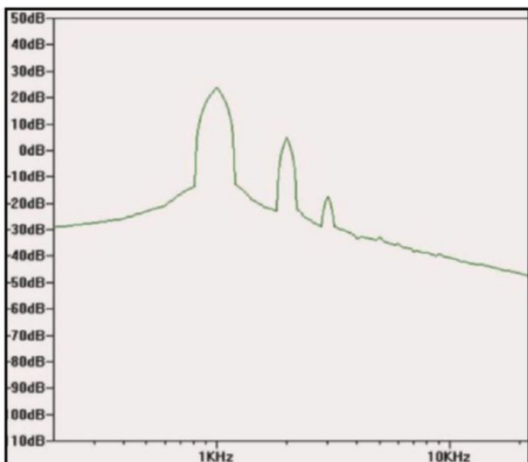


Photo 6: The FFT of the first-stage output wave shows mainly second and third harmonics.

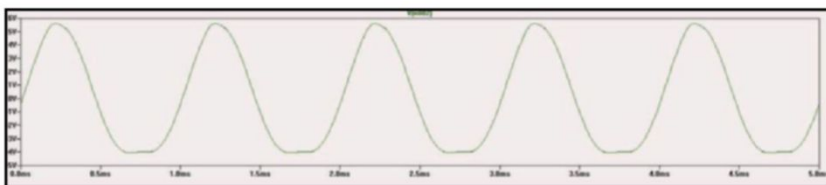


Photo 7: The wave at the output of the amplifier clips first on the negative half-cycle.

About the Author

Dr. Richard Honeycutt bought a *Popular Electronics* magazine from the rack at a bus station in 1960. It contained an article on building a transistorized audio amplifier, which captured his interest. He followed up by subscribing to several electronics magazines. To avoid the cost of parts, he built up a “junk box” by disassembling trade-in TVs from his uncle’s furniture store. His dad bought him the Van Valkenburg, Neville, and Nooger Basic Electronics book set, so he learned about tube circuits as well as transistors. He started repairing electronic devices about 1965, earning his First-Class Commercial FCC license in 1969. He has worked part-time in broadcast radio engineering and audio electronics repair since 1968, in addition to full-time work in acoustical and audio system design, plus a 20-year stint teaching electronics at the college level.



input of 10V at the control grid. The waveform is shown in **Photo 7**, and its FFT is shown in **Photo 8**. You can see a predominance of second harmonics, with higher harmonics decreasing monotonically as frequency increases. Total harmonic distortion at this output level is about 15%. When a 100mV input is applied to the amp with the volume control full up, the first stage outputs a signal whose voltage is above the clipping level of the output stage. Thus, if the output stage is driven into distortion, the first stage output will be clean: unless the output is severely overdriven, all you’ll hear is output-stage distortion.

Other options for modifying the “case amp” include marrying the input stage to the Fender ‘57 tweed output stage or the Vox AC4’s output stage. Either choice would require a different power-supply design, and would give more output power. Or if you simply replaced the input stage of the Vox with that of the Silvertone (again, making the necessary power-supply design changes), you would get the different tone-control action provided by the Vox’s more elaborate design. This choice would also provide tremolo.

Although amp manufacturers have responded to the demand for excellent small musical instrument amplifiers, the DIYer who enjoys rolling his/her own may benefit from this survey of some “old-timers.”

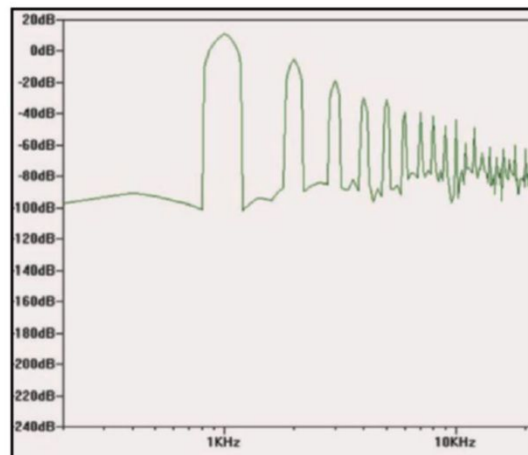


Photo 8: The FFT of the output wave shows decreasing harmonic amplitude as frequency increases.



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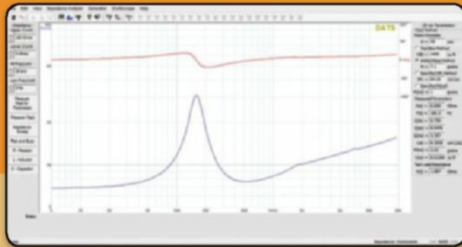
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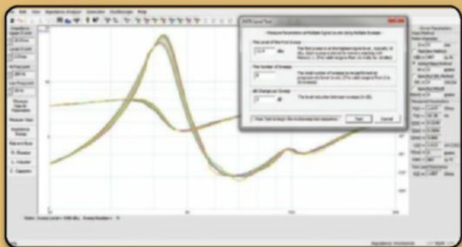
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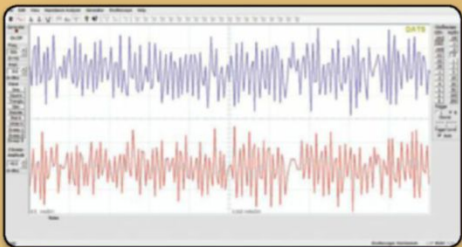
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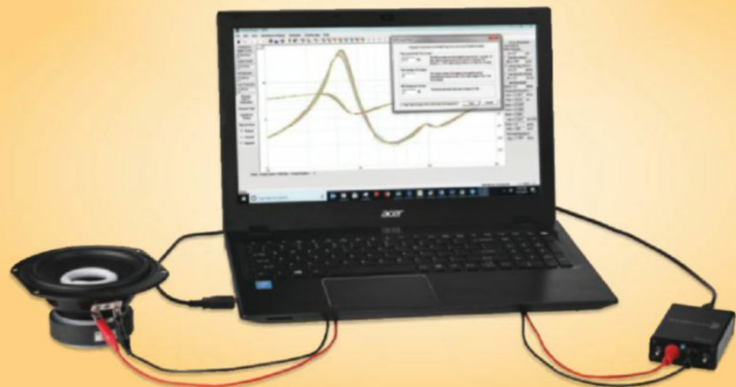
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