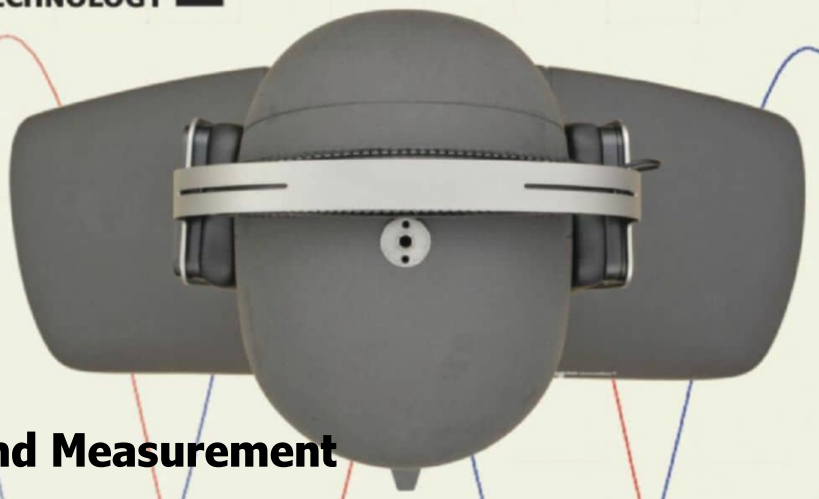
INNOVATIONS IN AUDIO • AUDIO ELECTRONICS • THE BEST IN DIY AUDIO

www.audioxpress.com

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY



Market Update

Test & Measurement

Evolution in Audio Testing

By J. Martins

Practical T&M

REST for Automated Test and Measurement

By Matt Taylor

Measuring MEMS Microphone Arrays

By Joe Begin (Audio Precision)



Market Update Extra

Voice Capture and Interface Systems

A Critical Front-End for Innovative Applications

By J. Martins



Development and Design

Sound Level Evaluation for Raspberry Pi 4

By Ben Chen (Sfera Labs)

Sound Control

Sound Levels How Loud Is It?

By Richard Honeycutt

Fresh From the Bench

KEF KC62 Subwoofer with Uni-Core Driver Technology

By Oliver Masciarotte and Kent Peterson

MARCH 2022

NEW

Western Electric

MAKER OF ELECTRON TUBES AND HIGH FIDELITY

91E Integrated Amplifier



Available in nickel, gold, and black.
Matched pair Western Electric 300B
electron tubes included.

THE FINEST INTEGRATED EVER MADE

Introducing the 91E, a new integrated amplifier from Western Electric. Meet ambitious, single-ended (SE) sound inspired by the 91A, its famous predecessor in high-fidelity for early Hollywood motion pictures. A groundbreaking new SE design evokes a classic feel with a sonic quality and resolution fit for today's listeners.

Experience over 20 watts per channel—a first for the 300B. Developed over two years, the 91E's patented circuitry intelligently steers the parallel feed constant current source, allowing the 300B to deliver more than double its previously rated output of 7-8 watts. You've never heard the 300B like this before.

FOR PRICING & AVAILABILITY VISIT

westernelectric.com/91e

HAMMOND

MANUFACTURING / DESIGNING
TRANSFORMERS & ENCLOSURES
SINCE 1917

Hammond Manufacturing | www.hammondmfg.com

March 2022 **ISSN 1548-0628**

www.audioxpress.com

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

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

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

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

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

Advertising rates and terms available on request.

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

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

© KCK Media Corp. 2022
Printed in the US



The Internet of People

Right after CES 2022 closed its doors it became obviously clear that we are still looking at the perspectives of another difficult year as the world struggles to overcome the effects of the global pandemic.

CES happened—albeit closing one day early—and just served to show how much we need to get back to business with real demonstrations, booths, and show halls. And as the industry looks at the uncertainties ahead, there are critical decisions to be made.

For those companies that have been working on innovative technologies and solutions that really need to be seen and heard—and need the momentum that only a trade show can generate—this is particularly difficult. To hold off on major product launches because of fears of missing the boost effect of an international trade show, or just in hope that, "things will soon be back to normal" is no longer sustainable. And too many audio companies clearly didn't work on a plan suitable for the circumstances.

CES 2022 was mainly noted for the absence of most familiar audio brands, and the strong focus on automotive and mobility—including Sony announcing its intention to enter the automotive market, not in systems, but with actual vehicles.

Important to note, the Consumer Technology Association (CTA) reported that the consumer technology industry overall will continue to see strong growth, even under these circumstances. Among the drivers for the demand are smartphones, automotive tech, health and wellness, smart home technologies, and streaming services. Audio technologies, including voice, content, and services are deeply embedded in all those. And new consumer electronics products show an increasing trend to offer audio systems (speakers, amplifiers, and microphones), while edge AI-powered voice interfaces are being adopted, taking over from virtual assistants. *audioXpress* is covering that angle with a dedicated Market Update report, which we released just in time to include in the online version of this March issue.

In our regular online reports and *The Audio Voice* newsletter, I wrote about all these trends—and noted some of the 10 most important audio innovations at CES 2022—all now available online at audioXpress.com

For this edition of *audioXpress* we also dived deep into audio test and measurement, with an extensive product and solutions update. As I noted in previous reports on that topic, the trend toward connectivity and collaboration had already become a priority since 2020. The global pandemic strongly affected many industries and markets, and has accelerated an evolution toward new methods of working and increasing collaboration between different players in development, manufacturing, and across the product lifecycle. Now, after two years feeling the effects of a phenomena that constrained or interrupted existing globalization trends and changed the relationships and priorities in the product production and supply chain, we can confirm that most of the trends characterized a year ago by *audioXpress* have been reinforced.

In education, instrumentation companies have completely adapted to the requirements of modern university teaching labs. Accelerated by the pandemic, remote learning technology is becoming the new norm. And as research and development continues to accelerate, companies and organizations are also adapting to new methods of working, and test and measurement platforms are increasingly being connected to cloud-collaboration tools and using data sharing interfaces, all the way to quality testing.

As the pandemic continues to impose severe restrictions on travel, the effect is particularly felt over relationships with core manufacturing hubs in Asia, creating the need for remote supervision, calibration, and even set up and maintenance of test and measurement systems, whether used in the lab or in the production lines.

The test and measurement manufacturers are evolving to respond to these needs. The specific challenges for audio test and measurement are very similar to those being felt by audio manufacturers in general—meeting the need for connecting people through connected devices.

J. Martins
Editor-in-Chief

The Team

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

Technical Editor: Jan Didden

Regular Contributors: Bruce Brown, Bill Christie, Bob Cordell, Joseph D'Appolito, Dimitri Danyuk, Vance Dickason, Jan Didden, Scott Dorsey, Gary Galo, Chuck Hansen, Richard Honeycutt, Charlie Hughes, Mike Klasco, David Logvin, Oliver Masciarotte, Nelson Pass, Christopher Paul, Bill Reeve, Michael Steffes, Nora Wong, Stuart Yaniger

OUR NETWORK

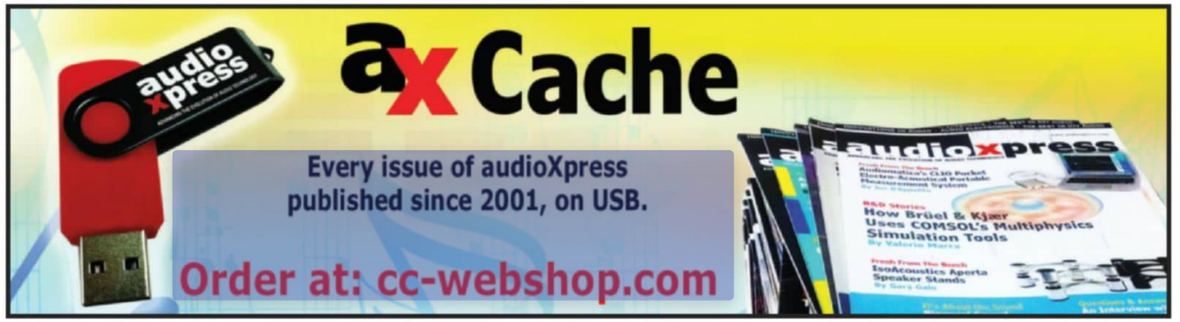
audioxpress | circuit cellar | LIS Loudspeaker Industry Sourcebook | Voice Coil

SUPPORTING COMPANIES

ACO Pacific, Inc.	49	Menlo Scientific, Ltd.	62
All Electronics Corp.	59	NTi Audio, Inc.	55
Amplitrex Audio Products, LLC	54	Outline s.r.l.	51
AXPONA	24	Parts Express International, Inc.	68
COMSOL, Inc.	9	Profusion plc	23
Ellisys Corp.	61	Purifi Audio	53
GaN Systems	45	QuantAsylum USA LLC	21
GRAS Sound & Vibration	43	Redco Audio	47
Hammond Manufacturing, Ltd.	3	SB Audience	17
HEAD acoustics, Inc.	19	Triad Magnetics	67
Hottinger Bruel & Kjaer, Inc.	15	Virtins Technology	35
iSEMcon, LLC	59	Warkwyn	47
KAB Electro-Acoustics	53	Western Electric	2
KLIPPEL GmbH	65		

NOT A SUPPORTING COMPANY YET?
Contact Peter Wostrel (audioxpress@smmarketing.us, Phone: 978-281-7708, Fax: 978-281-7706) to reserve your own ad space in the next issue of the magazine.

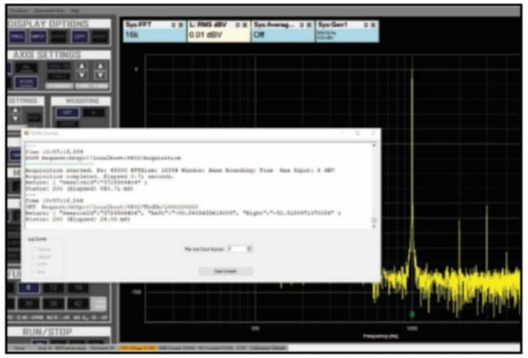




Volume 53 – No. 3 March 2022

audioxpress

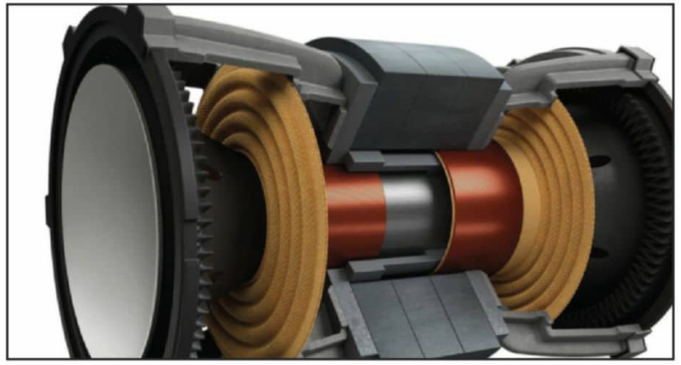












Features

8 Enter the (Tiny) Dragon
KEF KC62 Subwoofer with Uni-Core Driver Technology
By Oliver A. Masciarotte and Kent Peterson
Following the review of the KEF LS50 Meta speakers for *audioXpress*, this article details the latest KC62 compact subwoofer, combining KEF's force-canceling and Uni-Core technologies. Oliver Masciarotte reviews this subwoofer and speaks with Jack Sharkey, Senior Technical Engineer at GP Acoustics. Then, Kent Peterson measures the subwoofer using the Klippel Near Field Scanning system (NFS).

60 Sound Level Evaluation for Raspberry Pi 4
By Ben Chen
The article details the possibilities of Sfera Labs' Exo Sense Pi—a compact multi-sensor module with a Raspberry Pi 4 that also offers a built-in microphone—as a sound meter. The author illustrates the main concepts involved and possible software applications for sound level evaluation with the Raspberry Pi.

Columns

SOUND CONTROL
56 How Loud Is It?
By Richard Honeycutt
Over the years, our perception of loudness has changed and while sound can be measured in many different ways, we must be sensitive to what is the proper sound level for all concerned, not just what is possible.

Market Update—Test & Measurement

18 Evolution in Testing
By J. Martins
This market update focuses on the latest product launches and solution updates announced over the past 12 months for audio test and measurement plus an overview of the main market trends for this critical segment, which continues to evolve and grow.

Edition Focus—Test & Measurement

42 REST for Automated Test and Measurement
By Matt Taylor
This article makes the case for using web interfaces and embracing web standards for automated testing of audio equipment, fundamentally exploring the possibilities of REST—which stands for Representational State Transfer—a widely accepted set of guidelines for creating reliable web APIs.

48 Measuring MEMS Microphone Arrays
By Joe Begin
The simultaneous acquisition and measurement of signals from a microphone array presents a variety of challenges to developers seeking to evaluate their designs. This article details what to measure and how to test and evaluate multichannel microphone arrays.

Market Update*—Voice Capture & Voice Interface Systems

69 A Critical Front-End for Innovative Applications
By J. Martins
This article is an update on dedicated voice capturing and voice front-end solutions. From sub-miniature and MEMS microphones to complete voice front-ends for voice recognition and voice interface applications, this is an intense area of activity for developers.

* Bonus content that appears in the Digital Edition, log into your account to view

Departments

4 From the Editor's Desk
5 Client Index

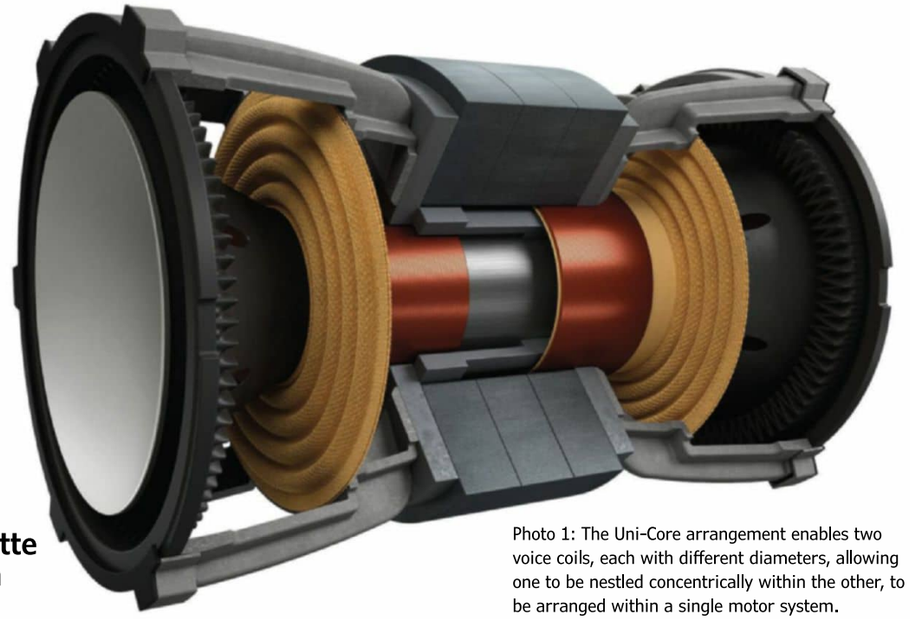
Websites

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

@audioxp_editor audioxpresscommunity
linkedin.com/company/audioxpress

KEF KC62 Subwoofer with Uni-Core Driver Technology

Enter the (Tiny) Dragon



By
Oliver A. Masciarotte and Kent Peterson

Photo 1: The Uni-Core arrangement enables two voice coils, each with different diameters, allowing one to be nestled concentrically within the other, to be arranged within a single motor system.

Following the review of the KEF LS50 Meta speakers for *audioXpress*, this article details the latest KC62 compact subwoofer, combining KEF's force-canceling and Uni-Core technologies. Oliver Masciarotte reviews this subwoofer and speaks with Jack Sharkey, Senior Technical Engineer at GP Acoustics. Then, Kent Peterson measures the subwoofer using the Klippel Near Field Scanning system (NFS).

I can't count how many times I've visited homes and studios and found subwoofers set up so wrong that I could instantly identify that a sub was in the house. Subwoofers, especially modern ones meant for music, are supposed to be aurally unobtrusive, adding a base of bass to loudspeakers that can't reach the bottom octave or two.

Enter The (Tiny) Dragon

Herald the KC62 microsub, which is shorthand for KEF Compact 6" x 2. Unlike most of the competition, the KC62 is not ported. Also unlike the competition, the KC62 has dual drivers. Actually, there is very little that the KC62 has in common with other small subwoofers. This little cube goes one better than the traditional force cancelling, opposing woofer configuration. To save space, and my goodness does it ever, designer Jack Starkey and his team at KEF designed "Uni-Core," a novel coaxial voice coil approach (**Photo 1**). Coaxial loudspeakers are not unusual and, when designed right, can provide point-source imaging. KEF is known for its widely respected and refined coaxial designs. Usually, those quotidian coax driver designs are a tweeter surrounded by a mid-woofer, and are both configured to launch their sound in the same direction.

In an attempt to game the immutable *pas de trois* of loudspeaker physics—cabinet volume plus low-frequency extension and efficiency—Starkey fused two force-cancelling, opposing low-frequency drivers by devising a common motor. The two voice coils each have different diameters, allowing one to be nestled concentrically within the other. This arrangement allows the voice coils to "... travel within their own gap without colliding." Two drivers mean more surface area, while the Uni-Core arrangement saves space. Utilizing the same constituents as 2019's larger and more conventional KF92 sub, the cones are of a laminated aluminum-over-paper construction.





Photo 2: Here is a close-up view of the KC62 driver in mineral white.

The KC62 is an active, acoustic suspension design for outstanding transient behavior and flatter frequency response. However, a sealed cabinet incurs higher internal pressure and higher temperatures for both the voice coils and the two 500W Class-D amplifiers. To address those obstacles, an extruded aluminum enclosure was chosen, clad in either an all–black or all–white finish. A metal housing plus 1000W of power amplification allows the product to cleanly play louder without resonant coloration, and go a good 10Hz to 25Hz lower than other micro subs. The little devil is spec'd at 11Hz to 200Hz ±3dB. You may ask, just how small is a "micro" sub (**Photo 2**)? In this case, we are talking a munchkin–sized 246mm × 256mm × 248mm (9.68" × 10.07" × 9.76"). What it lacks in size is made up for in bulk, weighing in at a surprisingly hefty 14kg (30.9lb). Hiding inside that little fatty is one KEF custom power supply for the electronics, supplying low voltage bits such as the CPU and DSP, and another PSU for the amps themselves.



Photo 3: Cross-section of the P-Flex surround at high excursion shows undistorted folds.

Not only is the design active, it's largely digital, with DSP for flat and extended response, while offering a range of boundary compensation and other accommodations for placement and environment. Given the design, one would expect long throw drivers, and KEF addresses that challenge with a novel folded surround rather than the more easily distorted half-roll approach. Its patented "P-Flex" surround is a pressure-resistant, pleated design (**Photo 3**).

According to KEF, the shape is optimized "… to resist the deformation caused by the changing internal air pressure of the cabinet, whilst allowing extended, linear movement" of the drivers (**Figure 1**). As an added extra, the signal processing incorporates "Smart Distortion Control Technology" (SDCT), a hybrid system combining DSP pre-correction with indirect cone motion sensing and feedback.

A Different Stripe

Although the KC62 has a subwoofer mode along with a Low Frequency Effects (LFE) mode, most folks think of subwoofers—when they think of them at all—as the LFE part of a home-theater setup. That is a decidedly old school slant right there, y'all. It's true that subs entered the common lexicon through consumer acceptance of Dolby Surround and its 5.1 setups in homes, but modern subwoofers are far more capable than those one note wonders of the 1980s. Granted, the subs included in home-theaters-in-a-box, the kind you find for $200 to $600 at your local big box store, are only good at vaguely approximating explosions and the like. They are not meant for or capable of providing satisfying musical enhancement.

The KC62 is a whole other animal. It provides a visually inconspicuous, aurally subtle but essential foundation for musical enjoyment. It unobtrusively "pressurizes" the room, filling in an octave-and-a-half that you may not know you're missing. During the review period, I only had stand mounters in the house and, in all cases, I found the midrange felt more realistic and complete when I added that tiny bit of extremely low-frequency energy.

Without the sub, the music was pointier, subjectively more mid–dominant regardless of which speaker I was using at the time. Bringing the sub into the equation made for a more enveloping and complete sense of pretty much everything I listened to. The KC62 provides a solid foundation on which to build a convincing sonic portrait.

Obviously, some material has little or no bottom two octaves, but that's the exception. That said, a few notes of caution are indicated…If you like it really loud, the KC62 isn't a good fit. When KEF makes a Uni-Core big sister, then that would be the model to check out. In the meantime, the beefier and more traditional KF92 ("maximum output," 110dB versus the KC62's 105dB) might be a better choice. If you have open baffle main speakers, likewise the KC62 is most definitely not the droid you're looking for. Open baffle speakers want open baffle subs. Madisound has a nice Linkwitz-style, open baffle Z sub kit that's inexpensive and looks easy to build.

True Confessions

Now I must confess to a totally un-audiophile indulgence. With friends over to our house for entertaining, I routinely dial up a moderate loudness curve that boosts both top and bottom while aggressively high-pass filtering below 20Hz to prevent excess excursion. This allows us to easily hold a conversation while still blanketing the house in background music. I found that goosing the sub gain past its "normal" amplitude setting gives the loudness a substantial increase in impact and fun factor without interfering with speech intelligibility. Since the rotary controls on the KC62 are detented, it's easy to return thing to "normal" after fiddling.

Control Freak

Let's talk about those controls…There's no plethora of slightly discombobulating knobs and switches as you'd find on the typical plate amp-based sub of yore. The controls that KEF provides are quite straightforward—the classic crossover frequency and volume knobs, along with a five position EQ switch for room gain compensation. A polarity invert slide switch, suspiciously marked "PHASE," and a Mode switch round out the grouping (**Photo 4**).

When I asked Sharkey about an alternative POLARITY label, he corrected me by saying that, since they only switch phase 180°, "…then yes in a sense it is a polarity switch but that wouldn't be the proper way to describe it: It's a phase adjustment." Mode is a two-position slider—the MANUAL setting is for use with a stereo line level input, while the LFE mode is for when the KC62 is being fed by the mono LFE output of the A/V Receiver (AVR). When using an upstream device, like an AVR, you can set the crossover via the LFE channel on your amp/processor, sending the desired frequencies to the subwoofer. When using an LFE from the AVR, setting the sub to LFE defeats the internal subwoofer crossover in favor of the AVR's crossover. If you are not using an upstream device to set the crossover,

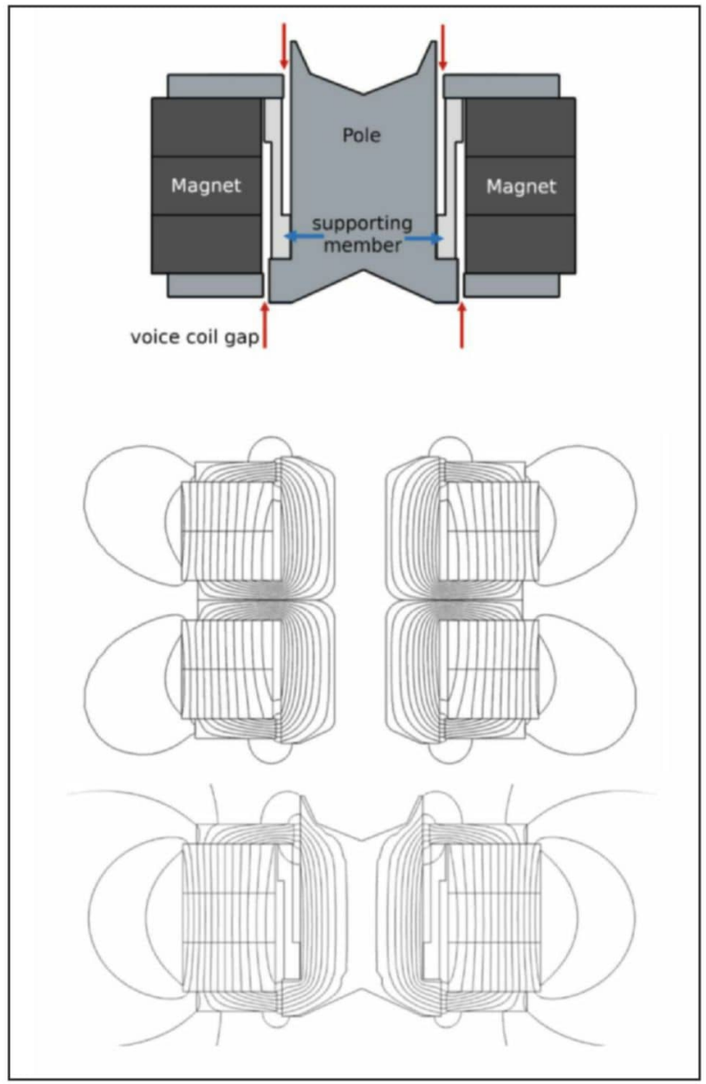


Figure 1: Uni-Core provides for more symmetrical and uniform flux where it counts.

About the Author



Oliver A. Masciarotte has spent more than 40 years immersed in the tech space, working on manufacturing, marketing, and product development for many pro and CE audio manufacturers including dbx, a/d/s, Lexicon, Sonic Solutions, and Minnetonka Audio. His client roster is as diverse as Apple, Harper Collins, NASA Johnson, NPR, and Universal. His writings include a book covering file-based music for the home and more than 100 articles for sundry trade publications. A member of the Audio Engineering Society (AES), Society of Motion Picture and Television Engineers (SMPTE), Project Management Institute (PMI), and Digital Cinema Society (DCS), he is currently co–founder and CMO of MAAT Inc., a digital audio software manufacturer.



Photo 4: The KC62's control panel shows the classic crossover frequency and volume knobs, along with a five position EQ switch for room gain compensation. A polarity invert slide switch, suspiciously marked "PHASE," and a Mode switch round out the grouping.

then you'd set the switch to Manual, which engages the subwoofer's own on-board crossover. Being able to manually control it prevents contention problems between the on-board crossover and the amp/processor's own crossover.

All processing is performed in the digital domain, and the EQ settings inform the embedded DSP. Choices are Room, Wall, Corner, Cabinet, and Apartment. The first three are self-explanatory, with each of the settings shelving the low frequencies differently. Wall is somewhat less steep, Corner is a little steeper, "…Cabinet is very steep, while Apartment is kind of shallow then gets very steep below around 50Hz" to attenuate structural-borne vibration traveling through walls and floors. Sharkey mentioned that all of the EQ settings, "…start around 200Hz or lower and then vary according to the setting to compensate for Boundary Gain."

I tested the KC62 in our living room, which serves the dual purpose of listening room as well. It is a decidedly typical space, not ideal for high-fidelity pursuits but that's what many folks are faced with. Due to furniture placement, I had only four locations that would work, and three of them resulted in extreme and intolerable room node excitation, with huge dips and nulls as one moved about the space. The fourth position turned out to be quite ideal. Without applying any active room acoustics compensation, my favorite being Dirac, the low end was fairly consistent regardless of seating position.

My test configuration included the $200 KW1 wireless transmitter kit. This optional add-on is essential for optimal placement as it frees you from interconnect length constraints. It operates in the 5GHz radio band, delivering audio at up to 24-bit/48kHz. The KW1 works with five different KEF sub models, and two $100 KW1 receivers (also sold separately) accommodate two simultaneous subs for more even low-frequency coverage. I fed the USB-powered transmitter from Parasound's JC 5 convenient parallel RCA aux outs, while the easily installed miniature receiver plugged directly into a pin grid block on the KC62's control panel. The receiver also sports an LED that acts as a reassuring idiot light when properly paired.

To set the controls, I decided to use what most consumers have on hand—their ears and a phone. I started by selecting the appropriate boundary compensation, and twisting the gain knob and crossover frequency to minimum. After that, it was a simple matter of playing some test material‡ while progressively increasing the crossover frequency until I closed the gap between speakers and sub. For objective verification, I employed Fabien Lefebvre's outstanding SPLnFFT measurement app for iOS. Since the KC62 is DSP-driven, it wasn't ideal to turn it off and on to bypass the sub, so I visually made a mental note of the volume knob position, and dialed the gain down then back to the reference setting to bring the sub in or out.

After about 15 minutes of futzing around, I arrived at a very pleasing group of parameters. The sub was not obvious unto itself, but when disabled suddenly produced a major letdown. "Where'd that bedrock go?" Over the course of a few hours of listening with a wide variety of music, I arrived at a default setting that I could easily veer off from when I wanted dance floor low end, and instantly arrive right back at my reference when the booty shaking had been satisfied.

I can find little fault in the KC62. My time with the product was illuminating in a variety of ways. A well-behaved sub, the KC62 is certainly that, contributes the appealing low-frequency underpinning that most speakers cannot provide or that one cannot afford. It places the midrange in proper perspective, psychoacoustically completing what one doesn't know one needs until it's missing. When properly set up, a sub will not call attention to itself. Instead, it plays the vital role of aural fundament, on which your speakers invisibly rest.

Despite its size, the KC62 handily competes with units both larger and more costly, failing only when stomach-churning levels of bottom octave are needed. You pay more for performance, and this little dragon delivers. Speaking of size, the KC62's extra-small form factor and curvaceous good looks combine to provide a very high spousal approval factor.

The team at KEF have followed its award-winning LS50 Meta with the perfect low-frequency complement. (Read my *audioXpress* June 2021 review at https://audioxpress.com.) Together is better. The reasonably priced stand mounters and sub occupy a tiny amount of floor space, yet dish out heaping oodles of wonderfully high fidelity sound. If you're considering a quality sub purchase, I suggest you try to include some time with a KC62 in your schedule.

Author's Note: ‡ I used the AudioTest application from Katsura Shareware, setting up a 10Hz to 80Hz slow sweep.

Inside the Mind of Jack Sharkey, Senior Technical Engineer at GP Acoustics International

Oliver Masciarotte: What was the design brief? To create the world's heaviest sub in the world's smallest footprint?

Jach Sharkey: The world's "smallest footprint" is spot-on, and the necessary result of doing that the right way was the weight!

Masciarotte: What were the design challenges and how were they overcome?

Sharkey: Obviously the biggest design challenge was to create an enclosure for a large subwoofer in a very small package. The challenge was to create a "small" subwoofer that didn't compromise on performance but was able to compete with much larger subwoofers in a super consumer-friendly package for the way people are setting up their listening spaces today.

Masciarotte: As with the LS50 Meta, you chose a novel approach to the problem; the concentric voice coils. Why the expense embodied in extra R&D, tooling, extended lead time, etc., when you could have gone the "easy" route of low profile, opposing drivers?

Sharkey: Since 1961 we've been about engineering solutions instead of just making something that might enhance performance. We think outside of the current tech box and seek ways to solve problems without regard for the cost or time involved. Blade and KC62 are prime examples. Using opposing drivers would not have saved as much space as we wanted as the concentric voice coils did, but doing concentric voice coils that perform at a very high level required a lot of other design and research efforts. It's the typical design journey—one thing leads to another, and there are tons of compromises along the way; the key is picking and choosing what compromises to live with. The weight of the KC62 is a perfect example of that: In order to perform at the level we wanted KC62 to perform at, the box was going to be necessarily heavy. We've never been about doing things the "easy" way—you can't really innovate and cut corners at the same time. The extended lead time is really just a result of the pandemic and all of the surrounding supply chain issues that all of us are confronting at the moment. Under normal circumstances KC62 would have been readily available and shipping without delay.

Masciarotte: Tell me about the enclosure…why aluminum instead of resin/composite or something else?

Sharkey: In a cabinet housing this powerful of a system, there would be serious (pressure) equalization forces on the cabinet walls. This would cause intolerable vibrations that would color the sound as well as force us to put in so much bracing that the internal air volume would be decreased, hence decreasing the low-frequency response of the subwoofer (compromised internal volume leads to decreased performance). To overcome this, an extruded aluminum cabinet was used. Aluminum has a higher elasticity than wood for a given thickness, so the use of a thinner wall was possible, allowing for a larger internal volume. Curved surfaces are also significantly more rigid than flat surfaces. The aluminum extrusion is a curved shell forming a monocoque or structural skin…This structure ensures there are no stress points, which might compromise cabinet integrity causing distortion. We could therefore reduce internal bracing and not have to worry about the strength of the enclosure when responding to the internal air pressure force.

Masciarotte: Is this product manufactured in China to your specifications?

Sharkey: We are a Tier One manufacturer, which means we control every aspect of production from concept to packaging as the product leaves the factory. We have a totally owned, and internally designed by our industrial and engineering teams, factory in Huizhou China. But as a Tier One manufacturer, we don't use contract labor, contract factories, sub-assembly factories, or any of the like—it's all done by our employees to our standards, which is a bit of a rarity in our industry today.

Masciarotte: Why did you go the "old fashioned" way of manual controls?

Sharkey: We did both, so the end-user has a choice on how to control and operate the sub. You can either use the manual controls or if you are using the KEF Connect app with an LS50 Wireless II for example, most of the subwoofer functions can be controlled via the app. As far as embedded software and DSP for the sub, because of the above and how subs are generally thought of for high-end audio, we opt more for the "straight-wire-with-gain" mindset—less is ultimately more when it comes to sonic fidelity.

audioXpress

KEF KC62 Subwoofer

The Measurements

The KEF KC62 subwoofer was measured at Warkwyn's facility using the Klippel Near Field Scanning system (NFS) in combination with Klippel's In-Situ Compensation module (ISC) to examine distortion, compression, and frequency response curves in the different "Mode" settings.

By Kent Peterson

The KEF KC62 subwoofer was measured at Warkwyn's facility using the Klippel Near Field Scanning system (NFS). The NFS delivers a 360° acoustical radiation balloon allowing for an examination of the radiation pattern, on/off-axis frequency response at any angle or distance. Of course, for this examination, it would be rather ridiculous to examine subwoofer radiation balloons as they are mainly omni-directional, and we all know what those balloons would look like. So, we also employed Klippel's In-Situ Compensation module (ISC) to examine distortion and compression of the KC62. The ISC, in conjunction with the NFS software, allows a user to make certain measurements including CTA-2010A and B after using Klippel's signature room removal abilities.

For the measurements of the KC62, we used a calibrated ACO Pacific free-field microphone with the 4048 pre-amp and a 7052E capsule (**Photo 1**). Measurement points around the speaker totaled 2000 and were processed with a resolution of 18 points per octave and from 20Hz to 20kHz. The length of the stimulus was 0.68 seconds with an averaging of 4.



Photo 1: The KEF KC62 is shown mounted for measurement.

Frequency Response and Various Settings

The KC62 has some great flexibility incorporated into its amp section, which enables users to tailor the sub to various listening locations in their homes. These are "Room/Wall/Corner/Cabinet/Apartment" settings that apply various filter options, as well as a 4-dip switch option, which adjusts various high-pass filters from 40Hz through 100Hz.

A detented, rotational crossover knob that adjusts from 40Hz to 140Hz. For the purposes of our analysis, all dip switches were set to 0/bypass, the mode set to "Manual" and the crossover was rotated completely to the 140Hz setting to allow the sub to play its complete range. Volume for the system was set at the maximum position. As shown in later portions of this analysis, various voltages were then injected into the sub to analyze system compression (**Figure 1**).

Figure 2 illustrates the frequency response of the KC62 while bypassing all filter options, with the high-pass filters set to its maximum (140Hz), and then for each of the EQ setting available for room optimization.

Leaving you to your own devices (or by using the LFE mode), the system response is centered around 150Hz and extends up to 400Hz and as low as 20Hz, 10dB down (we did not measure below 20Hz as output at that point was nominal). Inserting your own crossover filter in this mode may allow the user greater flexibility by shifting the crossover point higher and delivering greater output between 100Hz and 200Hz.

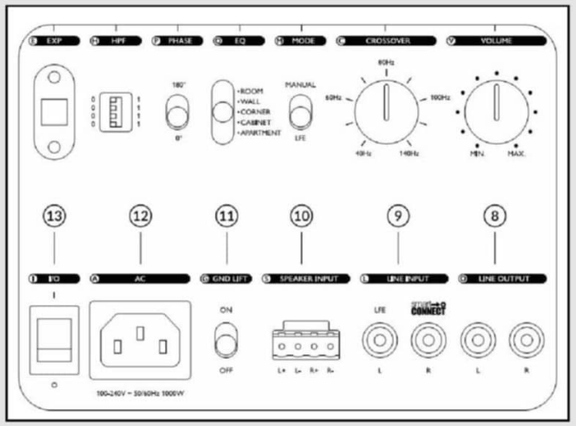


Figure 1: The manual shows rear panel selection options.

HBK
HOTTINGER BRÜEL & KJÆR

MATCHING THE CHARACTERISTICS
OF HUMAN SPEECH AND HEARING

Realistic Testing Of
Voice-Operated Devices

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

HBK offers market-leading products that facilitate more realistic speech synthesis and listening by reproducing human characteristics with very tight tolerances on accuracy to ensure repeatability. Our mouth and ear simulators – led by the new High-frequency Head and Torso Simulator (HATS) – accurately match human audio characteristics, thereby ensuring both quality and efficiency in your testing and product development.

Learn more about the HBK solutions at www.hbkworld.com/smartspeaker-testing

HBM
Brüel & Kjær

TECHNOLOGY BY
Brüel & Kjær

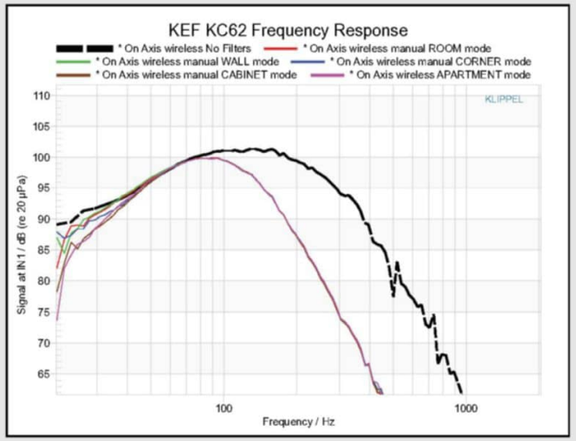


Figure 2: This graph illustrates the frequency response curve comparison of the KC62's "Mode" settings.

With the various EQ settings engaged, the center frequency is lowered to between 80/90Hz and applies a low-pass filter with a 10dB down point at 180Hz. Where the EQ seems most effective is lower in frequency, between 20Hz and 50Hz. Of these modes, the Wall mode and the Room mode exhibited the greatest output down to 20Hz, with the Room mode exhibiting a small 1dB bump at 22Hz.

The Apartment mode and the Cabinet mode have the greatest amount of attenuation down to 20Hz, with a 10dB difference between those and Room and Wall modes. Either way, these differences are very low in frequency and 5dB to 10dB lower than no filtering whatsoever. Though probably effective in keeping the drink coasters from rattling in your neighbor's apartment, they are subtle changes to be sure.

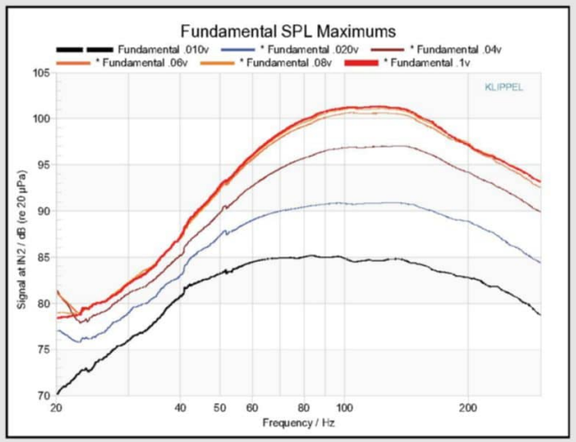


Figure 3: Here are the results of our comparison of compression vs. fundamental.

Compression and Maximum Output

In pondering what measurements would be most beneficial to the casual listener and reader, maximum output is certainly one of them, especially when examining a subwoofer that has little to discuss in terms of radiation pattern and very wide frequency response. First, we decided to examine when and where any DSP compression would begin to flatten frequency response and curtail SPL.

Beginning with 0.01V, we swept the sub while increasing voltage by 0.01, and then in subsequent steps of 0.02 until output had maxed, and no further output could be had at 0.1V. Keep in mind that we tested a powered system here with the volume knob positioned at maximum. So, it's important to note that the voltage levels are ratios and, depending on where your volume level is, your voltages are going to vary radically if you are one of those people who want to set up a meter to monitor your sub input. The main take-away here is maximum decibel output and how that affects your frequency response. Your easiest measurement tools here are a good mic, a calibrated analyzer, and about 1 meter.

At 0.01V, there appears to be very little compression, however, even a small increase in voltage (0.01V) appears to engage compression below 50Hz. Subsequent steps to 0.1V only result in an increase of 6dB to 10dB in the lowest frequencies. Remembering our human association with the decibel; a 3dB change can be considered human threshold, a 6dB increase can be considered "clearly noticeable," and a 10dB change a doubling (or halving) of apparent loudness. Turning up this sub to its maximum from 0.01V should result in an apparent doubling of output in the lowest frequencies until a point where compression limits max output at 80dB to 85dB (consideration of A-weighting here, which is valid, will make your lower frequencies measure far less over the lower spectrum of course).

Compression between 60Hz and 200Hz is less than that of the lower frequencies and results in a 6dB increase in output until 0.04V is reached. At 0.06V the real clampdown happens before capping at 0.1V, where no further output could be measured above 101dB and at 100Hz (**Figure 3**).

Tone Burst Center F	Weighted & Limited Maximum SPL 2π	Weighted & Limited Maximum SPL 4π
20Hz	68dB	62dB
25Hz	83dB	77dB
32Hz	96dB	90dB
40Hz	95dB	89dB
50Hz	105dB	99dB
63Hz	107dB	100dB
80Hz	106dB	100dB
100Hz	105dB	99dB
125Hz	107dB	100dB
160Hz	96dB	90dB

Table 1: To quickly examine how the KC62 might fare against the current CTA-2010B standard, we detailed the last decibel reading encountered before distortion failures are registered.

Tone Burst Measurement/CTA-2010B

Finally, a tone burst measurement was performed to quickly examine how the KC62 might fare against the current CTA-2010B standard. For reporting, we detailed the last decibel reading encountered before distortion failures are registered (**Table 1**). Center frequency reporting for 2010B is listed with corresponding decibel levels and for two scenarios; 4π (full space, or how we performed the measurements), and 2π (half space, which is more applicable to "real-world" conditions). However even with 2π measurements, your results may even be higher by a decibel or two due to placement near walls.

The values reported in Table 1 represent the last pressure level that "passed" the 20210B landmarks before distortion failures were encountered in each band. As per the red type, for the center frequencies of 20Hz and 25Hz, at no time during the tone burst measurement was there a result of "pass." There just wasn't enough acoustic output way down low to overcome the distortion values. Though to be completely fair, the chamber we used for this measurement is closer in the building to HVAC so it may be that the measurement this low in frequency is compromised by the noise floor. Nevertheless, the values listed in Table 1 very closely mimic our frequency response measurements and those of compression; in the lowest frequencies compression came on almost immediately, which is another factor for consideration.

Finally, no voltages are listed here as we simply turned the volume all the way up as it is a powered sub.

Final Thoughts

I did have a chance to do some spot listening of the KC62 during the time I had to work with it. Though, sadly, not paired with the KEF LS50 Meta cabinets we examined a few months ago. But when I did have the chance to listen, I was very impressed with how they sit in a mix and am obviously impressed with the product line; well-thought-out and very well manufactured. Are they the loudest/lowest sub out there? No. But that's not where I think this sub sits in the marketplace—it is a refined little subwoofer, made with quality materials and sophisticated engineering and will work great in your library, office, or living room and deliver just the oomph you might be missing.

About the Author

Kent Peterson is sneaking up on 30 years in the Pro Audio and Acoustics field. Having designed hundreds of audio systems for professional install and approximately 100 miles of noise walls, Peterson now spreads sawdust on employee "accidents" managing Warkwyn in Saint Paul, MN.



Evolution in Testing

This market update focuses on the latest product launches and solution updates announced over the past 12 months for audio test and measurement plus an overview of the main market trends for this critical segment, which continues to evolve and grow.

By
J. Martins



Keysight has launched a new portfolio of Smart Bench Essentials (SBE) lab bench instruments in 2021. These are targeted at remote access labs and teaching institutions that require a unified graphical interface with integrated data management and analysis, also allowing sharing instruments and data remotely with peers.

Audio test and measurement entails multiple applications that some manufacturers try to combine, while others target more effective tools for specific tasks, some more time sensitive, others completely data critical. Companies in this sector have perfected those strategies with new products and improvements to existing solutions over the past year, which was rich in announcements.

Our previous market update on this topic expanded our overview to more generic test bench and instrumentation solutions, in an effort to characterize the broader test and measurement (T&M) sector. The perspective enabled us to better understand how practices in education, research and development, and manufacturing are now adjusting to completely new methods, including the convergence with the latest generation of CAD, modeling, and simulation software tools. That's a trend that continues to gain strength, including for audio test and measurement.

Another of the notable trends we highlighted last year resulted from the fact that people are more connected than ever and that—with the effects of the global pandemic being particularly felt over the supply chain and restrictions in traveling—we have started to see a clear trend toward equipment and solutions that leverage connected resources, collaboration, and even remote monitoring and supervision. And that's also an approach that manufacturers of audio test and measurement equipment are addressing, even if just barely starting the process.

Accelerated Consolidation

In 2021 we could see a clear trend for acquisitions and consolidation in the test and measurement segment, directly affecting some key names in the audio industry.

In 2021, Amphenol Corp., the global provider of high-technology interconnect, antenna, and sensor solutions, announced the acquisition of MTS Systems Corp. MTS was the company that had acquired PCB Group in 2016, becoming the owner of PCB Piezotronics and Larson Davis. Amphenol subsequently sold the test and simulation business of MTS to Illinois Tool Works (ITW), as part of the acquisition strategy.

PCB Piezotronics specializes in the design and manufacture of force, torque, load, strain, pressure, acoustic, and vibrations sensors. The pioneer of Integrated Circuit-Piezoelectric (ICP) technology, PCB's instrumentation is used for test, measurement, monitoring, and control requirements in many different industries. Also the result of

HEAD acoustics
VoCAS
The Benchmark is Real Life
VoCAS is software that objectively tests and evaluates speech recognizers for example in the smart home, vehicle, or smartphone in a realistic, optimally replicated the dynamic nature of real life environment.
REAL-LIFE-PROOF.
www.head-acoustics.com/VoCAS

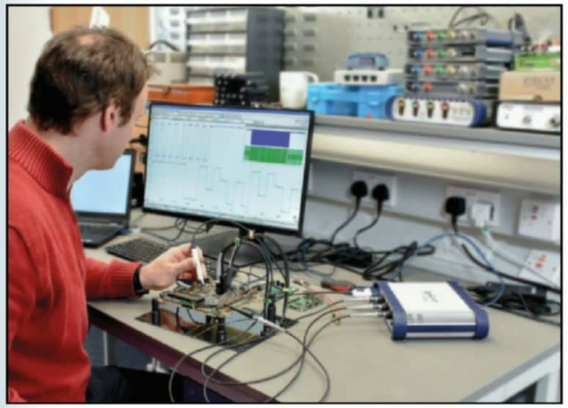
multiple acquisitions and mergers—within MTS and before—PCB Piezotronics has four major divisions, all of which offer targeted sensor technologies. This includes PCB Piezotronics' core business division with piezoelectric, ICP, and sensors; IMI Sensors, for industrial vibration, pressure sensors, and accelerometers; and Larson Davis, with its full line of noise and vibration measurement instrumentation, which includes sound level meters and outdoor noise monitoring systems.

Other PCB companies include Accumetrics digital telemetry systems; The Modal Shop, structural vibration and acoustic sensing systems and services for various applications; Endevco test and measurement solutions; and Modern Microstructures, a pure-play foundry specializing in the fabrication of custom microelectromechanical systems (MEMS) and thin-film devices.

Also in 2021, it was announced that the measurement-focused business group established by investment company Battery Ventures, which included Audio Precision, GRAS Sound & Vibration, and imc Test & Measurement, would be consolidated under a new identity and single brand: Axiometrix Solutions.

Combining a strong group of well-established brands that have served electronics, aerospace, automotive, and audiology industries for more than 35 years—each with a rich history and diverse roots in the US, Denmark, and Germany—Axiometrix is now a global business with locations in North America, Europe, and Asia. As the intended strategy confirms so far, the company's three main product lines will continue to be reflected by its three industry-leading brands in each of their respective segments.

A group competing directly with Axiometrix that also followed a similar consolidation strategy was Hottinger Bruel & Kjaer (HBK), the entity that resulted from the merger of Brüel & Kjær Sound and Vibration A/S (BKSV) with Hottinger Baldwin Messtechnik GmbH (HBM) in 2018. Both were already owned by UK-based investor Spectris plc, which decided to merge the respective businesses into HBK. The merger aims to redefine the concept of a complete solution, where the two companies will unite their expertise and provide an integrated offering, encompassing sensors, acquisition, preparation, evaluation, and engineering services. While other markets (familiar with the German company HBM Test and Measurement) might accept HBK, within the audio industry, the trend will be to refer to the resulting company simply as Bruel (no one knows how to say Kjær...). Both BKSV and HBM are leading brands in their respective disciplines. Bruel in sound, noise, and vibration, while HBM is focused on industrial control and weighing. What the combination will allow is an evolution toward the trends we mentioned earlier, with better software and data acquisition and evaluation solutions. To date, Bruel continues to do business with its own identity and website.



Offering another example of increasing flexibility in T&M, in 2021 Pico Technology expanded its PicoScope 6000E Series to address high-performance debugging and design verification requirements in advanced electronic systems, including a free-of-charge Software Development Kit that enables users to write their own application, with code examples and drivers for Windows, macOS, and Linux, and drivers to interface with NI LabVIEW and MathWorks MATLAB.

The Economics

Economic data available for the test and measurement market is very scarce and there is much less for the specific audio industry instrumentation market. Multiple studies characterizing the segment entail a wide variety of applications, from electronic test equipment and laboratory-specific solutions to data acquisition and scientific analysis, to a large variety of industrial and manufacturing solutions. This makes it difficult to understand the growth, which is the only common factor described in recent market research reports. According to one fairly recent study from Quince Market Insights, the global Test and Measurement Equipment market is estimated to be worth $28 billion USD in 2021 and is projected to grow at least 4% annually in the next few years. Figures and trends that are confirmed by previous reports from Technavio.

And that growth is directly related to large investments in communications, such as 5G networks, data centers, and wireless networks, as well as automotive, aerospace, medical, electronics, and semiconductor sectors. Companies that are reportedly feeling the benefits of those investments include Rohde & Schwarz, which is supplying key testing technology for 5G; Yokogawa Test & Measurement in automotive power solutions, data, and production line testing; and some of the largest players such as Anritsu, Keysight, National Instruments, Teledyne LeCroy, and others. To get a sense of the market, suffice it to say that Tektronix (now part of conglomerate Fortive), a company that is much larger than any of the manufacturers of test equipment for the audio industry, doesn't even make it to the top 10 of the largest companies in the market reports we analyzed.

And yet, Frost & Sullivan estimates the market opportunities just for electronic test and measurement to be worth much more. According to a pre-pandemic article by Frost & Sullivan, the electronic T&M market encompassing semiconductor automatic test equipment (ATE), radio frequency (RF) and microwave (MW) test equipment, digital test equipment, field testers, and data acquisition (DAQ) generates well over $10 billion USD in revenue annually, and predictions for 2022 pointed to $13.18 billion USD, with significant growth resulting from 5G, automotive technologies, and semiconductor ATE investments.

With audio and key sensor technologies increasingly a priority in the consumer, automotive, and industrial spaces, it becomes obvious that we will continue to see further opportunity for consolidation, while the transition to software and connected systems will open opportunities also for new players.

Audio has become ubiquitous, from smartphones to the smart home, nearly all consumer products have audio features, and from development to manufacturing, requires reliable, efficient testing of loudspeakers, microphones, integrated components, and more.



Working together with Rohde & Schwarz, HEAD acoustics created a test solution that was able to successfully verify the communication quality of 5G (VoNR)-capable devices, using its hardware platform labCORE and measurement and analysis software ACQUA, with the latest generation CMX500 wideband radio communication tester from Rohde & Schwarz.

Company Resources

Amphenol Corp. | www.amphenol.com
Audio Precision | www.ap.com
Axiometrix Solutions | www.axiometrixsolutions.com
Brüel & Kjær | www.bksv.com
GRAS Sound & Vibration | www.grasacoustics.com
Hottinger Baldwin Messtechnik GmbH | www.hbm.com
Illinois Tool Works Inc. | www.itw.com
imc Test & Measurement GmbH, www.imc-tm.com
Larson Davis, www.larsondavis.com
PCB Piezotronics, Inc.| www.pcb.com
Quince Market Insights | www.quincemarketinsights.com
Spectris | www.spectris.com
Technavio | www.technavio.com



HEAD acoustics

New Generation of Artificial Heads with Advanced Types of Artificial Ears



The new HEAD acoustics HMS II.3 LN HEC artificial head emulates the acoustically relevant structures of the pinna and ear canal with an improved replication of the ear impedance, providing higher dynamic range.

Meeting the needs of current audio systems development, German audio and communication test and measurement specialist HEAD acoustics introduced a new and much improved design for its artificial heads paving the way for new era of communication and audio quality measurements. The new generation of artificial heads ensures a new level of realism, precision, and flexibility when coupled with HEAD acoustics' modular concept approach, meeting the most diverse of applications in the field of acoustic measurement technology.

The new generation of artificial head measurement systems is available in three base variants, including a low-noise artificial head with a human-like ear canal (HMS II.3 LN HEC), the standard low-noise artificial head model (HMS II.3-LN), and an artificial head focused on voice and audio quality testing (HMS II.3). The modular concept enables users to exchange different types of artificial ears as required, benefiting from anatomically correct artificial ears, and a fullband-capable two-way artificial mouth.

According to the company, the new anatomically correct artificial ear design available in the human-like ear canal (HEC) model is able to better reproduce the ear canal geometry and the human ear impedance. The low-noise occluded ear simulators feature an impedance response matched to that of the human ear, while the built-in highly sensitive microphones provide a best-in-class dynamic range. The very low inherent noise of 16 dBSPL(A) makes the head-and-torso simulator the ideal choice for measuring close-to-the-ear and in-ear audio and communication equipment. Due to the maximum sound pressure level of 148 dBSPL, the artificial head is also suitable for measurements of far-to-the-ear transducers in hands-free systems, in-car communication systems, or smart speakers. The head-and-torso simulator provides geometric and acoustic characteristics according to the ITU-T P.58 recommendation, and thus replicates all acoustically relevant structures of the human anatomy.

In addition, numerous detailed refinements and an extensive range of accessories help to make life easier for acoustics engineers. While the low-noise occluded ear simulators of the artificial head simulate the receiving direction, a two-way mouth loudspeaker reproduces the complete spectrum of the human voice, allowing for narrowband, wideband, super-wideband and fullband measurements. And with the new HMS II.3 LN HEC, users can create aurally accurate voice and audio recordings for evaluation using HEAD acoustics' measurement and analysis software ACQUA.

The HMS II.3 artificial head can also be equipped with an artificial nose (AN-HMS), enabling acoustic measurements of audio and communication devices where a nose ensures secure hold of augmented reality glasses and virtual reality headsets. Users can also mount all artificial head variants on the height-adjustable tripod HMT III or on the torso box HTB VI.

HEAD acoustics' newly developed artificial ear with HEC is now standardized as Type 4.4 artificial ear in Recommendation ITU-T P.57. The HEAD acoustics HEC artificial ear not just fulfills the Type 4.4 artificial ear requirements, it is also the first ITU-T standardized artificial ear to meet the low-noise requirements for artificial ears as defined in the new Chapter 7 of Recommendation P.57.

www.head-acoustics.com



The HMS II.3 LN HEC realistically emulates all the acoustically relevant structures of the human ear and enables exchanging ear simulators and pinnae quickly and easily.











AXPONA

AUDIO EXPO NORTH AMERICA

APRIL 22-24, 2022 • CHICAGO

RENAISSANCE SCHAUMBURG HOTEL & CONVENTION CENTER

HEAR HERE!

GREAT MUSIC. INCREDIBLE SOUND. ALL IN THREE DAYS.

Whether your passion is analog, streaming, headphones, speakers, the music or all of the above, AXPONA is where you can discover the latest gear from the brands you love to create your ultimate listening experience. And you can connect with fellow enthusiasts, designers, and industry experts.

Feast your ears on more than 200 exhibits to really experience what the newest audio products can deliver, including:

- **Turntables**
- **Loudspeakers**
- **Headphones**
- **Vinyl**
- **Electronics**
- **Accessories**

See you at AXPONA!



Live Concert – April 22, 8:00 pm

Friday Night Blues Night!

TORONZO CANNON

Tickets on Sale NOW! Visit www.AXPONA.com

For information on exhibiting, contact Mark Freed: mark@jdevents.com, 203-307-2688

Health & Safety:
AXPONA is committed to the health and safety of our exhibitors, attendees, partners and staff amid the COVID-19 pandemic and as we prepare for the AXPONA 2022 event. For more information visit: https://www.axpona.com/health-safety.asp

HEAD acoustics

Audio Measurement and Analysis Flexibility with ACQUA 5 Software

The well-established ACQUA measurement and analysis environment from HEAD acoustics recently received the first iteration of the new generation 5 software. ACQUA's significantly reworked software structure provides even greater flexibility for voice and audio quality evaluation, while paving the way for future expansions. Designed for the requirements in applications as diverse as communication systems, Bluetooth, ANC headphones, and smart speakers, ACQUA now supports faster hardware configuration, perceived loudness analysis, and MDAQS audio quality evaluation.

In combination with HEAD acoustics' labCORE audio interface, the software forms a precise, automated and flexible system for R&D, conformance tests, and quality control of devices and systems.

ACQUA includes a multichannel analysis system for diagnosis of acoustic and/or electric transmission paths up to 192kHz. With the measurement and analysis software, users can gather and evaluate measurement data in a simple and quick manner via measurement descriptors (Single Measurement Descriptor or SMD). These SMDs are pre-defined, adaptable, and embedded in the database structure. And all telecommunication-specific tests comply with international standards (e.g., ETSI, ITU, 3GPP, GCF, PTCRB, and GSMA). These core features make ACQUA a reliable foundation for a complete measurement chain: From the generation of complex excitation signals through the automated conduct of measurements and the analysis of measurement signals to secure storage and documentation.

ACQUA 5 introduces the new "Workplace Settings" window that enables users to collect and configure hardware items, select and define calibration, and combine assignments. The user can define and configure his local workplace, including the use of head-and-torso simulator (HATS), microphones, loudspeakers, and radio testers. All properties for a single hardware item are found in one single place, considerably streamlining working with ACQUA. Assignments of equalizations, calibrations, and corrections are distinctive, and hardware setups are easily verifiable.

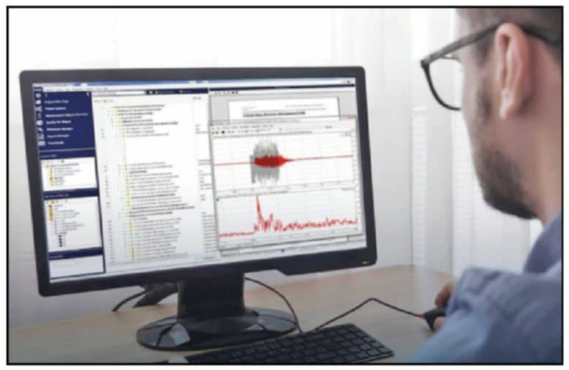
The analysis possibilities with the software have also been extended with new loudness calculations for speech communication, according to Recommendation ITU-T P.700, which enables subjectively perceived loudness analysis, not just the traditional loudness rating calculations. The new psychoacoustic-based loudness calculation allows a much more realistic calculation of the speech loudness perceived by the user. Even non-LTI systems, such as AGC and compression schemes, can be evaluated. This provides an excellent analysis of all measures integrated by manufacturers targeting the increase of the subjectively perceived loudness without producing excessive sound pressure levels.

Another new feature is the full integration of Multi-Dimensional Audio Quality Score (MDAQS) evaluation, previously available in beta. MDAQS provides unique audio quality evaluation parameters for all types of audio equipment ranging from headsets to loudspeaker arrangements. With ACQUA 5, the new perceptual-based audio quality measure is integrated in the company's analysis portfolio for the first time.

Also extremely important for current applications, ACQUA 5 addresses the fact that often more than one HATS is used in laboratories. Either these are used for different purposes or more than just one HATS is used for the measurements (e.g., with in-car-communication systems or interfering talkers in smart home applications). Consequently, the DRP/ERP correction can now be configured for each HATS individually.

A variety of additional improvements are found in ACQUA 5, ranging from SMD handling to scripting possibilities up to more optimized database features.

www.head-acoustics.com



HEAD acoustics' ACQUA measurement and analysis software enables users to optimize voice and audio quality in all kinds of products and markets.



ACQUA, in combination with HEAD acoustics' labCORE audio interface, forms a precise, automated and flexible system for R&D, conformance

Echo Test + Measurement

Specifically Made for Audio Testing!

Nestled in a quaint building in downtown Santa Barbara, CA, Echo Test + Measurement is a little company with a big impact on audio test and measurement for consumer electronics. Founded in 1978 by Milo Street as Street Electronics, the company had early success in building text-to-speech cards for the Apple II and licensing custom DSP hardware and software to large computer and semiconductor companies (Analog Devices, HP, Sony, Motorola, Rockwell, and AKM).

From the 1990s through the first two decades of the 21st century, Echo Digital Audio established itself as one of the premiere pro audio sound card suppliers with the Mia, Gina, Layla (PCI-based) and AudioFire (Firewire) sound card lines.

Echo expanded from its earlier work in speech compression, to music synthesis, game audio, and music recording, building hardware for every major type of audio transport technology: Bluetooth, FireWire, USB, Ethernet, and PCI, among others.

During the later years of this successful run as a sound card company, a growing number of DIY audio enthusiasts and bloggers started performing measurements of audio equipment at home using free PC software, and sharing their experiences online. The same thing happened with a few very large consumer electronics companies that started using the AudioFire sound cards for production line testing. This eventually led to a collaboration to design and make a sound card-based system dedicated to test and measurement, leading to the first Echo AIO products in 2014.

Since 2014, Echo Digital Audio has fleshed out the AIO Test System, changed the company name to Echo Test + Measurement, and shipped more than 25,000 units to consumer electronics manufacturing plants all over the world. Echo has now completely transitioned to a test and measurement company and no longer offers traditional audio sound cards.



Echo Test + Measurement has created a robust, affordable, and comprehensive hardware platform that is USB 2.0 audio class compliant, so it can be used to build bespoke test systems that are especially suitable for high-volume testing and verification applications.

"With over 40 years of sound card expertise, we found ourselves in a unique position to make a significant impact on custom test and measurement systems for consumer audio manufacturing. Now, instead of spending many man-hours to configure audio sound cards to work properly as audio test systems, you can plug-and-play an Echo AIO Test System and use the time saved to make more and better products!," states Milo Street, Echo Test + Measurement CEO and hardware guru.

With its deep history and expertise in sound card interfaces, Echo has taken all the DIY-ishness out of creating a test system based on sound card technologies.

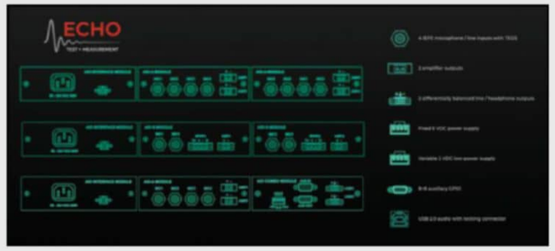
So what sets the Echo AIO Test System apart from traditional sound cards?

According to Kris Jackson, VP of Sales for Echo Test + Measurement, "Echo products have been renowned for decades for audio performance, longevity, and reliability. Since 2014, we have built on that legacy to deliver accurate, proven, and cost-effective audio test hardware. Today, Echo provides audio test solutions for mass production to some of the best-known consumer electronics companies in the world."

Echo AIO uses test industry standard connections and interfaces, such as IEPE microphone connections on BNC connectors with CCP power, and TEDS readers built into each channel. Output options include line level, headphone, and power amplifiers on EuroBlock connectors with added sense resistors built in for making impedance measurements. There is even a 10-channel TDM bus module for making chip level (I^2S) measurements. Other available options include battery simulation, GPIO inputs and outputs, and Pressure, Temperature, and Humidity sensors (unique to Echo). All these features that come standard on Echo AIO are costly add-ons to traditional sound card-based test setups.

The AIO Test System also enables the user to control all parameters of the hardware using three different methods. First, by using the AIO Control Panel application, a simple to use hardware setup and control app. Second, through the AIO Command-Line application that allows hardware configuration under program control in a test sequence. And finally using the AIO Interface Library, a dynamic library with a standard C-language interface allowing complete control of the AIO from any programming language.

The Echo AIO test system is modular, allowing for greater flexibility and system expansion. Two AIO modules can be fitted in a chassis. Maximum analog channel count is currently eight analog mic/line inputs by four outputs (line, headphone, or amplifier). Maximum digital channel count is 10x10 (TDM). Combining an analog module and a TDM module in the same chassis achieves a system maximum of 14 inputs by 12 outputs.



With affordable audio interface solutions that are proven and reliable, Echo Test + Measurement uses the slogan "bench quality and factory tough!"

The AIO Test System is also cross-platform and supports Windows, Mac, and Unix-based systems. It supports WASAPI and CoreAudio drivers in addition to ASIO (recommended).

Finally, these solutions from Echo work with any audio test software that interfaces to the mentioned audio drivers, including LabVIEW, MatLab, APx500 Flex, Soundcheck, ARTA, and others. Users can also write their own interface using common C languages and the Dynamic Library. In effect, compatibility and ease of interfacing makes Echo AIO an attractive choice for adding to existing systems or building completely new testing environments.

And the company intends to expand the range further through 2022.

www.echotm.com

Audio Precision

APx v7 Audio Measurement Software Updates



Audio Precision (AP) has released the latest update to its APx500 audio measurement software, version 7, adding new enhancements suited to a wide range of audio test applications. In parallel with release 7.0, Audio Precision also introduced APx500 software subscriptions, offering APx users an alternative in how they license APx software, and access nearly any software version and option.

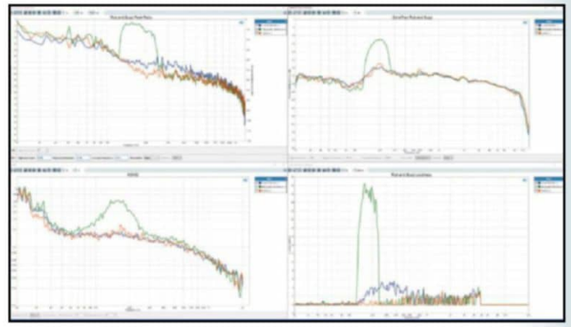
With the new APx500 software version 7.0, Audio Precision is adding support for GRAS SysCheck2-capable microphones, including the newly announced 246AE 1/2" CCP free-field microphone, and the 246AO pre-polarized pressure field microphone. The combination of SysCheck2 and APx enables the user to verify the accuracy of the entire signal chain without physical access to the microphone or potential disruption of the test setup.

Developed by GRAS Sound & Vibration—which like Audio Precision is now also part of Axiometrix Solutions—SysCheck2 is a system designed to actively verify whether the sensitivity of a microphone has drifted since its last calibration. In the first collaborative effort by the GRAS and Audio Precision engineering teams, APx500 release version 7.0 enables an APx user, in conjunction with an APx517B, APx1701, or GRAS 12Bx power module, to run SysCheck2 and receive a pass/fail indication for each connected SysCheck2-capable microphone, without the need of a calibrator or physical access to the microphone.

From a microphone in production test fixture to an array of microphones in an anechoic chamber, the combination of SysCheck2 and APx enables the user to verify the accuracy of the entire signal chain without physical access or potential disruption of the test setup.

Rub & Buzz Detection

In mid-2021, AP released the APx500 audio measurement software, version 6.1. With this update, AP added new measurement results, options, and test signals for production test of loudspeakers, headphone drivers, and microphones. For speaker designers and manufacturers the new APx software introduced new options for Rub & Buzz, SoneTrac, High-Order Harmonic Distortion, and Rub & Buzz Loudness defect detection methodologies.



In APx500 Measurement Software Version 6.1, Audio Precision introduced multiple Rub & Buzz (upper left) defect detection methodologies, including also SoneTrac (upper right), High-Order Harmonic Distortion (lower left), and Rub & Buzz Loudness (lower right).

Previously available in APx software, the Rub & Buzz method uses high-pass tracking filters to remove the fundamental signal and detect defects as excursions in the residual waveform's crest factor or peak ratio. SoneTrac is a Bose-developed method to improve Rub & Buzz measurement that filters the residual signal and ratios it to the RMS of the total signal to reduce the noisiness of the measurement result. High-Order Harmonic Distortion (HOHD) is a classic method for Rub & Buzz detection. It uses the total harmonic distortion (THD) ratio but only of harmonics above the 10th, 10-35, 20-200, etc., which is a simple way to account for frequency masking effects. Finally, Rub & Buzz Loudness applies a psycho-acoustic loudness model to the residual signal to calculate the perceived level of the Rub & Buzz based on well-established models of the human perception of sound.

To support the addition of the HOHD and Rub & Buzz Loudness measurements, AP introduced the Fast Sweep signal with APx500 release version 6.1. Fast Sweep, as its name implies, is an extremely fast stepped frequency sweep with no input or output ranging between steps, continuous transition between steps to minimize transient effects, and has a total sweep time that rivals the speed of a logarithmically swept sine (chirp) signal.

A few of the advantages of the Fast Sweep are the lack of transient ripples at low or high frequencies, the ability to sweep from low-to-high or high-to-low frequency, the support for measuring harmonics above the 20th, and precise control of the number of discrete points used in measurements.

New APx Version 7.0 Software Features

With release 7.0, APx software now enables the use of Fast Sweep in open-loop testing scenarios, such as the testing of smart speakers. Fast Sweep is the primary stimulus signal used when utilizing either HOHD or Rub & Buzz Loudness measurements.

Suited to anyone performing spectrum analysis, v7.0 brings several distinct improvements to the Fast Fourier Transform (FFT) analysis capabilities of APx software: Arbitrary lengths, windowless analysis, and level triggering. By allowing arbitrary FFT/DFT (Discrete Fourier Transform) lengths, users can now always find a combination of transform buffer and sample rate to arrive at convenient frequency resolution. With windowless analysis (where "Window" = None - move to bin center), engineers can perform synchronous, windowless FFT analysis on non-synchronous signals.

Power users who prefer Bench Mode, and those performing real-time adjustments on the devices under test (DUTs), will benefit from the enabling of Transfer Function in Bench Mode. In v7.0, these Bench Mode-oriented customers now have access to the Transfer Function measurement, and the complex frequency response function for a device, including the magnitude and phase (or real and imaginary components), using any broadband signal (e.g., speech, music, or noise).

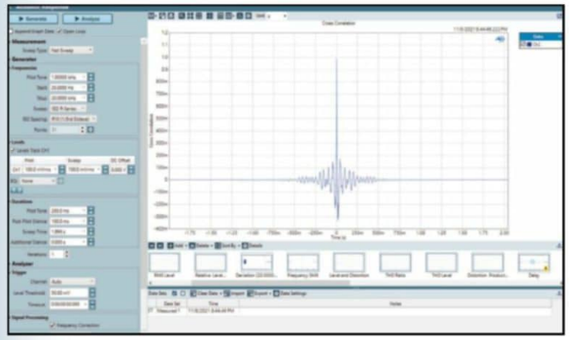
Software Subscriptions

To provide APx users greater flexibility in how they access APx500 measurement software, Audio Precision is introducing software subscriptions as an alternative to perpetual software licenses. While a perpetual software license—which grants a permanent, non-exclusive and non-expiring right to use the software—will still ship with each new analyzer, software subscriptions provide another path for existing APx users to access the latest software release and associated enhancements.

The new subscription models grant a time-limited (1-, 3-, or 5-year periods) license to use the software and provide access to all versions and all options (excluding PESQ and POLQA). When the subscription expires, the analyzer reverts to the software version and options for which it was previously licensed.

For users and organizations that prefer to retain perpetual licenses, software upgrades (SW-UPG), software maintenance (SW-MAINT), and extensions (SW-EXT, sold with new analyzers only) remain available, but only include software options purchased for a given analyzer. New hardware-based analyzers are always sold with a perpetual license, however APx500 Flex can be purchased with either a perpetual license or a subscription.

www.ap.com



With release 7.0, APx software now enables the use of Fast Sweep in open loop testing scenarios, such as the testing of smart speakers. Fast Sweep is the primary stimulus signal used when utilizing either High Order Harmonic Distortion (HOHD) or Rub & Buzz Loudness measurements.



Acoustic Response measurement using Open Loop Fast Sweep in APx500 version 7.0.

SysCheck2 Intelligent Acoustic Sensor System

Danish measurement microphone specialists GRAS Sound & Vibration, now a part of Axiometrix Solutions, announced two new microphones with SysCheck2 functionality allowing for remote-controlled health check of an entire measurement chain. The new microphones are the GRAS 246AE 1/2" CCP Free-field Standard SysCheck2 Microphone Set, and the GRAS 246AO 1/2" CCP Pressure Standard SysCheck2 Microphone Set, both featuring self-verification.

SysCheck2 is a GRAS-patented technology for verifying measurement chain integrity. It is a tool that allows remote, rapid health checks on microphones, channel gain, and cable integrity, which enables users to quickly validate their calibration with a single click and no physical interaction. The technology is easy to set up with standard cables making it adaptable for any test setup.

The verifications are made on each SysCheck2-enabled microphone connected to a CCP power module with transducer electronic data sheet (TEDS) support and measurement software with a single click. It also provides on-demand environmental data regarding temperature, barometric pressure, and humidity.

SysCheck2 functionality adds value to production line setups, where it will help reduce downtime as well as increase efficiency and data reliability. It is essential for test setups when microphones are placed in dangerous or hard-to-reach locations as well as for distributed measurement setups where many microphones need to be validated in a short period of time.

Users of Audio Precision APx analyzers who want an integrated solution that is easy to set up and provides end-to-end self-verification will benefit from SysCheck2 seamless integration to AP's analyzers for a plug-and-play solution.





The new GRAS 246AE 1/2" CCP Free-field Standard microphone and the GRAS 246AO 1/2" CCP Pressure Standard, both featuring SysCheck2 self-verification.

In the first collaborative effort by the GRAS and AP engineering teams, full SysCheck2 functionality is ready for use out of the box when connected to APx500 release v7.0 Measurement Software with GRAS 12BA, 12BB, or 12BE (coming soon) power modules and an Audio Precision APx series analyzer, or with an Audio Precision APx series analyzer with CCP and TEDS read/write capability.

GRAS also offers full functionality by way of providing a software development kit (SDK). This makes all features easily accessible with a suitable CCP-based power module, analyzer and data acquisition system, depending on the software platform.

New SysCheck2 Microphones

Coinciding with the introduction of SysCheck2, GRAS released two microphone sets: the new GRAS 246AE measurement microphone for general use, and the new GRAS 246AO model, more targeted for audiology applications. Both are 1/2" CCP microphone sets, which are improved versions of GRAS 46AE and GRAS 46AO—with the added SysCheck2 functionality. GRAS 246AE and GRAS 246AO can both be accessed with many data acquisition systems and offer seamless plug-and-play operation with Audio Precision's APx data acquisition and Siemens Simcenter Testlab software.

The GRAS 246AE with SysCheck2 is a 1/2" free-field microphone set and, as such, is optimized for all acoustic applications where the location of the main sound source is known, and the microphone can be pointed directly at it ensuring 0° incidence. It is ideal for general acoustic testing and conforms to IEC 61094 WS2F and IEC 61672 Class 1.

The GRAS 246AO microphone with SysCheck2 is a 1/2" high-precision pressure microphone suitable for laboratory work and ideal for production line testing in coupler setups for test of hearing aids, earphones, headphones and headsets. It conforms to IEC 63181.

www.grasacoustics.com



An externally polarized condenser measurement microphone cannot work on its own. It needs to be connected to hardware that can act as an interface between the raw signal output and the data acquisition unit or measurement device. GRAS Sound and Vibration added six new power modules to power CCP or LEMO measurement microphones with seamless integration of microphone sensitivity via TEDS. The GRAS 12BE, 12BC, 12BF, and 12BD microphone power modules have the ability to take advantage of the latest GRAS SysCheck2-enabled microphones.

NTi Audio

A Growing Ecosystem of Reliable Measurement Solutions

NTi Audio AG has been successfully growing, even during the difficult circumstances of the global pandemic. The company continues to expand its range of leading test and measurement solutions, including software development for its core products. In the company's recently expanded headquarters in Schaan, Liechtenstein, NTi Audio is seeing expanded development, production and sales activity, and the company received ISO14001 certification for its operations, underscoring its commitment to sustainability, with products that are as durable and as easy to repair as possible.

Among the reasons for the company resilience, its user-friendly and reliable product range continues to cover the demands in acoustics, audio, noise, and vibration applications, while more and more acoustics professionals are choosing to rely on NTi Audio for its calibrated and certified solutions, with worldwide support.

Since the company's founding, in 2000, following a management buyout from Neutrik of the existing Test Instruments department, NTi Audio established a strong reputation in the field of acoustic measurement technology with solutions that clearly met the needs of professionals globally. As an example, NTi Audio developed an entire software package for quality control of loudspeakers, which was based on a new, patented process for the automatic detection of audible defects. This decision helped NTi become a leading provider of quality control systems in the loudspeaker industry, with thousands of devices in daily use. The company also evolved its developments to meet acoustic measurements using portable audio measurement devices, originally by integrating a method for determining speech intelligibility in its first portable sound level meter, the Acoustilyzer and its successor, the XL2 handheld analyzer, which helped propel NTi Audio in acoustic, noise and vibration measurement applications.

The XL2 Audio and Acoustic Analyzer became a tool for acoustics experts as NTi expanded its support for professional Environmental Noise measurements and Room and Building Acoustics, with a complete ecosystem of calibrated hardware and certified software solutions.



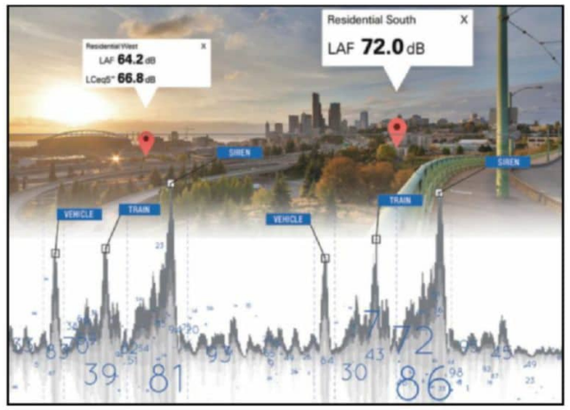
The NTi Audio team photo in 2020, celebrating 20 years of the company's founding, following a management buyout from Neutrik.

NTi Audio's most popular product, the XL2 Analyzer is a powerful Sound Level Meter supporting calibration according to the international IEC61672 standard. Each XL2 calibration is recorded with an individually numbered certificate, listing all relevant information about the tested sound level meter, and specifies the measuring equipment used during the test. The document also contains the detailed measurement results and an assessment of the test.

In order to reliably maintain the accuracy of a measuring system, periodic calibration is necessary. Many standards and quality systems require regular, documented calibration of the measuring equipment used; usually in a one-year cycle. NTi Audio offers this service, both at its headquarters in Schaan and at its international branches worldwide. Users can conveniently register a measuring device online for calibration and the website guides them to the nearest NTi Audio calibration center and provides information on costs and duration of service and transport.

AI for Noise Classification

NoiseScout is a complete noise monitoring solution from NTi Audio, which is also seeing increasing international adoption. Noise levels are recorded onsite by the XL2 Sound Level Meter and are available for remote monitoring and download. NoiseScout is aimed at both short-term noise assessments and long-term monitoring applications, with results displayed in a web browser and measurement data presented online in charts and dashboards. Multiple noise level meters can be monitored simultaneously within a map view, thus providing localized geographic visualization for all noise levels at a glance.



NoiseScout, the complete noise monitoring solution from NTi Audio, is now able to sample and compare sounds using AI and find pattern matches.

To assist in identifying the cause of a noise level alarm, NoiseScout now provides a text description of the content of the .wav files that are recorded during an alarm event. This is achieved using AI analysis and Noise Classification, which NTi Audio recently implemented. The AI-generated text indicates the possible cause of the alarm, and reduces the necessity to listen to the actual .wav file.

The .wav files of audio samples from intervals where levels were high are compared to a library of classified sound samples with a pre-trained audio event classifier that predicts audio events based on a dataset on the NTi Audio internal servers. The functions of this classifier form the central part of the newly implemented AI system. These functions use pattern matching to determine the closest matches between the current audio sample and the library.

The result is then weighted according to the sound pressure levels within the .wav file. Each pattern is assigned to predefined classes, and the class labels of the closest matches chosen. A score value is assigned to each match. The score is a measure of the accuracy and precision of a best-match classification aggregated across the whole .wav file. Scores are higher when there is less background noise. The text description and associated score are displayed when the score is greater than 20.

This AI analysis and Noise Classification assist in identifying the cause of noise level alarms, reducing the necessity to listen to the actual .wav file, and thereby saving time. Of course, the .wav files are still available for a more in-depth analysis.

New M2340 Class 1 Measurement Microphone

In 2021, NTi Audio announced a valuable new 1/2" measurement microphone option for the XL2. The new M2340 model is a development of the existing M2230, and is also a Class 1 measurement microphone, optimized for a flat, free-field response. In addition, the M2340 features self-verification circuitry and consumes significantly less power than the M2230. The M2340 contains the preamplifier and is an ideal omnidirectional solution for noise monitoring.

All microphones from NTi Audio for the XL2 Analyzer include an electronic datasheet. The Automated Sensor Detection (ASD) of the XL2 automatically reads this datasheet, and recognizes the microphone model, sensitivity, and calibration data. This ensures accurate measurement results.

But thanks to the System Self-test (Charge Injection Check—CIC) in the new M2340 microphone, the whole system used for measurements can be periodically verified and documented remotely. This is important in particular for stand-alone, remote-controlled measurement stations, which are usually employed for long-term level monitoring. Any damage to the monitoring microphone overtime can be detected using this self-test feature.

With the Remote Measurement interface, the XL2 makes all commands to control the M2340 and to query the measured values available to the user, while the monitoring system (e.g., NoiseScout) handles recording of the reference spectrum, regular verification and generating appropriate alarms in case of a significant level change or error.

Low-noise circuitry has a low impedance and therefore consumes more power. The new NTi M2340 measurement microphone achieves a good compromise between power consumption and low noise. According to NTi Audio, compared to the M2230, the current consumption has been reduced by 66% to 0.8mA in the M2340. The noise floor increases by only 1dB, while the maximum processable sound pressure level has been increased by 1dB. This makes the M2340 most suitable for power-saving applications such as long-term measurement stations.

Software Improvements for XL2

NTi Audio's ongoing customer-driven policy of regularly introducing free functional updates to the handheld XL2 continues to increase the value for this already comprehensive and powerful portable solution. All XL2 Sound Level Meter owners received added support for the System Self-test (CIC) function of the new 1/2" M2340 Measuring Microphone.

Another innovation in the area of outdoor monitoring was the integration of the correction values for horizontal sound incidence when using the M4261 Measuring Microphone together with the WP61 Weather Protection kit. This means that in addition to the proven WP30 for 1/2" microphones (type M230 and M2340), a solution is now available that fits the M4261 Class 2 microphone.

The newly introduced support of the ANSI/ASA S12.2-2019 standard for noise curve measurements provides a solution for another field of application. This update is especially useful for users in the American and Asian regions that want to analyze noise in offices, etc.

And the XL2 Vibration Meter also recently received an update with an optimization of the automatic naming of recorded wave files. This is useful when analyzing data, as it makes it easy to detect, for example, overdrive during the measurement, and prevent undetected incorrect data.

Finally, if the Extended Acoustic Pack option is activated on the XL2, users are given the choice to select an A-weighted level, limited to the frequency range 20...200 Hz. This level can be used, for example, as an indicator of low-frequency sound at open-air events.

www.nti-audio.com



New NTi Audio M2340 Class 1 measurement microphone for the XL2



All XL2 Sound Level Meter owners continue to receive free functional updates that increase the value of this portable measurement solution.

17463759

Klippel

The New Klippel SCN Near Field Add-On (SCN-NF)

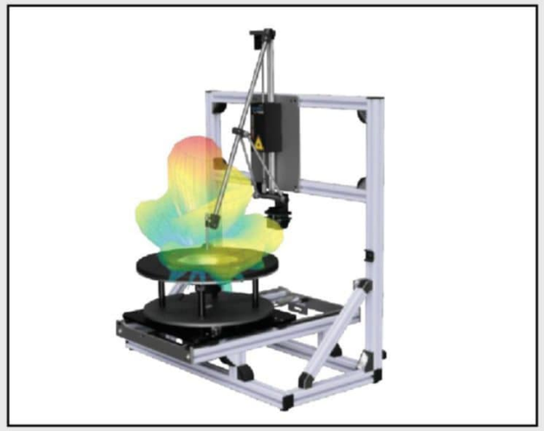
In 2021, Klippel released its new SCN Near Field Add-On (SCN-NF), which accurately performs all the most relevant transducer measurements on a single hardware platform.

The hardware add-on (SCN-NF) upgrades the popular Klippel Vibration Scanner hardware (SCN) to a Multi-Scanning Workbench for scanning with other useful sensors (e.g., microphones, probes, and magnetic sensors). Acoustic directivity scans can be done in normal (non-anechoic) rooms using the holographic Near Field Scanning (NFS) technology, while the large Near-Field Scanner ("Carousel") allows both full- and half-space measurements, and the Multi-Scanning Workbench is focused on half-space configuration (using a round baffle).

This new powerful addition to the Klippel Scanning Vibrometer is a smart alternative to using an anechoic chamber, and the fastest way to get full directivity holographic measurements, extrapolate SPL to any point in 3D half-space and obtain balloon, contour, polar, and sound power plots.

The radiated direct sound into half-space is determined with high accuracy based on acoustic holographic methods also used in the Klippel NFS Near Field Scanner System, allowing the most relevant transducer measurements can now be done on the same hardware, in an average office room.

The SCN-NF adds 2pi acoustical measurement capabilities to the same Scanning Vibrometer platform used for laser scan diaphragm vibration. The SCN hardware is extended with a microphone in addition to the existing laser sensor. Automated axis control ensures repeatable, precise and fast positioning of the microphone and laser sensors, in combination with a round baffle for measuring transducers up to 10" (30cm) in diameter, or compact (smart) speakers.



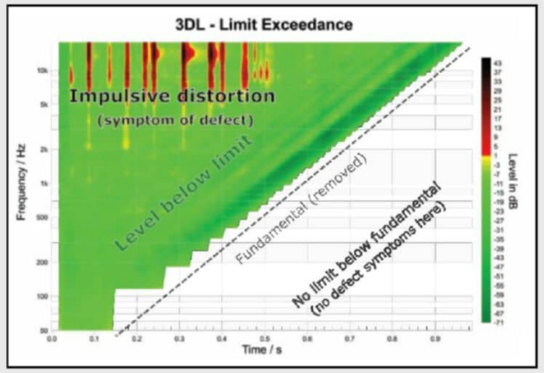
SCN Scanning Vibrometer with the SCN Near Field Add-On. A round baffle allows removing acoustic shortcut and edge diffractions, reducing baffle vibrations to almost zero, and exploiting symmetry to speed up testing, among other benefits.

Holographic measurements use a model based on spherical wave components, weighted by spherical harmonics and frequency dependent coefficients, providing a full solution for the acoustic wave equation. For this holographic model, multiple near-field sound pressure measurements are performed on the plane surrounding the sound source and used as boundary values. This holographic model then describes the radiated acoustic wavefield of the measured sound source in a tridimensional format. The multifunctional measurement device supports the IEC 60268-21 and IEC 60268-22 standards.

Klippel New dB-Lab Version 212

The Klippel dB-Lab software is offered in dedicated versions serving each application area. The latest major software update to its R&D and Quality Control (QC) platforms is an important expansion of the available tools in particular for speaker measurements, and brings many small but useful tools and updates to existing modules. The main software platform dB-Lab 212 now provides a shared sensor management with support for Klippel Multi-Scanning Workbench (formerly SCN) hardware, providing directivity, sound power, and room correction. The new MTON module offers comprehensive measurement of distortion, while the new linear simulation module LSIM complements the available simulation tools with more options for speaker and enclosure design.

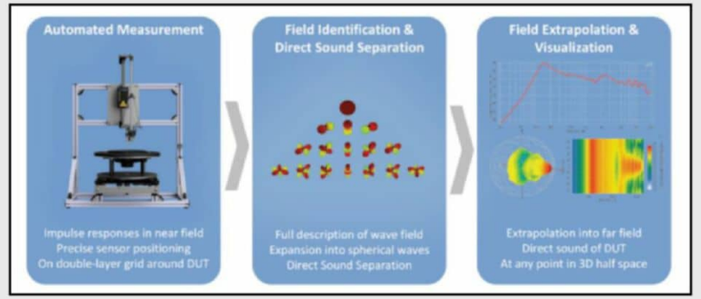
For Quality Control, the Klippel QC software has also been upgraded with a new automation control interface and multichannel support for any Windows or ASIO audio interface, as well as wave-file-based open-loop testing.



Detected loose particle failure in 3D spectrogram limit surface; limit area defined by harmonic order of chirp signal.

The updated software includes more flexible options for testing and synchronization and for smart or stand-alone audio devices.

Using the new Automation API software interface, complex test scenarios can be implemented and controlled. This API is the successor of the long existing IO-Monitor interface, which is still supported. It can be integrated easily in popular script languages (e.g., Python). The before-mentioned sensor management routine is available for Klippel analyzers, sound-card-based interfaces, digital audio devices and wave file analysis. Sensor files can be shared with R&D software and result charts are correspondingly scaled and labeled.

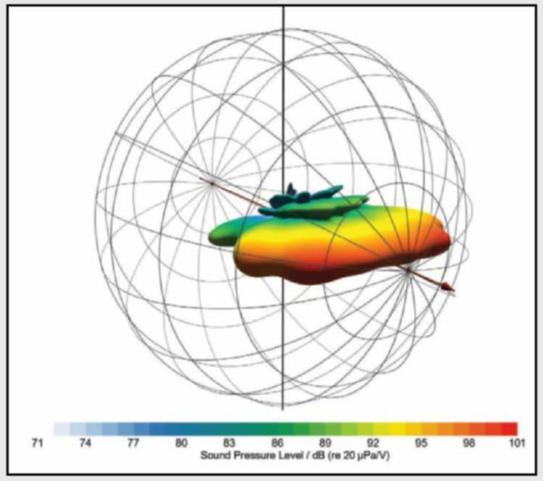


Methods and results using holographic measurements with the Klippel Multi-Scanning Workbench.

2022 Updates

Already in 2022, Klippel has released a first minor software update (free of charge) for the Klippel Analyzer System introducing new features for both the dB-Lab 212 and QC7 (R&D and QC) applications. The Time Frequency Analysis (TFA) module is dedicated to the analysis of waveform signals from Klippel test data or any imported .wav file in time and frequency domain and now it received a major revision and feature update. In addition to the spectrogram analysis options based on Short-Time Fourier Transform (STFT), wavelet or auditory filter banks with superior time resolution, this powerful post-processing tool now also provides an interactive bandpass filter and playback function for subjective distortion analysis and diagnostics. Replacing the obsolete PLAY module, the TFA can now load even long .wav files in order to obtain signal characteristics and pick a time range for detailed analysis. Overall performance and usability were improved.

Now officially released, the QC 3D Spectrogram Limits (3DL) is an offspring of the TFA, dedicated to the particular requirements of QC testing. This add-on for the chirp-based Sound Pressure Task (SPL) of the Klippel QC software uses the auditory filter bank shared with the TFA to generate a time-frequency plot of the measured DUT response. Applying automatically generated "3D spectrogram limits" based on golden reference DUTs reveal the signature of any irregular distortion and abnormal sound caused by loudspeaker defects. Any values exceeding the limit threshold are highlighted clearly in the spectrogram. In addition to the well-established Rub & Buzz measurement in the time domain (impulsive distortion, IEC 60268-21), the 3DL provides new benefits, such as detecting abnormal behavior anywhere in the spectrum, identifying external (uncorrelated) disturbance, detailed defect root cause analysis, as well simplified Rub & Buzz filter settings.

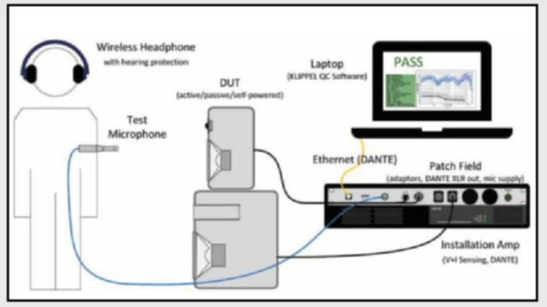


Balloon plot generated with the new NFS visualization in dB-Lab

The QC Stand-alone Software version is dedicated to all testing applications that do not require the Klippel Analyzer Hardware. Together with the External Synchronization (SYN) trigger add-on, this solutions is highly suitable for any closed loop (third-party test frontend) and open-loop (.wav-file based) test scenario. It only requires any device running Windows and a Klippel USB license dongle.

In addition to output-based audio testing according to IEC 60268-21, the QC Stand-alone solution now also supports all remaining tasks of the QC software based on voltage and current measurements such as Impedance (IMP) for Thiele-Small (T-S) parameter testing and Motor + Suspension Check (MSC) for patented nonlinear parameter testing (e.g., voice coil offset). This allows creating powerful and yet price-efficient test setups using smart amplifiers with built-in voltage and current sensing, such as the Powersoft Mezzo series, using a Dante network connection for professional quality control in cost-sensitive applications (e.g., rental companies, service stations, etc.).

www.klippel.de



Klippel QC Stand-alone Software setup example

Outline

New Speaker and Acoustic Measurement Solutions

Recognized globally for its innovation and manufacturing quality, Italian loudspeaker manufacturer Outline is a respected name in the global sound reinforcement industry. Founded in the northern Italian city of Brescia in 1973 by Guido Noselli, Outline's continued development of loudspeaker technology created numerous innovations and milestone designs in professional audio.

As a result of five decades of continuous research, Outline has produced the ET250-3D turntable, a "transversal" measurement instrument that provides the data for precise analysis of a loudspeaker's characteristics. Initially conceived for in-house use exclusively, the turntable design evolved to become an industry standard, in daily use by some of the world's largest technology companies and also by many loudspeaker manufacturers, as well as universities and research institutes throughout the world.

High-profile users include Apple, Google, Amazon, Microsoft, Intel, Lucasfilm, Dolby, MIT, and many more, where it is used on a daily basis to measure everything from antennae, solar panels, hearing aids, and smart devices to loudspeaker systems and transducers, mobile comms devices, microphones, domestic appliances, and even medical equipment.

The original turntable concept of a high-precision, computer-controlled turntable capable of placing significant loads with perfect accuracy in a measurement environment has now been further developed, with the creation of two new models to suit different speed and load requirements.

The Outline ET50-3D and ET250R2-3D turntables both feature a new Ergal aluminum hardware design, which achieves military-spec mechanical resistance while considerably reducing the overall weight. Additionally, both feature a wider choice of rotation speeds with enhanced maximum handling loads and a simple user interface with an input/output communications port based on TTL signals. However, the key to their accessibility, data acquisition, and control is the network interface, allowing remote control via standard Ethernet protocols—an area that Outline always pioneered for its speakers and line array systems.



Outline's ET System measurement turntables are available in the ET50-3D and the ET250R2-3D models to suit different speed and load requirements.

Each turntable has an individual IP address and the user simply connects this to an existing LAN to control it via the provided software or via one of several third-party measurement systems. Control libraries for Matlab and Python are available and both devices are controllable by any software with Ethernet control capabilities.

This facilitates automated polar measurements via seamless integration with various measurement software suites, including Audio Precision, Audiomatica's Clio, Listen Inc.'s SoundCheck, Klippel, NTi Audio, and ARTA. The combination of Outline hardware and the user's choice of control software enables precision placement of the object under test, with repeatability and accuracy of the test results.

As with all Outline products, the ET50-3D and the ET250R2-3D measurement turntables are engineered and manufactured to the highest standards, with the models capable of safely and accurately orienting loads of up to 200kg (440 lb.) and 1000kg (2200 lb.) respectively.

Globe Source Radiator

Designed in collaboration with the Department of Mechanical And Industrial Engineering at the prestigious Brescia University in Lombardy, and engineered in Italy by Outline, the new Globe Source Radiator (GSR) high-power omnidirectional sound source solution provides a vital tool for acoustic analysis. Complemented with the Outline NG01 noise generator that can be controlled with a radio remote control, and the portable Outline L3000 DSP amplifier, the GSR offers a high-SPL solution for acoustics measurements in all environments, including large venues and spaces.



The Outline Globe Source Radiator (GSR) is a high-power omnidirectional sound source that helps eliminate diffractions, reflections, and other anomalies, which can compromise the accuracy of data measurements.

The Outline GSR is a high-power omnidirectional sound source that delivers a perfectly spherical and physically robust design, generating a homogeneous sound field from its specially shaped loudspeaker membranes to assure perfect isotropic directivity.

Numerous essential acoustic parameters may be derived from measurements made using it as a sound source. These include Clarity Indices (C50, C80), Intelligibility Indices (AlCons, STI, RASTI, etc.), Reverberation Time (RT60), Early Decay Time (EDT), Soundproofing Power, Impulse Response and Intensity (G), and others. The Outline GSR solution is also fully ISO10140, ISO354, ISO3382, and DIN 52210 compliant.



The Outline GSR kit includes the NG01 Noise Generator and L3000 DSP compact amplifier.

Crafted out of top-quality void-free birch ply and finished in super-tough polyurea coating technology, the robust spherical enclosure features an Ergal aluminum inner structure that houses 12 5" neodymium-magnet transducers and is the first such device to include a bass reflex design to extend its frequency response. The GSR's custom-designed transducers feature high power-to-size ratio Neodymium magnets with additional ferrofluid cooling, thus increasing thermal power dissipation and making it possible for the GSR to handle up to double the power of conventional loudspeakers.

Rated for 1000W RMS (4000W peak) total power handling, the Outline GSR is capable of producing a maximum SPL of over 130.3 dB, a figure far in advance of any other comparable product presently available and a remarkable output from a spherical loudspeaker only 31cm in diameter.

The GSR is also available as a part of a complete user-ready kit, comprising a GSR, NG01 Noise Generator, Outline's L3000 DSP compact amplifier plus all the necessary cables, a transportation bag, and a set of mounting accessories, providing a portable setup ready for immediate deployment. The Outline NG01 noise generator can be controlled with a radio remote control and delivers White or Pink noise with XLR balanced outputs. A dedicated self-powered subwoofer kit is also available and is suitable for applications where measurements of low frequency data are required.

www.outline.it/measurement



Ellisys

Advanced Bluetooth LE Audio Capture and Analysis Features

Ellisys, a leading worldwide provider of test and analysis solutions for Bluetooth, Wi-Fi, USB, Thunderbolt, and other wireless and wired communications technologies, supports the development of emerging Bluetooth LE Audio products, including controllers, stacks, and devices. Since the announcement of the Bluetooth LE Audio specifications in January 2020, the company's Bluetooth protocol analysis systems have been continuously expanded to support developers.

Ellisys Bluetooth test and analysis solutions are used by developers and test labs worldwide. The company's solutions include the Ellisys Bluetooth Qualifier (EBQ) platform, and several protocol analyzer tools supporting both Bluetooth radio types—Low Energy and Classic (BR/EDR). EBQ is a comprehensive compliance, validation, and development system for Bluetooth technology, targeting the behaviors of the lower communications layers, including implementation of nearly 2,000 test cases. Ellisys protocol analyzers include the ubiquitous Tracker, Explorer, and Vanguard systems, offering deep features sets designed to meet a variety of requirements.

Features supporting Bluetooth LE Audio specifications include the new tzero tracking technology for full and accurate capture of isochronous traffic on connected and broadcast links, an innovative auto-detection capability built into a test equipment grade integration of the power-friendly LC3 audio codec, and support for several new audio profiles.

LC3 and CNew Specs

The Low Complexity Communications Codec, or LC3, is particularly ideal for Bluetooth Low Energy as it provides a high degree of quality, even at its lowest data rates. Support available on all Ellisys Bluetooth analyzer models, includes detailed decoding for LC3 traffic, together with a new feature, based on an Ellisys-designed, test equipment grade LC3 codec, which allows for automatic determination of advertised LC3 configuration parameters.

Historically, test equipment implementations have required a complete and error-free capture of (wirelessly transmitted) audio codec configuration parameters to properly capture, characterize, and replay audio. With this auto-detect innovation, even with otherwise critical configuration packets corrupted by interferences or low signal strength, LC3 audio is still recognized, understood, captured, and available for further analysis.

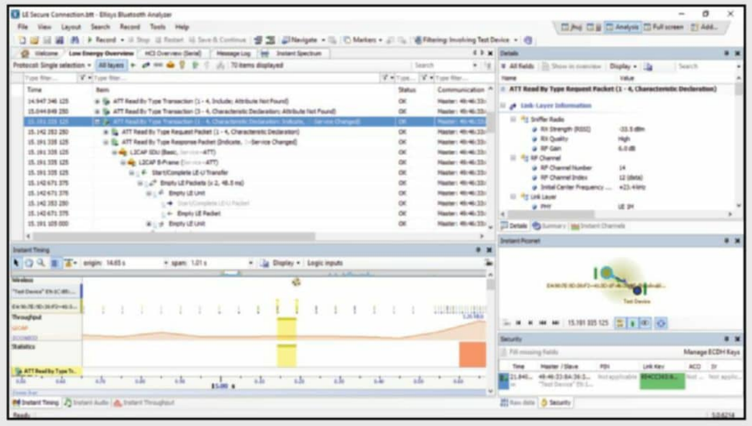
Additionally, Ellisys added support for several recently released audio specifications that define services and profiles that can operate over Bluetooth Low Energy atop its GATT protocol. These include:

- Audio Input Control Service (AICS)
- Volume Control Service (VCS)
- Volume Offset Control Service (VOCS)
- Basic Audio Profile (BAP)
- Audio Stream Control Service (ASCS)
- Media Control Service (MCS)
- Telephone Bearer Service (TBS)
- Coordinated Set Identification Service (CSIS)

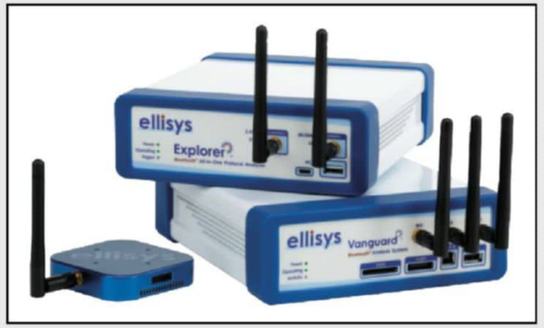
Bluetooth 5.3 Specification

With the release of version 5.3 of the Bluetooth Core Specification in July 2021, Ellisys also released a new suite of validation tests, called the Quality Suite, that exceed the scope of qualification test requirements defined by the Bluetooth Special Interest Group (SIG).

This latest version of the specification adds several new core features, including power-saving enhancements, security updates, and channel selection improvements. With the initial release of the Quality Suite, Ellisys extended its complement to nearly 2,000 tests. While adherence to the requirements defined in the latest Test Case Reference List (TCRL) from the Bluetooth SIG, goes a long way toward ensuring Bluetooth interoperability and performance goals, the addition of the Quality Suite takes that a step further by covering test areas not defined by the TCRL. Many of these tests are implemented based on suggestions from the Ellisys customer base, and most focus on later-generation Bluetooth features.



Ellisys Bluetooth Tracker Software



Both test suites are available on the Ellisys Bluetooth Qualifier (EBQ) dual-mode radio controller development and qualification test system. The EBQ is available from stock to Bluetooth SIG-recognized test labs, known as Bluetooth Qualification Test Facilities (BQTF) and Bluetooth Recognized Test Facilities (BRTF), and to Bluetooth SIG member companies involved with radio controller and IP development. Ellisys protocol analyzer systems are available from stock either direct from Ellisys or from authorized distributors worldwide.

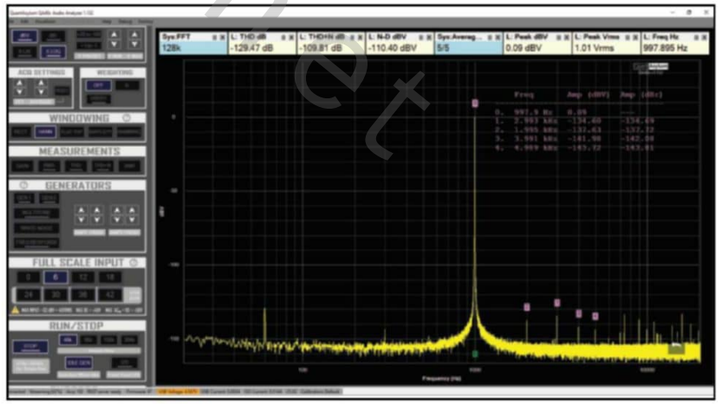
www.ellisys.com

QuantAsylum

Fourth Generation Audio Analyzer

QuantAsylum, the manufacturer of audio test and measurement equipment based in the Seattle, WA area, announced the release of the QA403 Audio Analyzer. This is the fourth audio analyzer released by the company since 2012 and shows the ability of its founder, Matt Taylor, to understand the needs of the industry, managing well the difficult circumstances of the disruptions in the supply-chain caused by the global pandemic. As with many other manufacturers, QuantAsylum was severely constrained by silicon shortages in 2021, and expects that to continue through 2022. The design of the QA403 Audio Analyzer takes components constraints into consideration to minimize the risk.





The QA403 software package allows quick access to common measurements with the main controls contained within a single graphical panel.

The new QuantAsylum QA403 design delivers up to +18dBV output (8Vrms, or 16Vrms balanced) in four ranges and accepts inputs up to +32dBV in eight ranges. Software is included and supports all common audio tests, including SNR, THD, THD+N, Frequency Response, RMS and amplitude measurements. The seamless environment means compact analysis without calibration in a USB-powered hardware package measuring just 177mm (W) × 44mm (H) × 97mm (D) or (7" × 1.75" × 4") and about 450 grams.

Front panel connectors include differential inputs and outputs and an expansion connector permitting I²S and I²C control of customer board-level hardware. Typical performance figures are about -110dB for THD+N, and -120dB for THD in loopback.

The target market for the product is companies wishing to put a programmable and cost-effective audio analyzer on more of their engineer's desks or in the factory. The software doesn't require a license, making it easy to move setups between home and work, or between the lab and the factory floor.

Following QuantAsylum's philosophy, programmability also makes it easy to write test applications in C# or Java or Python, or any language that can perform HTTP GET and PUTs. Additionally, built-in tests make it easy to perform a range of swept measurements quickly without any coding.

At the time of writing for this edition of *audioXpress*, shipping is expected to start in the second half of February 2022 and the company is taking orders placed on the website. The price is $549 USD.

www.QuantAsylum.com

audioXpress

ZUMI Systems

Headphone Testing for Development and Production

ZUMI Systems, a company based in Sendai, Japan, has been focused on providing complete measurement hardware and software systems targeting the needs of product managers and production quality control at a competitive price. The company's range of audio test systems are being used extensively in production testing with Tier One headphone factories in China.

ZUMI's headphone test and measurement platform is based on a purpose built proprietary embedded system optimized for these demanding requirements. This platform allows ZUMI to design testing systems adapted to very specific products at a lower cost.

Since 2018, ZUMI delivered its latest generation HPA85 test system, a fully integrated measurement instrument for headphones and earphones, which can be easily configured for different form-factors and using different ear couplers and simulators. The system executes complete QC measurements in a short amount of time and can automatically process ANC measurements and trimming. Best of all, it does not require advanced training to use.

As ZUMI Systems intended, this self-contained test solution is proving to be easy to use for production line testing and quality control, as well as running standard tests for product development.

In Daily Use

Based on testimony from multiple brands and factories currently using the ZUMI HPA85 test system in China, the solution has now been widely proven with multiple product categories and volumes, and is also used for product development situations, such as the characterization of filters for frequency response and ANC optimization.

ZUMI Systems' products are distributed by dB Enterprises, operating from Hong Kong, and the company's CEO David Lindberg chatted with some of the companies using the ZUMI HPA85 solution in China.

ONANOFF is a company based in Hong Kong, that focuses on essential, easy-to-use electronic accessories and lately is focused on headphones for children—a product category with very stringent requirements. Kieran Pollard, Brand Manager, chose ZUMI systems as its primary tool for development of wired, Bluetooth, and ANC feedback headphones and is now working on new hybrid ANC designs that have even more demanding specifications with which the HPA85 is able to meet, including testing prototypes for Bluetooth performance, battery life, and ANC feedback, feedforward, and hybrid profiles in a single system.

ONANOFF uses the ZUMI HPA85 at its headquarters in Hong Kong and at its factory partners. In those production operations, the systems benefit from the ability to log system activity and measurement data for analysis and quality control.

"ONANOFF is bringing a lot of new products to market every year. With ZUMI, I can personally switch on the system and be testing within 5 seconds. Apart from development tasks, it allows me to measure competitive products and produce measurements for marketing," Pollard explains. "A good example of this would be earpad qualification. It is a headphone part that has key performance in acoustic, comfort and cosmetics. I can get a bunch of samples from our earpad vendors, discuss the cosmetic and comfort decisions with the industrial design team, and measure them in the lab for the acoustic seal."



The ZUMI HPA85 headphone test system in use in a factory in China



Production line testing and quality control of headphones from ONANOFF



Testing planar magnetic headphones at TrueAnalog Strictly OEM in Guangzhou, China

"We now use the ZUMI system across the factories we work with, because we can have a non-technical person trained to produce a frequency response curve and benchmark or compare samples in 20 minutes. It is convenient and easy to use."

The only pending request from ONANOFF is for the system to have a connection for a (larger) external monitor, which is something that will soon be available, via USB.

Other ZUMI clients report the same ease of use and convenience and recommend it for use in production. Colin Ward of 1dBFs, a contracted engineering manager, uses it for frequency response testing, and phase testing. "The ZUMI is a basic tool that is very easy to deploy in factories and get operators testing quickly. It's easy to set up and use, delivers clear results—quick pass/fail criteria and understanding the errors if there are any—and is low cost. You do not need to be an engineer to use it or to even program things like parameters. In short, it is a useful tool that works very well in mass production applications."

Another testimonial comes from Philip Richardson, the CEO of TrueAnalog Strictly OEM, a loudspeaker driver and system manufacturer based in Guangzhou, China. The company specializes in the design, engineering, and manufacturing of driver units and turnkey systems. The company has been using the ZUMI HPA85 for development and testing of a planar magnetic driver for headphones. Before production, TrueAnalog used it to customize the headphone drivers to customer specifications and now it uses more units for production line test and quality control. "ZUMI is well oriented for use by a production line worker that can press a button and go. What I like about it is that the microphones are built into the head unit and we can just slide the headphones on easily. We use it for frequency response, and I like that it gives us an immediate comparison of the left and right signal in a system that is color coded. Plus, we can add our tolerances so in production the operators just watch for the red or green light. We have to say that we are overall impressed with this unit. It makes our life easier."

www.db-ent.com
www.zumisystems.com

Listen, Inc.

New SoundCheck Updates and AmpConnect 621 Multichannel Audio Interface

In 2021, the SoundCheck 19 release of Listen, Inc.'s flagship audio test and measurement system added powerful advancements for users making multichannel measurements, including a new multichannel RTA and WASAPI-exclusive mode. The inclusion of room acoustics measurements offers increased capability for those looking to evaluate devices in real-world situations. Many of these features also add value on the production line, minimizing setup time, reducing the chance of user-error, and ensuring smoother operation in automated situations. The SoundCheck 19 announcement also coincided with the availability of Listen's new high resolution multichannel interface, AmpConnect 621.



In 2021, Listen introduced important updates to SoundCheck and the new AmpConnect 621 Hi-Res Multichannel Audio Interface.

In the updated software, the multichannel RTA feature allows displaying multiple channels or signal paths and the spectrum settings such as gain, octave band, weighting, averaging and time-weighting, all independently configured for each channel for complete measurement flexibility.

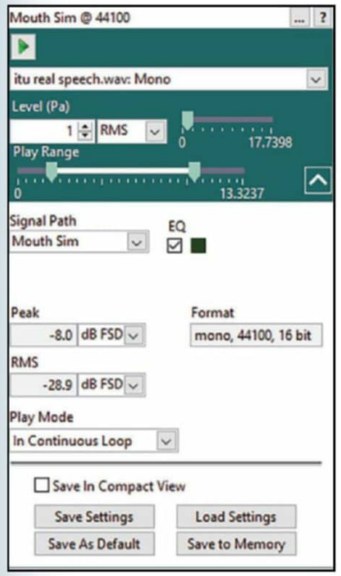
A power averaging calculation on the signal paths can be used for spatial and sound power measurements and all multichannel RTA functionality can also be used within a test sequence for automation. The ability to display multiple channels or signal paths presents a convenient view of many measurements across a wide range of applications. These include overlaying left and

right earphone responses to check for fit and seal, microphone array captures (e.g., in-vehicle cabin measurements), real-time adjustment of active noise cancellation filter, and more.

The new RT60 Room acoustics module in SoundCheck 19 offers extremely fast measurement of the reverberation time (signal decay time) and clarity (ratio of early reverberations to late reverberations) of any room or space. This functionality is useful for quantifying how speakers will interact with the acoustics of a particular type of room. The integration of the RT60 functionality within SoundCheck provides the user the assurance that measurements can be performed with calibrated signal paths.



The new AmpConnect 621 multichannel, multifunction, high-resolution audio test interface includes all the functionality required for audio testing in just one unit, including an amplifier, microphone power supply, line outputs, and digital I/O. And with a sample rate of up to 192kHz, it supports measuring high-resolution audio designs.



SoundCheck's new Signal generator

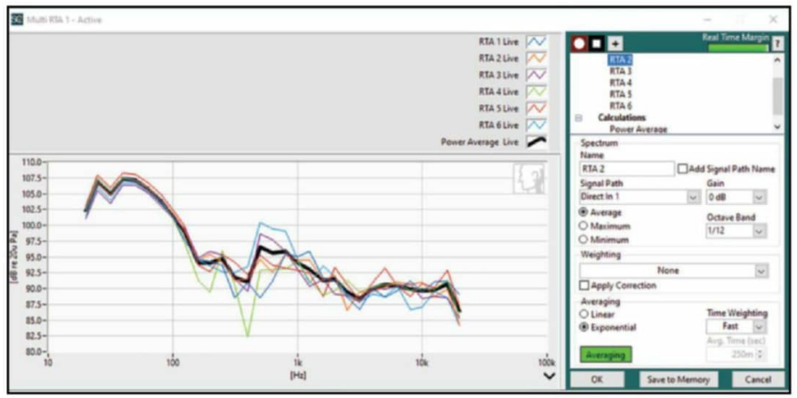
Another SoundCheck 19 enhancement is the option to use WASAPI-exclusive mode with Windows 10 audio devices. This advanced interface offers superior communication with audio devices, allowing SoundCheck to bypass the Windows audio engine, and send audio streams directly to the device. This delivers lower latency and ensures that sample rate conversion from the built-in Windows audio engine will not occur. It also enables SoundCheck to put an exclusive lock on the audio interface so that a measurement cannot be disrupted by system sounds. The ability to use multiple WASAPI devices simultaneously offers significant advantages for multichannel applications, particularly where the channel count requires multiple audio interfaces. WASAPI and ASIO devices can be combined to increase channel availability.

In both Windows and MacOS, SoundCheck now allows the Dock USB Controller Knob to be used to adjust the SoundCheck Signal Generator, simulating the feel and control of operation of a traditional manual analog signal generator. Used with other virtual instruments such as the RTA, spectrum analyzer and distortion meter, this offers a rapid way to identify defects by observing the response while manually fine-tuning the signal generator.

Listen AmpConnect 621

AmpConnect 621 is Listen's new high-resolution (up to 192kHz) multichannel audio interface, specifically developed for test applications such as automotive measurements with a six-microphone array, sound power measurements, or for measuring devices with multiple speakers (e.g., smartphones or soundbars). And it is also a great choice for measuring headphones and hearables.

The new AmpConnect 621 was designed with a clean front panel, with no buttons or control knobs, and showing only LED status indicators. It features six inputs, two line outputs, and one amplifier output. All inputs offer microphone power (SCM, IEPE and pre-polarized), to accommodate many types of couplers, accelerometers and reference microphones—even the new Brüel and Kjær 5128 high-resolution head-and-torso simulator.



Power Averaging shown on SoundCheck's new Multichannel RTA, introduced in SoundCheck 19, and now upgraded with real-time calculations

In addition, AmpConnect 621 also includes microphone power, full TEDS calibration support, impedance measurement and digital I/O in a single USB-connected unit. Designed for seamless integration with SoundCheck, it is controlled exclusively via the SoundCheck software. True plug-and-play makes configuration automatic and extremely fast whether it is a single installation or a large scale factory implementation; and there are no front panel buttons, eliminating the possibility of accidental changes to the settings.

Setup and configuration of AmpConnect 621 is extremely simple and fast with full plug-and-play functionality and TEDS support for automatic identification, configuration and calibration of microphones and accelerometers.

The two line level outputs plus a separate 50W amplifier route the output signal to any sources, powered or needing amplification, (e.g., artificial mouths, speakers, headphones, and automotive head units).

Impedance measurements using the amplifier output, along with output voltage monitoring, are performed entirely internally and do not require using any of the six external input channels, leaving them open for other measurements. This, along with digital I/O, simplifies operation as signals are all routed internally with no additional cabling required. The device connects to the computer with a single USB cable; the only additional cables required are those connecting the microphones and DUTs. This all-in-one approach offers improved ground loop immunity in comparison to discrete components.

As with Listen's other hardware products, AmpConnect 621 is built to last in production environments. The 2U rack-mountable unit is rugged with a new fingerprint-resistant matte finish.

SoundCheck 20 Introduces Perceptual Rub & Buzz Algorithm

Scheduled to be available by the time this edition of *audioXpress* is distributed, the new SoundCheck version 20 is the culmination of two years of original research by the Boston, MA-based audio measurement company and places this solution ahead of the industry in scope and depth of features. SoundCheck 20 contains a wealth of new features and enhancements for both production line and R&D users, particularly those making multichannel measurements. Significant developments include a new enhanced perceptual Rub & Buzz detection algorithm, which offers unrivaled repeatability and reliability.

The new enhanced perceptual Rub & Buzz (ePRB) algorithm is based on Listen's industry-standard 2011 Perceptual Rub & Buzz algorithm, with refinements that offer improved listener correlation and significantly better noise rejection. This makes limit setting easier and increases reliability, especially in production environments. This algorithm measures perceptual Rub & Buzz at the same time as all other end-of-line measurements with a single stepped sine wave stimulus, enabling this valuable analysis to be added to production line tests with no extension in test time.

Meanwhile, the multichannel RTA, initially introduced in 2020, was upgraded with real-time calculations such as curve addition, subtraction and averaging. This is useful for real-time multichannel testing such as power averaging of in-car maximum SPL measurements, and viewing ANC attenuation in real time. Static and live curves can be shown simultaneously on the same axes or on multiple graphs.

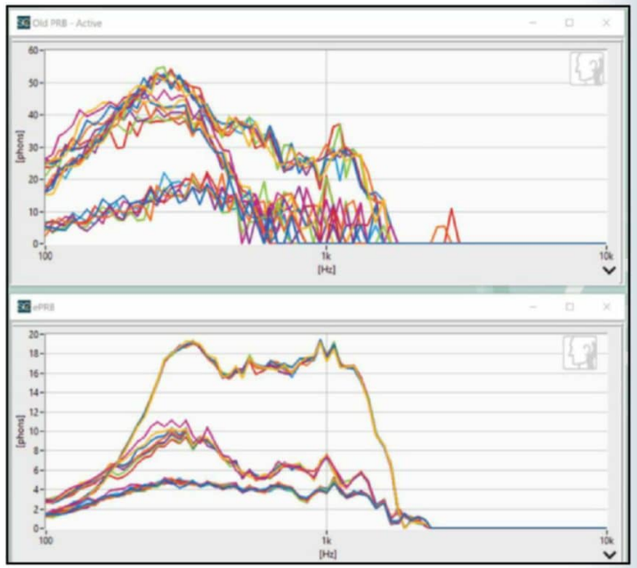
Updated Signal Generator, POLQA, and Communication Testing

A redesigned signal generator offers a new user interface for easier operation, and additional functionality, particularly for users testing with WAV files such as speech and music. For example, the start and stop time of a .wav file can now be selected either using time, or an analog slider. This makes it fast and simple to review particular artifacts of a long waveform. Precise user determination of the delay between multiple signal generators enables phase to be accurately controlled and accounts for the distance between loudspeakers.

An optional Perceptual Objective Listening Quality Analysis (POLQA) module brings the POLQA algorithm right into SoundCheck, where it can be used for perceptual measurements of speech degradation in communication applications ranging from telephones to smart devices. It may be used to assess the impact of noise reduction algorithms, evaluate Bluetooth degradation due to packet loss, or to analyze distortions introduced into the audio path.

New stimulus steps and post-processing capabilities have also been included to support Doubletalk measurements on communications devices. These include a silence stimulus step, and RMS level versus time and histogram post-processing steps. A pre-written test sequence for measuring Doubletalk is also available. Written by telecoms expert John Bareham, this sequence categorizes Doubletalk performance to two international standards, ITU-T P.502 and ETSI TS 126 132.

www.listeninc.com



Comparison of Listen's new Enhanced Perceptual Rub & Buzz (ePRB) Algorithm (lower) with the original Perceptual Rub & Buzz (PRB) Algorithm (top)

Practical Test & Measurement

REST for Automated Test and Measurement

This article makes the case for using web interfaces and embracing web standards for automated testing of audio equipment, fundamentally exploring the possibilities of REST—which stands for Representational State Transfer—a widely accepted set of guidelines for creating reliable web APIs.

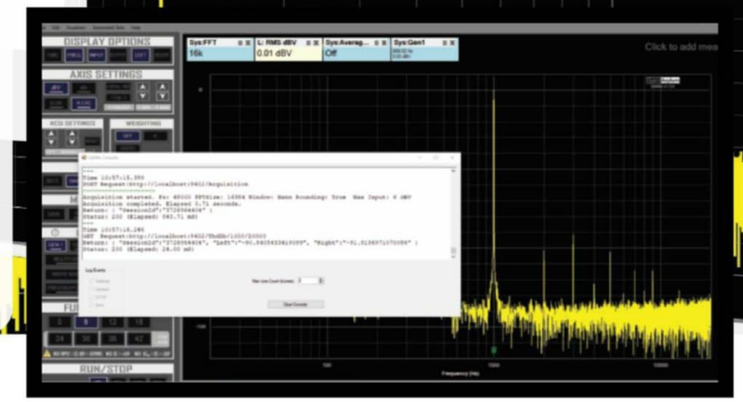


Photo 1: This is the QuantAsylum QA402 audio analyzer application, with a debug console on top, allowing you to see the processing of each HTTP request as the requests come across from the Jupyter application (or any application you might write). This makes debugging automation straightforward.

By
Matt Taylor

Over the decades, test and measurement solutions for manufacturing have evolved. HP Interface Bus (HP-IB) was born in the late 1960s to allow test instruments and computers to be connected together. The standard later became IEEE 488, often known as GPIB. And while the dominant physical interface has changed (USB and Ethernet are common today), the concept has largely remained the same with a widely used Application Programming Interface (API) eventually emerging known as VISA, for Virtual Instrument Software Architecture.

At the core of the VISA API is a driver for each instrument. The drivers handle mapping the strongly-typed test API to what is usually a loosely-typed text-based Standard Commands for Programmable Instruments (SCPI) control language that is sent over the physical layer.

And while the solution has worked well for the last 30 years, a concurrent but much larger revolution has taken place with a very similar problem statement: What is the best way for machines to talk to other machines in a generic sense? That is, how does a low-energy Bluetooth module communicate with a wearable computer? How does an app on your phone retrieve stock quotes from the Internet? How do you remotely set the state of a light switch in your home? These are all examples of Machine-to-Machine (M2M) communication.

In a generic sense, testing a product is no different than orchestrating an afternoon workout on the treadmill. The treadmill is reporting incline and speed, your watch is measuring heart rate, pulse oximetry, and temperature. Your phone aggregates that information, potentially adjusts the speed or incline, encrypts it, and uploads it to the cloud. Then it prepares a graph for you to view in real time as you work out. Later, you can go to the cloud via a web page, and do deep analysis on the data you've collected. From there, the sky is the limit on what can be done with the data (**Photo 1**).

What is important to understand are the pieces used to enable this seamless communication between so many disparate devices. Once you understand these elements, you'll see Test and Measurement in an entirely new light.

SysCheck2™

Validation of your calibration with a single click

The first intelligent acoustic sensor system with built-in self-verification

GRAS 246AE
with SysCheck2™
functionality



- Remote verification of the entire measurement chain (microphones, cables, input module and software)
- Easy setup—no physical interaction required
- Rapid validation of all channels with a single click
- Provides monitoring of environmental data
- Use of standard CCP-compatible cabling
- Connected to a CCP power module with TEDS support



SysCheck2™ is a GRAS patented technology for verifying measurement chain integrity. SysCheck2™ allows remote, simultaneous health check on microphones, channel gain and cable integrity.

SysCheck2™ functionality reduces production downtime and is particularly convenient when the microphones are placed where physical verification is time consuming or hazardous—for example, if the microphones are hidden in complex mechanical structures, such as couplers or specialized test boxes on production lines.

Read more at:

www.grasacoustics.com/syscheck2

GRAS Sound & Vibration
An Axiometrix Solutions Brand

What Is REST?

REST stands for Representational State Transfer. REST isn't standard. It's an architectural prescription for how the Internet is glued together. At the core are requirements for security, performance and scalability, uniform interfaces, and the ability to evolve clients and servers separately. When you retrieve a web page, you are following the REST recipe, often known as being RESTful. This means there's a client and there's a server. Data on each server is exposed through a URL, and that data might change several times per second. For example, the URL:

```
http://localhost/temperature
```

will connect to your local machine and query the temperature. And every time you call that URL, the value returned might be different. If you wanted to know the temperature on another machine, you might call:

```
http://myWatch/temperature
```

or

```
http://myThermostat/humidity
```

If down the road the server API needs to change for some reason, you can add a new API by simply extending the URL:

```
http://myWatch/V1/temperature
```

Legacy systems can call the old URL, and new systems can call the new URL.

The URLs I mentioned will rely on an HTTP GET method to achieve the read. But how do you add new data to a remote machine? That is handled by an `HTTP POST` or `HTTP PUT`. Just as when you fill out a form to purchase something on the Internet, the data is encapsulated in a well-formed data interchange format, such as XML or JSON, and then encrypted for safe transport.

In short, the last few decades have delivered a framework for M2M communication that is nothing short of remarkable. It has proven extensible, secure, robust, scalable, and performant.

The Benefits of REST

So, given this framework for M2M communication, how can we apply it to test and measurement? Let's look at the various pieces that come into play.

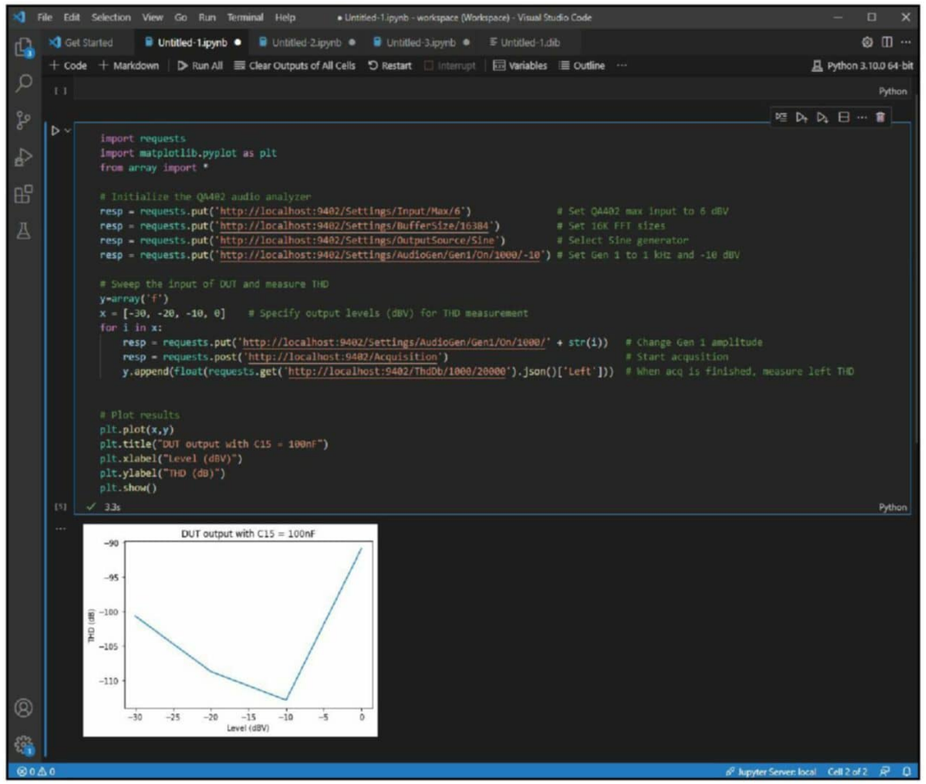


Photo 2: This image shows a Jupyter notebook using Python and direct HTTP calls to sweep and graph the THD measurement of a device under test (DUT). The calls are made using direct HTTP calls for illustration. Jupyter allows code, plots and rich text to live side-by-side, and is an excellent way to document the engineering evolution of a product during development. Jupyter is a common "front-end" for cloud computing and collaboration. At the bottom of the Jupyter code window, you can see it took 3.3 seconds to measure THD at four points using 16K FFTs.

Tools

As the M2M revolution has unfolded, a large set of overlapping tools have emerged to help cope with the challenges associated with M2M communication. At the center of it all is "big-data," data science, and the need to process extraordinary amounts of data pulled from a range of disparate locations. While the amount of data generated during test and measurement are small by big-data standards, we can benefit enormously from the analysis and scalability built into the various toolsets.

Free (and open-source) tools (e.g., Jupyter) permit an interactive environment where code, Markdown, and graphs can all live together (**Photo 2**). During development, this makes it easy to do something, such as plot THDN vs. frequency, change a capacitor, and run the THDN plot again, all while annotating your development journey in a rich-text markdown language.

Debugging tools (e.g., cURL) make it easy to create and debug complex M2M sequences if you are developing factory tests. And legacy tools (e.g., Matlab and LabVIEW) know how to speak REST already.

In short, the tools, ranging from development to deployment, are mature and pervasive, and often free. This becomes particularly important as manufacturing must scale.

Developers

Developers that understand test and measurement (GPIP, the interfaces, and the specific tools) are rare. It's a specialized skill that doesn't have much applicability outside of the factory realm. But M2M communication via REST is widely understood. Whether you are querying inventory from a supplier or handling credit-card processing, it's usually done through REST principals. And most developers understand it

About the Author



Matt Taylor holds a BSEE and is the president of QuantAsylum USA LLC, an audio test and measurement provider. Prior to QuantAsylum, he co-founded the Taiwanese electric scooter company Gogoro where he served as CTO. Before that, he worked at Microsoft, Motorola Cellular, and HTC. A lifelong musician, he fell in love with audio when he first entered a recording studio in the 1980s.



very well. Asking a developer to query data from a remote source via HTTP is a trivial task.

Languages and Environments

Languages such as Java, C#, and Python have also evolved. Making an `HTTP GET` or `PUT` or `POST` from a modern language is straightforward. There are no libraries needed, no new interfaces to learn, and no drivers. The operation, regardless of the underlying operating system is the same. On the QA402 Audio Analyzer, for example, if you want to know the THD of the last acquisition you made, with fundamental at 1kHz and a 20kHz bandwidth, you can type the following in your browser (or call from a program):

```
http://localhost:9402/ThdDb/1000/20000
```

And a JSON document is returned:

```
{
“SessionId”:”3307217056”,
“Left”:”-91.7134518113503”,
“Right”:”-94.3172168407987”
}
```

A JSON document is simply a collection of attribute-value pairs. We can see the left channel distortion was -91.73dB.

To request the distortion reading in Python would appear as follows. This line below is sending the HTTP request, and then asking for the value of the “Left” attribute in the returned JSON document. That value is a string, which is then converted to a float. Again, this is using standard language features. No drivers, no DLLs:

```
ThdDbLeft =
float(requests.get(‘http://
localhost:9402/ThdDb/1000/20000’).
json()[‘L eft’])
```

Of course, you’d want to abstract the calls shown above to give a layer of type-safety to the operation.

Performance and Scalability

An `HTTP GET` as shown for THD takes just a few milliseconds to complete. Even for large data structures the performance is very good. Requesting acquisition spectrum data (64K left channel and 64K right channel = 128K 64-bit doubles) data takes just 40mS or so. That’s largely because performance and scalability have driven the design of the Internet from the beginning, and languages such as Java and C# have followed suit. But it doesn’t stop there, as these languages have pushed further into advanced concepts such as asynchronous programming, which is a type of coding where the developer gets the performance gains associated with asynchronous calls without having to worry about the code overhead of callbacks. Asynchronous programming also makes it easy to parallelize a range of tasks. Modern programming languages have treated parallelization as a very important task, and as such there is strong language support for these important architectural elements.

In short, you can submit a range of measurement requests via HTTP before a single request has completed, and process them as they become available. And this can be done with code that is easy to write using modern languages.

Portability

With some planning, modern tools can allow you to seamlessly move a test environment from, say, Windows to Linux. You can develop in the lab on Windows or Mac and deploy to the manufacturing line onto low- cost Linux machines. Cross-platform applications are becoming increasingly common, and REST helps makes the transition seamless.

Virtualization

Docker is a software platform that relies on OS-level virtualization to deliver a software solution in a container. Docker exists to ease deployment of complex systems and allow environments to be evolved separately from each other. Each container is isolated from other containers, and every container can run different versions of key system libraries. Containers are small, usually needing just a few hundred MBytes of system resources. Running 10 containers or more on a single machine is common.

Docker allows you to set up an environment on one machine and move that exact environment to another machine AND run it alongside another container that has a completely different environment. For example, you might have your Product A factory test running in Java version X, and your Product B running in Java Version Y and your Product C running in C# version Z. Think of a very lightweight Virtual Machine.

You can get the tests running at your desk, and then deploy the container to your factory and run it on the same machine that is testing another product. The two test environments can be very different. And yet, they run alongside each other completely independently.

Debugging

The tools available to debug M2M communications are mature and pervasive. Some of the most potent are built right into your browser. But stand-alone apps (e.g., cURL) make it easy to see and understand every aspect of the communication, with detailed logs and timing. Understanding why a M2M session failed is straightforward.

Backward Compatibility

So, the capabilities might sound great. But how would legacy systems be addressed with a new approach? The answer is they’d be pulled in and used as they are today. For example, if you absolutely had to control a legacy piece of test equipment from a modern REST-based approached to factory testing, you could pull in the DLL that permitted you to control the legacy equipment in your new app. You’d probably risk some of the cross-platform compatibility at that point. But it’s a reasonable trade-off.

Conclusion

Testing is, at its core, machine-to-machine communications. And some serious innovation has occurred in this area over the last two decades, fueled by the same standards and protocols that power the Internet.

Is it time to throw away the legacy systems and start over from scratch? Probably not. That ship is large and will need lots of time to turn. The investments in test systems are enormous and time is always tight when shipping a product.

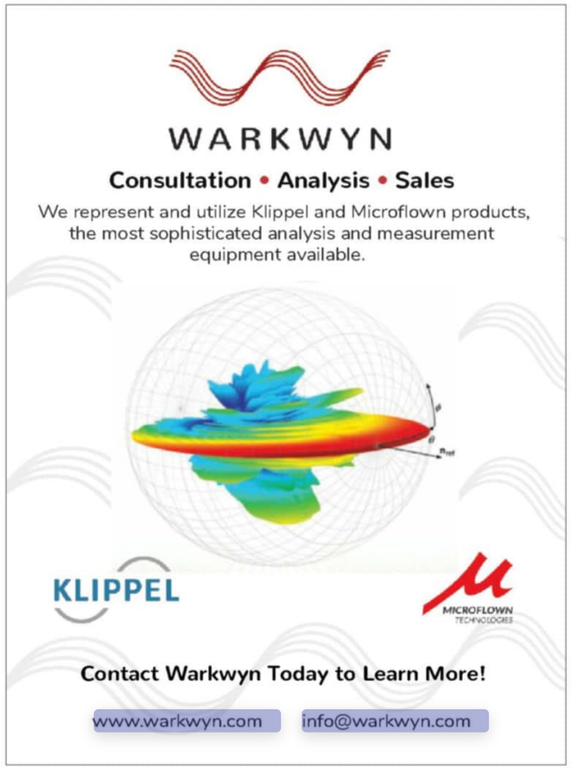
But the gains to be had from state-of-the-art software (languages, analytics, and cloud storage)—much of it open source—are very real. Rather than thinking about test and measurement as a unique problem requiring unique skills and tools, think of it as the same distributed machine-to- machine communication that runs the Internet.

From this new vantage point, your test team may very well venture in a new direction for their next-gen testing solutions. And fast, low-cost, easily deployed containers that perfectly replicate your engineer’s desktop environment are the outcome. And you can deploy several of these containers into your factory onto low-cost test bays running Linux.

References

“Project Jupyter,” *Wikipedia*, https://en.wikipedia.org/wiki/Project_Jupyter

J. Somers, “The Scientific Paper Is Obsolete,” *The Atlantic*, April, 2018, www.theatlantic.com/science/archive/2018/04/the-scientific-paper-is-obsolete/556676





Practical Test & Measurement

Measuring MEMS Microphone Arrays

By
Joe Begin
(Regional Support and Applications Director, Audio Precision)

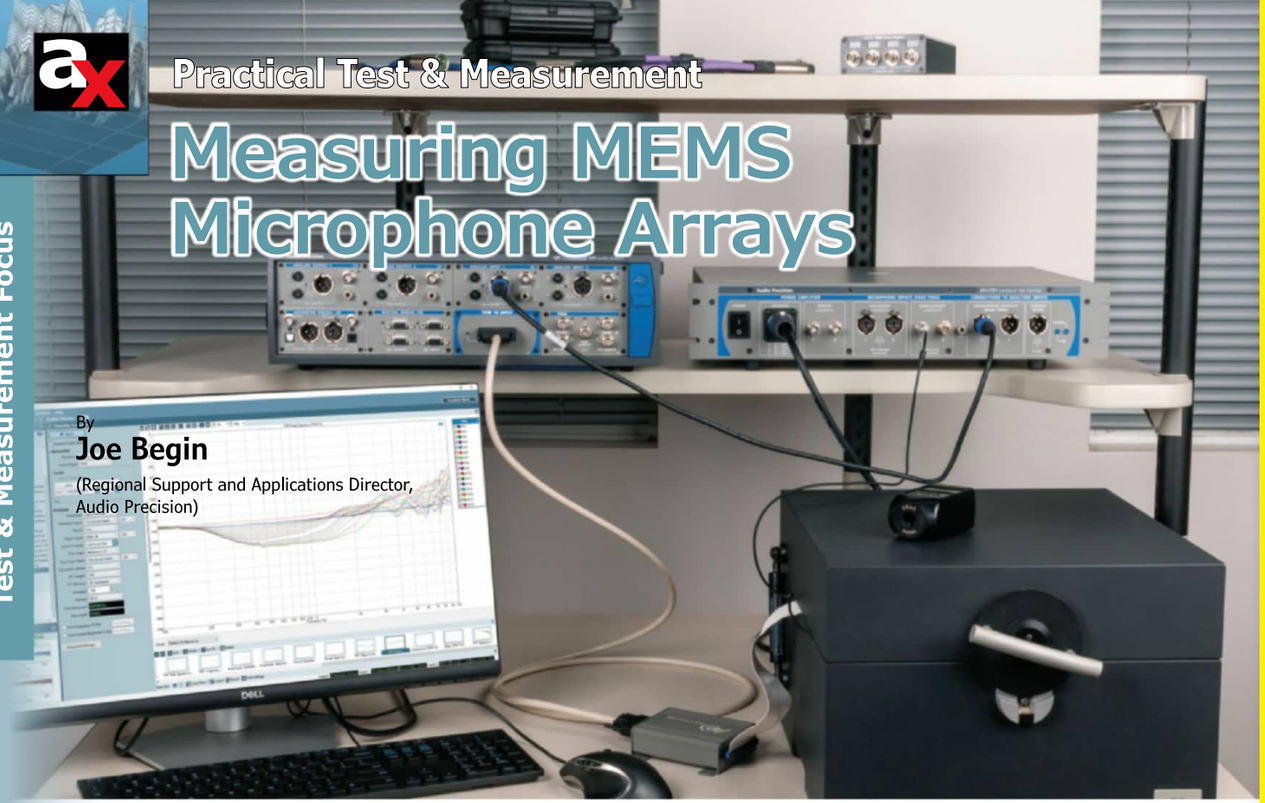


Photo 1: Measurement setup external to the anechoic test chamber

The simultaneous acquisition and measurement of signals from a microphone array presents a variety of challenges to developers seeking to evaluate their designs. This article details what to measure and how to test and evaluate multichannel microphone arrays.

Today's smart devices—smartphones, tablet computers, smart speakers, and voice-activated Internet of Things (IoT)—use MEMS microphones to facilitate hands-free communication, enable the command of a virtual assistant, and support noise canceling algorithms. Depending upon the device, there are typically two to seven (occasionally more) microphones.

While the use of multiple mics in a smart device can provide distinct benefits to the end product—for example, with multiple mics beam forming can be used to improve the signal-to-noise ratio (SNR) of the active speaker and reject background noise—it also complicates the testing process used to evaluate and characterize designs owing to the need for simultaneous, multichannel measurements.

In the development process, it is also critical to be able to accurately analyze inter-channel phase information for MEMS microphone arrays. The test configuration may require the ability to measure two or more digital microphones simultaneously and those digital microphones may require reference to the signal from one or more analog measurement microphones as well. In addition to that challenge, in some test scenarios the test system may be required to provide the system clock; in others it may need to synchronize to an external clock.

Microphone Array Testing

Conducting microphone array tests involves many stages and requires the following equipment: a generator/analyzer for stimulus and acquisition; a source loudspeaker to generate the acoustic test signal; an amplifier for the speaker driver; at least one measurement microphone (for reference); and a power supply for this reference mic. An anechoic test chamber and a sound level calibrator are also needed.

For the sake of this article, I employed the equipment shown in **Photo 1**. The equipment includes an APx525 audio analyzer with PDM 16 module and remote pod; an APx1701 Transducer Test Interface as both the speaker driver amplifier and the microphone power supply; an Interacoustics TBS25 anechoic test chamber with an integrated speaker; a 1/4" free-field

CELEBRATING OVER 40 YEARS!

ACOUSTICS BEGINS WITH ACO™



MEASUREMENT MICROPHONES
Polarized & Electret

Windscreens
3 to18 inch & Custom
Weather/UV Resistant



Community/Industrial
Noise Monitors/Alarms
Indoors and Out





Microphone Systems
Standalone
IEPE Powered
Phantom Power
Custom

ACO Pacific, Inc
Tel: +1-650-595-8588
www.acopacific.com
sales@acopacific.com
Serving End-Users and OEMs Worldwide

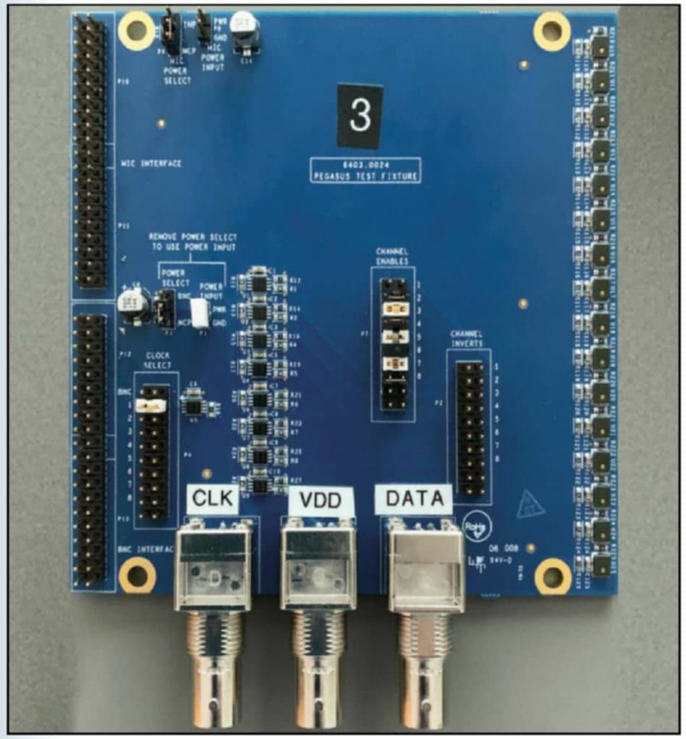


Photo 2: MEMS microphone array board with PDM 16 connectors for the microphones and interface

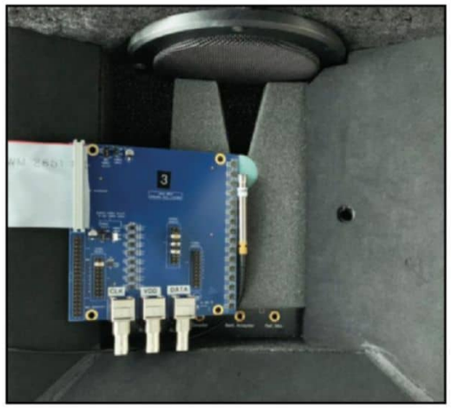


Photo 3: The microphone array test fixture in the acoustic test box

About the Author



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

measurement microphone; and a sound level calibrator.

The device under test (DUT) is a microphone array fixture developed in-house at Audio Precision (AP) as a demonstration board. The small PCB has 16 pulse density modulation (PDM) microphones mounted along one edge of the board with a constant spacing of 6.6mm (0.260") between microphones (**Photo 2**). It also has the same connector as the interface module (in this case, AP's PDM 16 module) and connects to this remote pod via a 40-pin connector and flat ribbon cable.

With microphones mounted in a straight-line array, this fixture was not developed for any practical microphone array acoustic measurements (e.g., beamforming), but it serves as a suitable DUT for demonstrating the simultaneous acquisition and measurement of multiple MEMS microphones.

Test Setup

Ideally, microphones should be tested in an anechoic chamber with a low level of background noise, using fixtures that precisely position the DUT and the reference microphone in the sound field. A single reference microphone can be used centered over the array, or multiple reference microphones can be used, depending on the size and arrangement of the array being tested.

In this example, a small desktop test chamber—the Interacoustics TBS25—designed for testing hearing aids was used. The TBS25 is typically used for acoustic tests in the speech frequency range as it provides approximately 40dB of noise isolation and is anechoic at 500Hz and above. It also has a full-range loudspeaker with a flat frequency response within ±3dB (when the chamber is empty) from 100Hz to 10kHz.

Photo 3 shows the microphone array test fixture in the chamber with the linear microphone array positioned on the axis of the loudspeaker. A 1/4" free-field measurement microphone is located next to the second microphone in the array, at the chamber's "Test Point" directly in front of and on axis to the loudspeaker. This is the center of the light blue circle in front of the speaker shown in the photo.

Test Procedure

To measure microphone performance, the analyzer was configured for acoustic input (to measure sound pressure from the measurement microphone). In turn, the microphone was calibrated to measure pressure in pascals and levels in dBSPL, using the Transducer Electronic Data Sheet (TEDS), and then checked with a sound level calibrator.

The DUT microphone array and the measurement microphone were placed in the test chamber and the analyzer output was set to Acoustic Mode. Regulation was used to calibrate the generator in dBSPL at a reference level and at one reference frequency (e.g., 94dBSPL at 1kHz).

The acoustic frequency response magnitude was measured at the Test Point. (It is advisable to repeat the measurement a few times to ensure consistency.) When satisfied, this relative level result can be exported and used to equalize the generator output, such that the frequency response magnitude is flat at the Test Point. Checks are made to the acoustic output level and, if required, the output regulation step can be repeated. Next, the input connector and settings can be configured before measuring the acoustic response of the microphone array using the same settings as for the frequency response magnitude.

Leveled Sound Field

While we have calibrated the output from the chamber speaker to ensure the acoustic response is flat at the test point, the generator

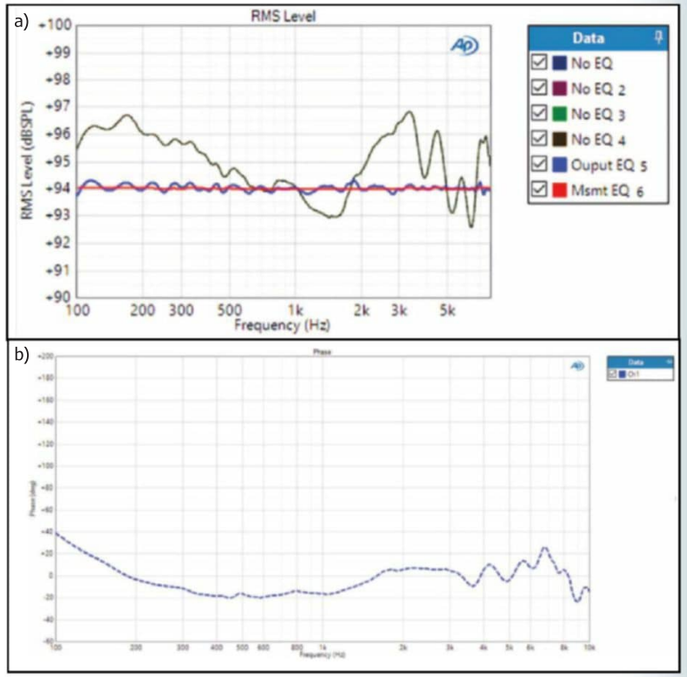
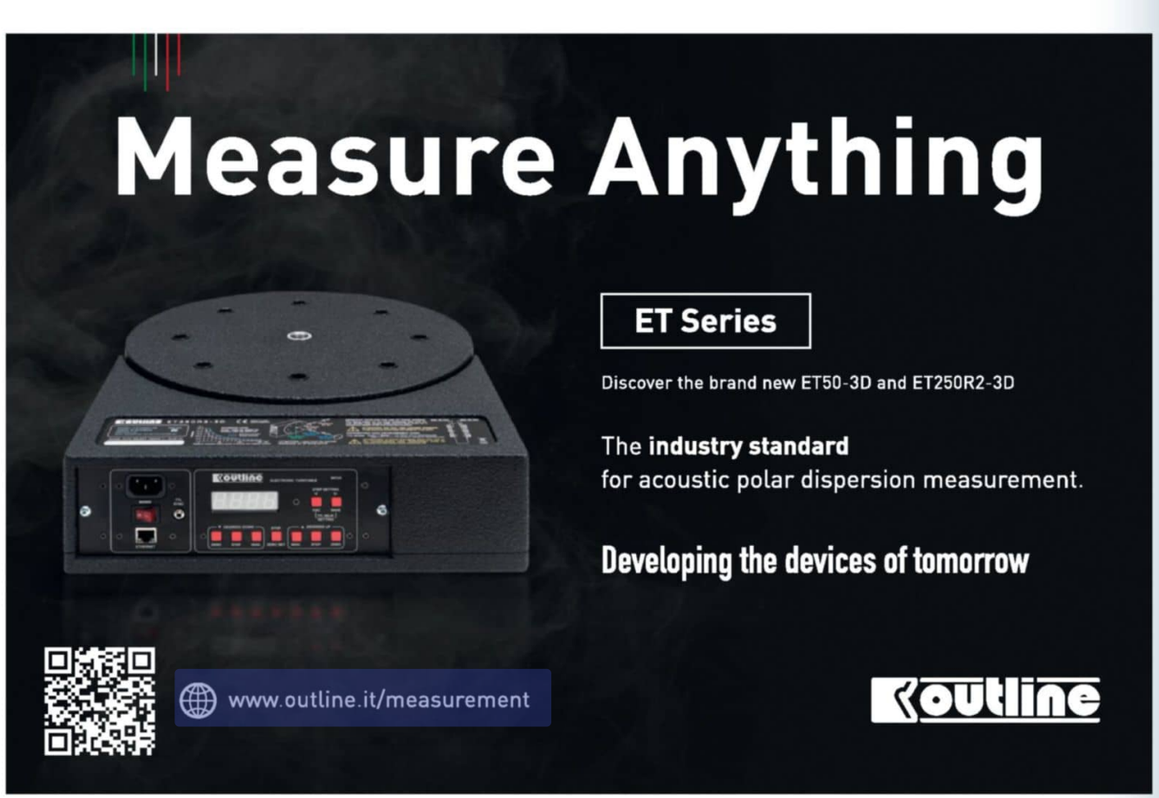


Figure 1: Sound pressure level (SPL) magnitude response at the Test Point (a); and graph showing phase (Input-to-Output Excess) at the Test Point, measured by the reference microphone (b).



output should also be equalized. There are two methods for accomplishing this: output EQ and measurement EQ.

Output EQ is applied in the signal path setup, using 30-pole digital filters to equalize all signals generated in a signal path. Measurement EQ is applied in specific sine-based measurements (including acoustic response). The amplitude of the signal is adjusted at every specified frequency to achieve a very flat response.

Figure 1a shows the frequency response magnitude measured at the Test Point in the chamber with no EQ, Output EQ, and Measurement EQ. The corresponding deviations from flatness were ±3.33dB (no EQ), ±0.28dB (with Output EQ), and ±0.07dB (with Measurement EQ).

It should be noted that this procedure only calibrates the sound field at the Test Point—the response will vary at other locations in the chamber. Only the response of the DUT microphone at the Test Point can be measured using this setup.

The measured phase response at the Test Point with Measurement EQ (**Figure 1b**), represents the phase between the reference microphone and the loudspeaker with propagation delay and any constant group delay removed from the phase calculation.

Microphone Array Response

Figure 2a shows the frequency response magnitude of the microphone array. It is important to note that Ch1, the blue trace (dashed line) represents the true frequency response of a DUT microphone, because only this microphone was at the Test Point, where the sound field was calibrated.

The shape of the Ch1 trace is very close to the typical frequency response of this microphone. The RMS Level at 1kHz is approximately -26dBFS. This is, by definition, the sensitivity of the microphone; the electrical response to a standard acoustic input of 94dBSPL at 1kHz.

Another interesting way to look at the phase data is in terms of inter-channel phase. The phase result in the Acoustic Response measurement can also be set to phase relative to Channel 1 (**Figure 2b**). In the fixture used in this example, Ch1 is the second closest microphone to the speaker and Ch2 is the closest. The average delay of 18.85µs between microphones shows up as a phase difference that increases linearly with frequency. At 10kHz, the phase difference due to one microphone spacing is about 67.9°.

The Acoustic Response Measurement also has a Delay result, which measures the delay of the signal from the speaker to each microphone in the array. **Figure 3** shows the relationship between the measured delay at each mic in the array and its distance from the reference position. Note that the curve is a straight line with a slope of 350m/s—very close to the nominal speed of sound in air at room temperature (343m/s). The difference may

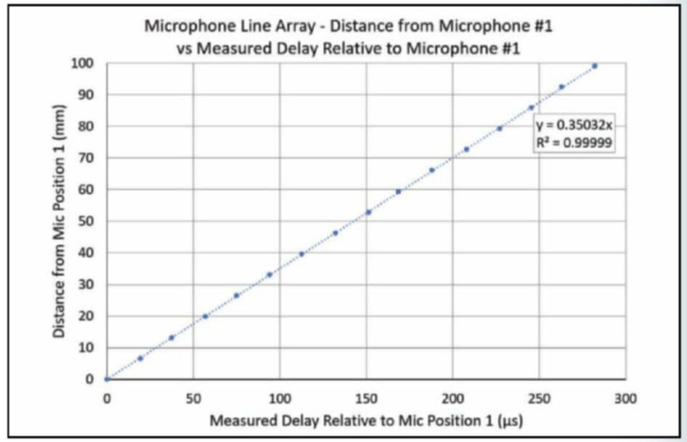


Figure 3: The delay measurements from Figure 2 are plotted vs. microphone position.

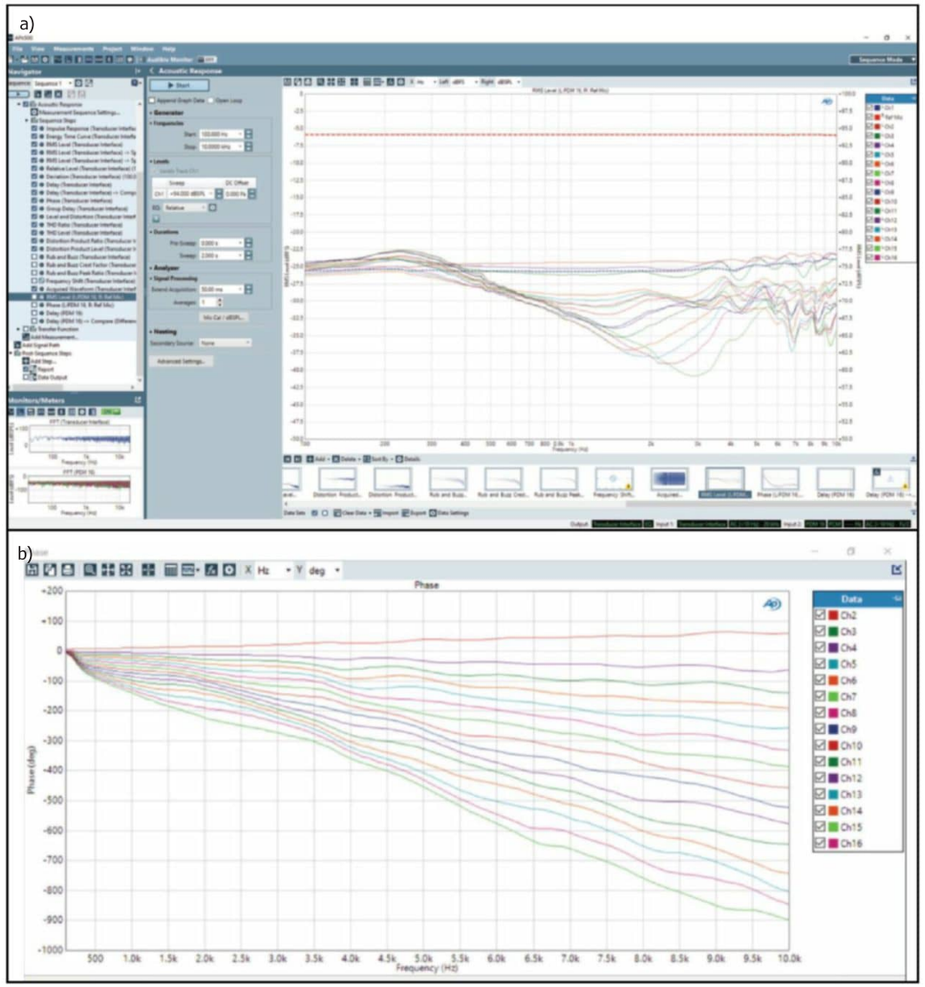


Figure 2: Frequency response magnitude of the 16 PDM microphones in the microphone array (a); phase response of the microphones in the array relative to Ch1 (b).





be due to error resulting from the small distance over which the measurements were made.

Transfer Function Measurements

Using transfer function analysis, the frequency response of microphone arrays can be measured using any broadband signal, including speech or music. Input signals can also be saved to .wav files as they are acquired and analyzed.

This can be useful in applications where the DUT has DSP that blocks sinusoidal signals from being transmitted (e.g., telephony, smart speakers, etc.). For example, users might want to measure transfer functions with speech signals and then run the acquired signals through a DSP algorithm.

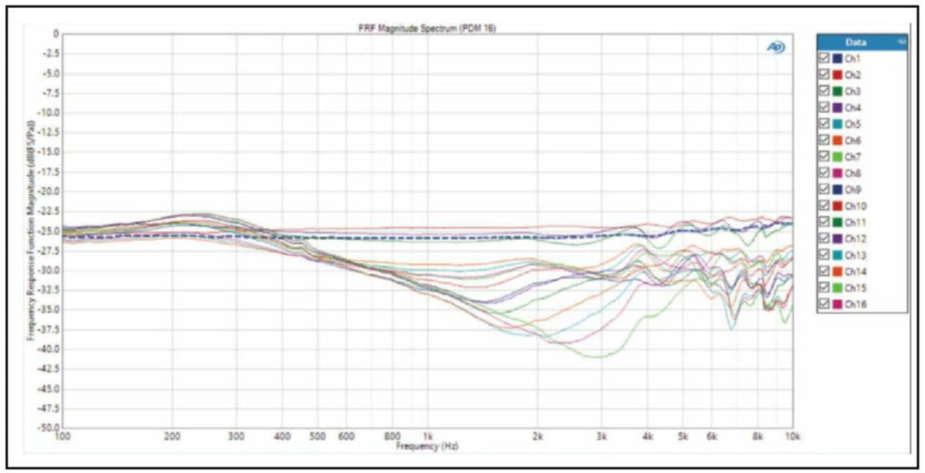


Figure 4: Transfer Function magnitude between DUT microphones and Ch1 microphone at Test Point.

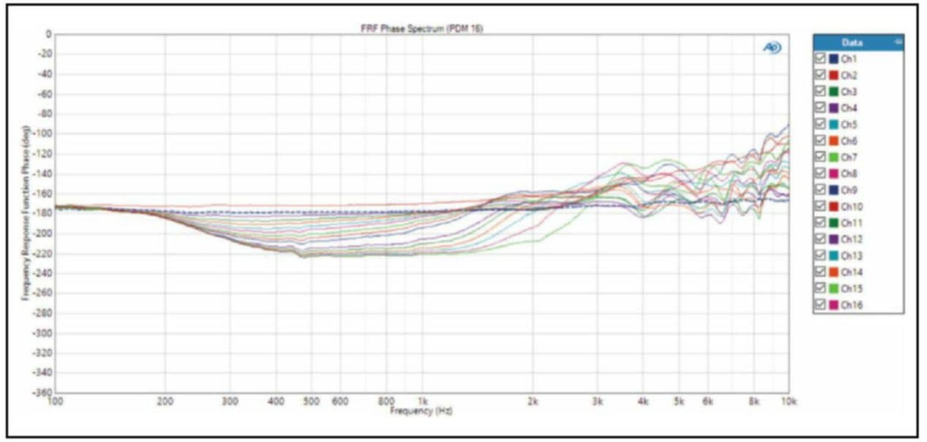


Figure 5: Transfer Function phase between the DUT microphone relative to the microphone Ch1 at the Test Point.

Figure 4 and **Figure 5** show the frequency response magnitude and phase, respectively, of the DUT's microphones relative to microphone Ch1 at the Test Point.

Final Thoughts

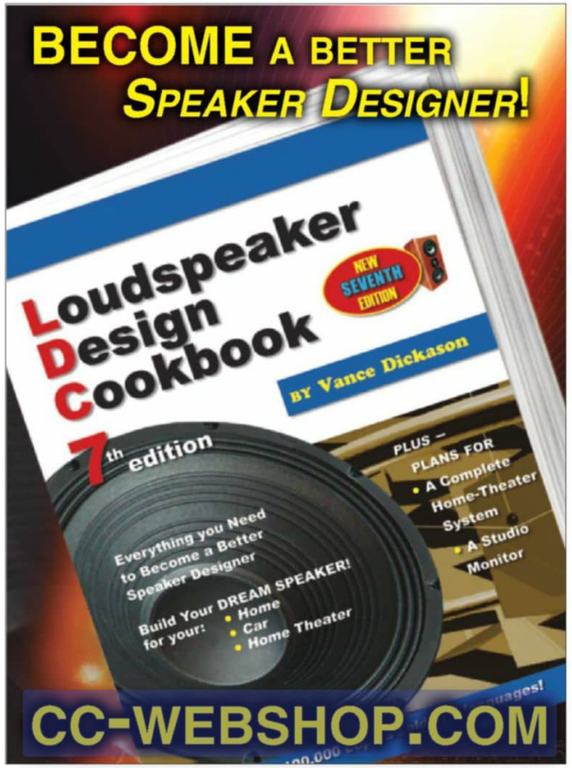
The rapid growth of smart, voice-activated devices—and the associated competition among various device manufacturers—is driving the ongoing need to design and produce improved MEMS microphones, microphone arrays, and DSP algorithms for speech processing and beamforming.

Verifying and characterizing the performance of these components and systems requires audio analysis systems capable of simultaneous multichannel acquisition and measurement.

AMPLITREX AUDIO PRODUCTS, LLC



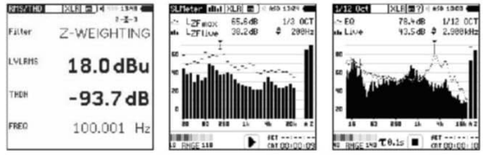
The AMPLITREX AT1000
Simple, accurate,
quality tube testing since 2004
www.amplitrex.com
amplitrex@gmail.com



Ready for any Challenge!

Discover an Audio Analyzer you can trust for your application:

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



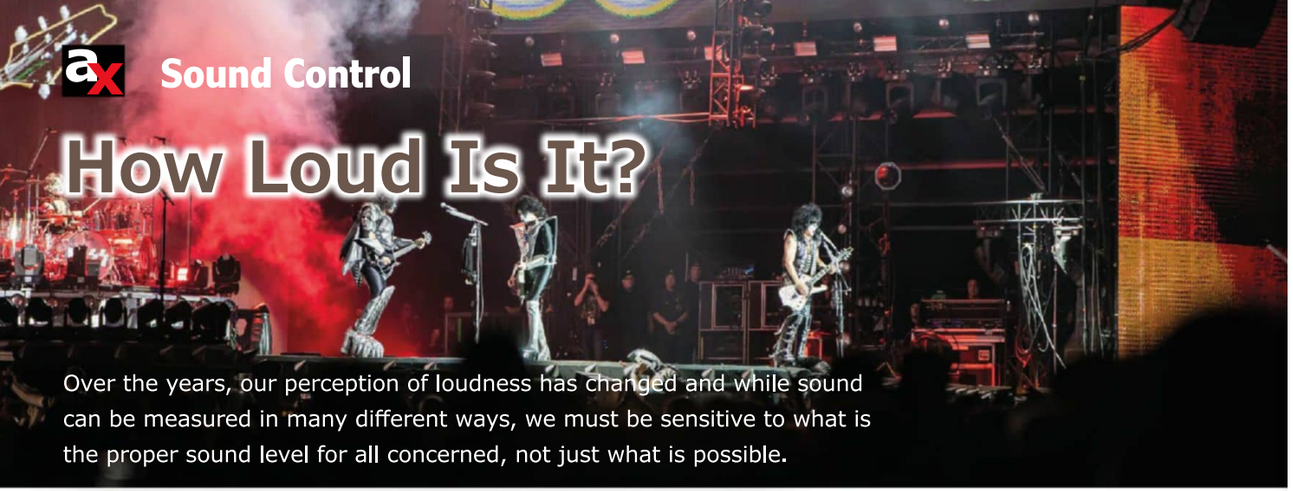
Distortion Analyzer | RealTime Analyzer | 1/12 octave Analyzer



NTi AUDIO | NTi Audio Inc.

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

How Loud Is It?



Over the years, our perception of loudness has changed and while sound can be measured in many different ways, we must be sensitive to what is the proper sound level for all concerned, not just what is possible.

By
Richard Honeycutt

Photo 1: Most people would be correct in expecting that a rock band is probably the loudest music source. Here is a photo of Kiss at Rock Robe Hellfest 2019. (Photo: Jean-Luc Poumarat)

When I was about six years old, my family and I attended local parades several times a year. I was impressed—and, truth be told, a little scared—by the feeling of thumping on my chest as the bass drums passed by. That seemed to me to be the loudest sound I ever heard. During my teen years, amplified rock music became popular, and my elders frequently commented about how loud it was. It didn't seem as loud to me as the bass drums had. I enjoyed listening to music turned up fairly high on my sister's portable record player, and sometimes my parents would complain about the volume. But a mono single-ended tube amplifier (50C5) driving a 5" speaker really couldn't get very loud. As the end of the 1960s approached, rock music concerts—both outdoor and indoor ones—became more and more notorious for their high sound pressure levels (SPLs).



Photo 2: A symphony orchestra playing at *ffff* (very loud), as in Tchaikovsky's "Pathetique," approaches the loudness of a rock band.

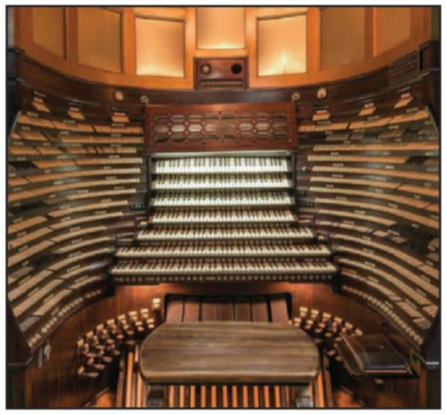


Photo 3: A pipe organ, and specifically the Boardwalk Auditorium organ in Atlantic City, NJ, is the loudest musical instrument in the world, according to the *Guiness Book of World Records*.

Then in the 1970s, contemporary worship music appeared on the scene, and most of it was generally perceived as being louder than traditional organ music and singing. My colleague, consultant/author David Kennedy and chaplain Michael Barton wrote a poem just before Christmas 2021 laying out the quandary of getting the audio levels right:

"'Twas the night before worship, and all through the house, not a creature was stirring, not even a mouse.

Yet when to worship I went, what I did find there, were two little ear plugs, there for my care.

I hustled and bustled, and went to my seat, ear plugs in hand, and into my ears discreet.

The music it rang, oh with such a clatter, all those around wondered, what was the matter.

I hoped and I prayed, that the sound would subside, so that God's spirit, would sit by my side.

I found that regardless, the height of the sound, that my heart found its home, in the slightest of sounds.

The sounds grew much louder, the bass ever so loud, that my feet could hardly, stay on the ground.

But then I remembered, that God often speaks, in a still small voice, which sent me him to seek.

My church says right out, that we like it loud, but for me it is more, than should be allowed.

My search found the sound, that cared for my ears, and helped soothe my heart, of its decibel fears."

Influencing Factors

So what factors influence an audience's perception of the proper loudness? First, or course, is acoustic comfort, which is very subjective, depending upon the listener's age, gender, usage of hearing aids, presence of tinnitus, and past experience (e.g., PTSD). Young children are more likely to be bothered by loud music than teenagers or older people.

Hearing aids cannot perfectly compensate for the loudness of music, so ironically, hearing-challenged listeners are more easily annoyed by high sound levels. Sufferers of tinnitus (ringing in the ears) might find that loud music triggers episodes of ringing. PTSD sufferers might experience unpleasant psychological consequences due to loud music, especially when it is percussive in nature.

One other factor in individual assessments of "proper" loudness is expectations. Most people would probably rate sound sources from loudest to softest in more or less this order: rock bands, marching bands, "contemporary" church services, bagpipe bands, symphony orchestras, jazz "big bands," traditional church services with pipe organs, and large choirs.

In fact, **Photos 1–10** show that these expectations are not necessarily correct. Although the levels of particular music sources are not entirely predictable, it is possible to rank-order various sources in order of loudness, with some accuracy. With peak levels typically ranging from 120dB to 150dB, rock bands are likely the loudest music source. But the second-loudest— a symphony orchestra at 120dB to 137dB—is often regarded as a "tame" sound source.

Another instrument usually considered "tame" holds the Guinness world record for the world's loudest single musical instrument. The Boardwalk Hall Auditorium Organ, which is housed in the Main Auditorium of the Boardwalk Hall in Atlantic City, NJ, has more than 33,000 pipes, seven keyboards, and 1439 keys.

Although I was unable to find the maximum SPL, this organ is said to be six times louder than a steam train's whistle.

The loudest sound source I ever had to support was a real surprise. A local church held a special worship service featuring a 200-voice choir accompanied by electrical instruments and drums. I expected to need a lot of sound to balance the choir with those instruments. What I did not expect was the sound power needed to balance the vocal soloists with the choir. During an a capella selection, the choir reached a level of 120dB. The choir was not miked, although the drums had to be miked to keep up with the choir.



Photo 4: A large a capella choir can be overwhelmingly loud.



Photo 5: A solo trombone is the loudest instrument in an orchestra.



Photo 6: A marching band at normal listening distance produces about the same SPL as a trombone at 1 meter.



Photo 7: A solo trumpet's maximum level is about 5dB less than that of a solo trombone.

Most people would say that a solo trumpet is louder than a solo trombone. However, this is a misconception based upon the trumpet's sound covering higher frequencies, to which the human ear is more sensitive. A solo trombone's maximum SPL at 1 meter from the bell is about 115dB, which is about the same as the maximum SPL of a marching band at normal listening distances. At 115dB maximum SPL, the solo trombone beats out a solo trumpet by about 5dB.



Photo 8: Many people who scorn the loudness of a lounge band (typically 100dB to 105dB) grew up dancing to the music of a jazz "big band" (95dB to 106dB).



Photo 9: Bagpipe bands may not be as loud as most people think.



Photo 10: Despite some opinions, contemporary worship bands are not always loud.

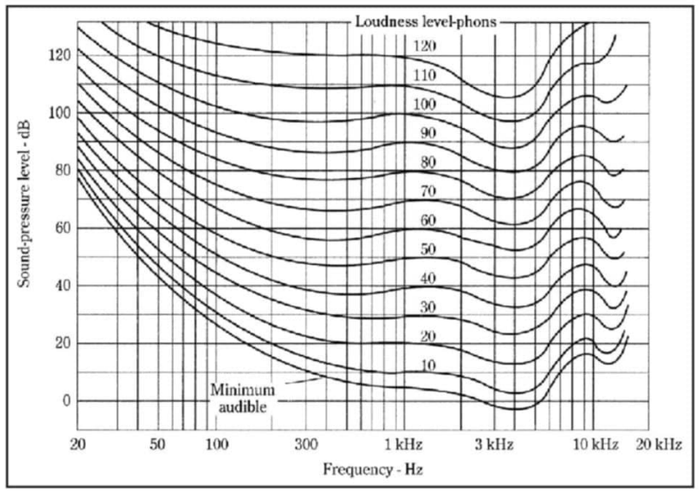


Figure 1: Technically, loudness is measured in phons, and is affected by both SPL and frequency.

Many people consider bagpipe bands to be extremely loud, but this reputation may be undeserved, caused by the fact that much of the bagpipe's sound is concentrated in harmonics that fall within the ear's most sensitive frequency range. The perceptual loudness is likely part of the reason that bagpipe corps have been at the forefront of most Scottish infantry for centuries. During World War I, the Germans called the kilted Scots soldiers "the ladies from Hell." A bagpipe band's maximum SPL is only about 102dB.

As Kennedy and Barton's poem printed earlier in the article makes clear, not all contemporary worship bands are equally loud. In the past few decades, these bands have evolved from mostly loud "Jesus rock" styles to more mellow music, and many sound operators have developed a greater understanding of the dangers and other disadvantages of overly loud sound. The result is that the average maximum SPL of a modern worship band is about 85dB to 100dB—a range that falls somewhere between an action movie and a typical *forte* pipe organ (not the Boardwalk organ!)

Other Factors

Other factors besides SPL and frequency affect perceived loudness. Probably because sound levels above about 90dB cause IM distortion within the ear canal, distorted sound seems louder than undistorted sound of the same SPL. Strongly percussive music can also sound louder than it would measure.

So why is it useful to know about the loudness of various musical sources? First, we must consider possible hearing damage. Sound at levels higher than 80dB (OSHA regulations say 85dB) can produce permanent hearing damage if the exposure is long enough. "Long enough" varies widely from one individual to another, and is affected by certain medications as well. Regulatory bodies quantify the maximum safe sound exposure in terms of a "doubling rate": The Occupational Safety and Health Administration (OSHA) says that an 8-hour exposure is safe if the level does not exceed 85dB, but at 90dB, the maximum exposure time is 4 hours—a doubling rate of 5dB. Other regulators and researchers advocate for a 3dB doubling rate. So for a two-hour concert, the maximum safe SPL would be 90dB (5dB doubling) or 86dB (3dB doubling).

For any person, there is a maximum instantaneous SPL at which immediate hearing damage will occur. This level varies from as low as 100dB up to about 150dB. (I personally consider 120dB to be the maximum safe peak SPL.)

People who suffer from intermittent tinnitus (ringing in the ears) often find that loud sounds trigger episodes of ringing, which may not stop until sometime after the exposure to the loud sound. Hearing aid users often find loud sounds annoying or uncomfortable. Negative psychological effects of loud sound can include elevation of blood pressure and other symptoms of the "fight-or-flight" reaction as well as apprehensiveness, perhaps progressing to agitation. These symptoms are worse among sufferers of PTSD. Sounds in which speech intelligibility is important (e.g., the spoken word and vocal music) can suffer from reduced intelligibility at sound levels above 90dB.

Final Thoughts

Our technology has taken us far from the days when a PA system used to support speeches addressed to a few hundred was a real achievement, to now, when overly loud sound can move beyond annoyance to possible permanent hearing damage! Along with this technical progress has come the change from "Wow, I can hear everything he is saying" to "turn that thing down!" As our friends poets Kennedy and Barton have reminded us, we must be sensitive to what is the proper sound level for all concerned, not just what is possible and perhaps pleasing to a select few.

Author's Note: The poem was printed by FOH *magazine and used by permission of the authors.*

Loudness

Technically, loudness is a perceptual term involving SPL and frequency—the human perception of loudness varies with both frequency and SPL, as you can see in **Figure 1**. However, in common usage, loudness and SPL are often considered synonymous. In this article, I use the term "loudness" in its common usage.

About the Author



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

Resources

"What is the World's Loudest Instrument?," The Liverpool Academy of Music, https://liverpoolacademyofmusic.com.au/worlds-loudest-instrument





Sound Level Evaluation for Raspberry Pi 4



By
Ben Chen
(Sfera Labs)

The article details the possibilities of Sfera Labs' Exo Sense Pi—a compact multi-sensor module with a Raspberry Pi 4 that also offers a built-in microphone—as a sound meter. The author illustrates the main concepts involved and possible software applications for sound level evaluation with the Raspberry Pi.

Photo 1: Exo Sense Pi is a compact and multi-functional device from Sfera Labs.

Exo Sense Pi is a compact and multi-functional device from Sfera Labs, based on the Raspberry Pi Compute Module 4, with several environmental sensors and connectivity options (**Photo 1**). It integrates temperature and humidity sensors, air quality, light, PIR-based motion detection sensors, and a digital microphone.

In addition to audio recording, the microphone can also evaluate ambient noise levels, simulating the behavior of a sound meter. Unfortunately, software and documentation about noise level evaluation are scarce online, so we developed and integrated its software to be used with the Exo Sense Pi (**Photo 2**).

This article will share the knowledge gained by Sfera Labs, illustrating the main concepts involved, hoping to provide guidance and help to those who intend to develop or integrate software applications regarding sound level evaluation with Raspberry Pi.

Sound Pressure Level (SPL)

Humans perceive the sound using the eardrum, a membrane that vibrates after a pressure change due to oscillations of the particles in the air that surround us, providing the sensation of hearing.

When a solid and elastic object (e.g., a tuning fork) is stressed mechanically due to collision, it starts to vibrate. The collision acquires potential energy, which is transferred to the surrounding air particles as kinetic energy, making them vibrate around their static equilibrium position.



Photo 2: Sfera Labs developed and integrated its software to be used with the Exo Sense Pi.

The measure of the over-pressure produced by the acoustic perturbation, in each time instant t, can be called $p(t)$. Because of the enormous range of sound amplitude, the pressure is used in logarithmic form, multiplied by 10, and compared to the reference acoustic pressure p_0=20μPa. The value of 20μPa is the absolute audible threshold for a typical human listener for a pure tone sound with a frequency of 1,000Hz (often considered as the threshold of human hearing, roughly the sound of a mosquito flying 3m away).

Since in sound level evaluation what is essential is the energetic significance of the sound field, the formula to compute the instantaneous sound pressure level (SPL) in decibels (dB) considers the square of the acoustic pressure instead of just the acoustic pressure:

$$L_p(t) = 10 \times \log_{10}\left(\frac{p(t)^2}{p^2 0}\right) = 20 \times \log_{10}\left(\frac{p(t)}{p_0}\right) = [dB], \text{where } p_0 = 2 \times 10^{-6}\, Pa$$

[Equation 1]

Intuitively, some may wrongly think that if decibel describes the SPL, then 0dB means a state of no sound present. Decibels measure a ratio: 0dB does not mean sound absence; it means a sound level equal to that of the reference level p_0:

$$20 \times \log_{10}\left(\frac{p(t)}{p_0}\right) = 20 \times \log_{10}(1) = 0, \text{if } p(t) = p_0$$

[Equation 2]

p_0 is a small pressure, but not zero. It is also possible to have negative valued sound levels (e.g., -20dB would mean a sound with pressure 10 times smaller than the reference pressure p_0).

Equivalent Continuous Sound Level (LEQ)

In electric systems and sound level evaluation processes, the RMS value is preferred to the actual value to indicate the

17463759

ellisys
Better Analysis



Audio Over Bluetooth® LE Is Here. Ready?

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



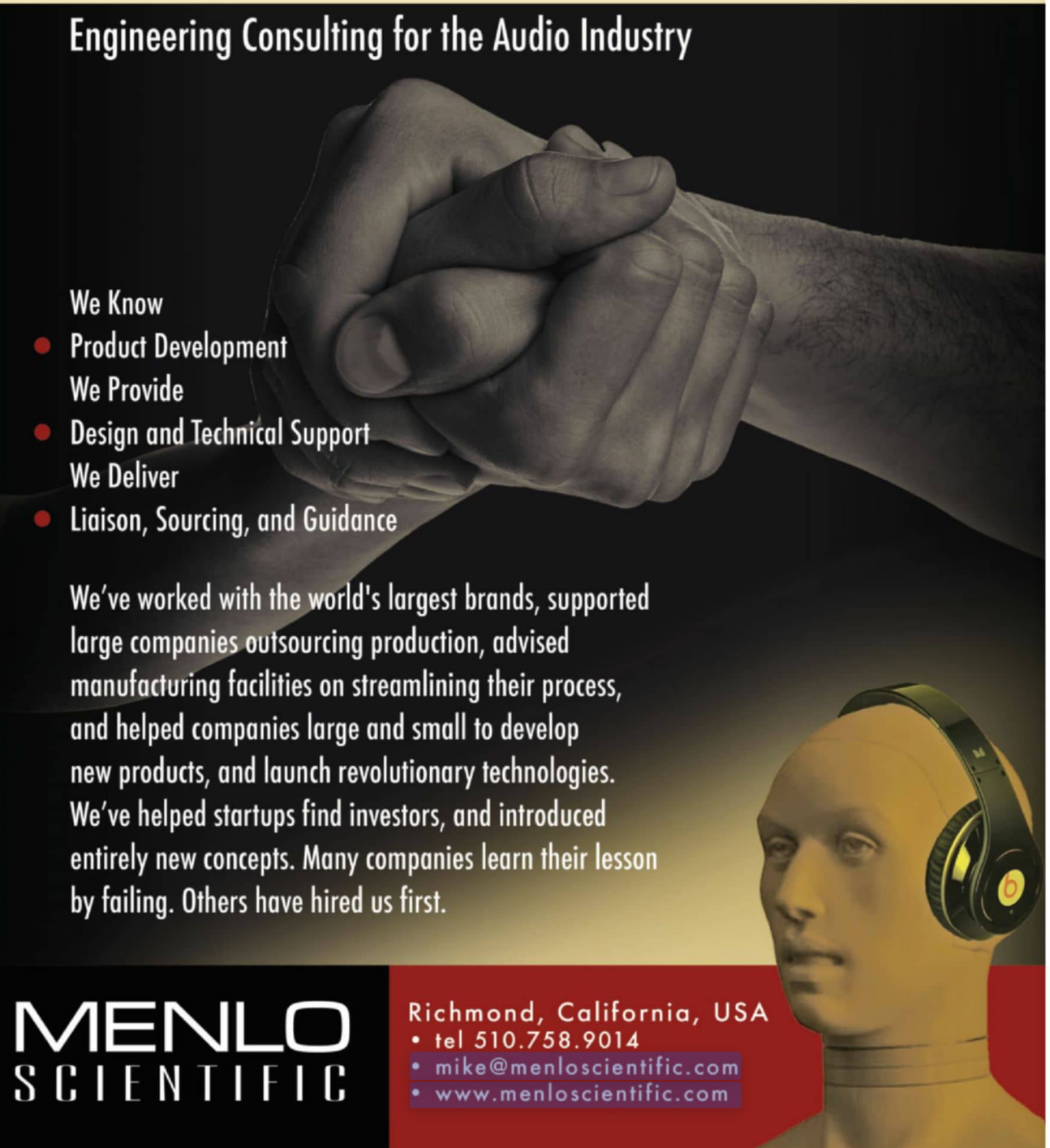
Bluetooth Tracker™
Ultra-Portable Bluetooth® Low Energy Protocol Analyzer
BLE + Wi-Fi 1x
www.ellisys.com/btr1

The Bluetooth® wordmark and logos are registered trademarks owned by the Bluetooth SIG, Inc. and any use of such marks by Ellisys is under license. Wi-Fi® and the Wi-Fi Alliance logo are trademarks of Wi-Fi Alliance. Other trademarks and trade names are those of their respective owners.

MENLO SCIENTIFIC

Engineering Success since 1983

Engineering Consulting for the Audio Industry



We Know
- Product Development

We Provide
- Design and Technical Support

We Deliver
- Liaison, Sourcing, and Guidance

We've worked with the world's largest brands, supported large companies outsourcing production, advised manufacturing facilities on streamlining their process, and helped companies large and small to develop new products, and launch revolutionary technologies. We've helped startups find investors, and introduced entirely new concepts. Many companies learn their lesson by failing. Others have hired us first.

MENLO SCIENTIFIC

Richmond, California, USA
- tel 510.758.9014
- mike@menloscientific.com
- www.menloscientific.com

Frequency	8Hz	16Hz	31.5Hz	63Hz	125Hz	250Hz	500Hz	1kHz	2kHz	4kHz	8kHz	16kHz
A-weighting	-77.8dB	-56.7dB	-39.4dB	-26.2dB	-16.1dB	-8.6dB	-3.2dB	0dB	1.2dB	1dB	-1.1dB	-6.6dB
C-weighting	-17.7dB	-8.5dB	-3.0dB	-0.8dB	-0.2dB	0dB	0dB	0dB	-0.2dB	-0.8dB	-3dB	-8.5dB
Z-weighting	0dB	0dB	0dB	0dB	0dB	0dB	0dB	0dB	0dB	0dB	0dB	0dB

Table 1: This is an example of a one-octave frequency weighting table.

quantity of noise/energy over a certain period. For example, in electrical engineering, the RMS value of an alternating current passing through a resistance is the value of a direct current that causes the same increase in the temperature of the resistance in the same period. For sound evaluation, the RMS (also called Equivalent Continuous Sound Level or LEQ) is the SPL value of a constant sound over the same period, which produces the same amount of energy/noise, defined by the equation:

$$L_{eq} = 10 \times \log_{10}\left(\frac{1}{T}\int_0^T \frac{p(t)^2}{p_0^2}dt\right) = 10 \times \log_{10}\left(\frac{1}{T}\int_0^T 10^{\frac{L_p(t)}{10}}dt\right)$$

[Equation 3]

The above two representations of LEQ are the same because:

$$10^{\frac{L_p(t)}{10}} = 10^{\frac{20\times\log_{10}\left(\frac{p(t)}{p_0}\right)}{10}} = 10^{2\times\log_{10}\left(\frac{p(t)}{p_0^2}\right)} = 10^{\log_{10}\left(\frac{p(t)^2}{p_0^2}\right)} = \frac{p(t)^2}{p_0^2}$$

The use of the LEQ is also very convenient for math operations because it is possible to add the LEQs of different periods and get a resultant LEQ value for the total of all periods, according to this equation:

$$L_{eq,T} = 10 \times \log_{10}\left[\frac{1}{T_1+T_2+\ldots+T_n}\left(T_1\times 10^{\frac{L_{eq,T_1}}{10}} + T_2\times 10^{\frac{L_{eq,T_2}}{10}} + \ldots + T_n\times 10^{\frac{L_{eq,T_n}}{10}}\right)\right]$$

[Equation 4]

Time Weightings as Classifiers

After finding the way to measure and quantify the SPL, scientists of the past asked themselves if it is also possible to recognize particular types of sounds. They made tests whose empiric results can provide a general indication about which time constants to use to recognize different classes of sounds:

- FAST time constant = 125ms. Used to replicate the natural response of the human ear.
- SLOW time constant = 1000ms. For environmental noise studies, especially for studies that span many hours or even days, because it is good at "ignoring" short, fast sounds (e.g., a door slamming or balloon popping).
- IMPULSE time constant = 35ms. Used in situations where sharp, impulsive noises are measured (e.g., fireworks or gunshots).

Frequency Weightings

Human eardrums are different from those in other species, and they have their distinct properties and behaviors. Humans can hear sounds up to the frequency of 20kHz, while dogs can hear up to the frequency of 45kHz.

We made speakers and microphones by imitation based on the same principle of the vibrating membrane as the eardrum. Unfortunately, they cannot perfectly replicate the different responses of the human ear to different sound frequencies because the human ear is less sensitive to very low or very high sound frequencies. The solution is the use of frequency weighting tables.

Sound frequency adjustment tables are a collection of sound level adjustments values for frequency bandwidth intervals, usually of 1 octave or fractions of octaves, applied during the sound evaluation. Three types of the most used frequency weightings are:

- A-weighting (unit of measure dBA): Used to replicate the behavior of the human ear
- C-weighting (unit of measure dBC): The C-weighted sound level does not discriminate against low frequencies and measures almost uniformly over the frequency range between 31.5Hz and 8kHz. This scale is helpful for monitoring sources such as engines, explosions, and machinery.
- Z-weighting (unit of measure dBZ): No frequency weighting is applied.

Table 1 shows an example of a one-octave frequency weighting table. For example, in A-weighting, the LEQ representing all the sound components with frequencies between 0Hz and 8Hz must be adjusted, adding -77.8dB. In comparison, -56.7dB is added to all the components with frequencies between 8Hz and 16Hz.

The Fourier Transform applies only to periodic signals. A random chunk of the audio signal is not periodic, so the results of the Fourier Transforms are affected by spectral leakage. The solution to the problem is applying a window function to decrease the amount of spectral leakage. See Resources for more information.

Microphone

The basic hardware to implement a sound evaluation system is a microphone connected to a central computation unit. In the specific case of Exo Sense Pi, the computation unit is the Raspberry Pi, and the microphone is the ICS-43432 digital microphone (**Figure 1**).

The ICS-43432 microphone offers multiple advantages including: Suitable sampling frequencies up to 52.8kHz (humans can hear up to 20kHz, so according to Nyquist sampling theorem, a minimum sampling frequency of 40 kHz is required to emulate human perception of noise); digital discrete output; easy audio samples configuration and retrieval using a I²S communication interface with Power Code Modulation (PCM) data representation and Linux ALSA audio drivers; low sensitivity error (±1dB); no need for calibration; small and compact dimensions; and easy integration with other devices.

I²S is an electrical interface used for audio data transmission. It is composed of one or more lines for data transmission; a line used to mark individual bits called "bit clock"; and a line used to delineate frames tolled "word clock" (**Figure 2**).

PCM is a digital representation of the audio signal. The combination of I²S and PCM provides the capability of a microphone to capture audio at a different sampling rate (Hertz), with different data formats (24-bit signed little-endian, 32-bit signed little-endian, ...), in mono or stereo channels. Luckily, there are existing open software libraries to handle their configuration and use (e.g., ALSA audio drivers in Linux OS), so there is no need to reimplement existing software.

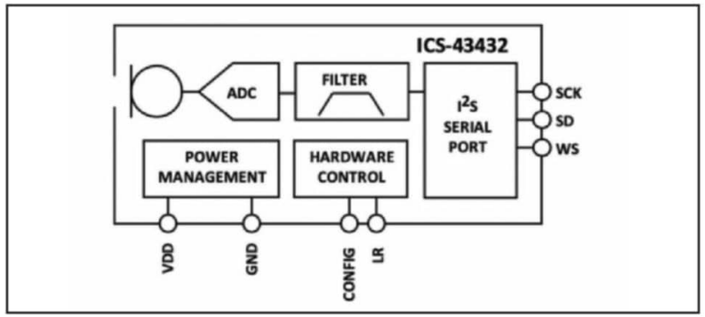


Figure 1: The basic hardware to implement a sound evaluation system is a microphone connected to a central computation unit.

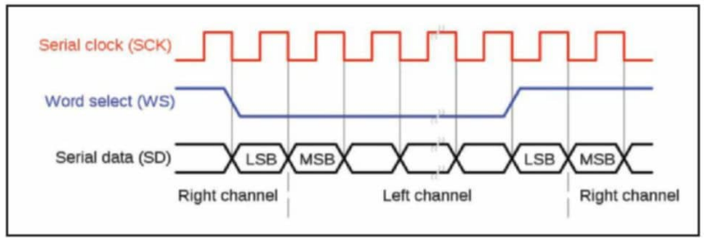


Figure 2: I²S is an electrical interface used for audio data transmission. (Image from "I2S." Wikipedia, Wikimedia Foundation, 17 November 2020, https://en.wikipedia.org/wiki/I²S)

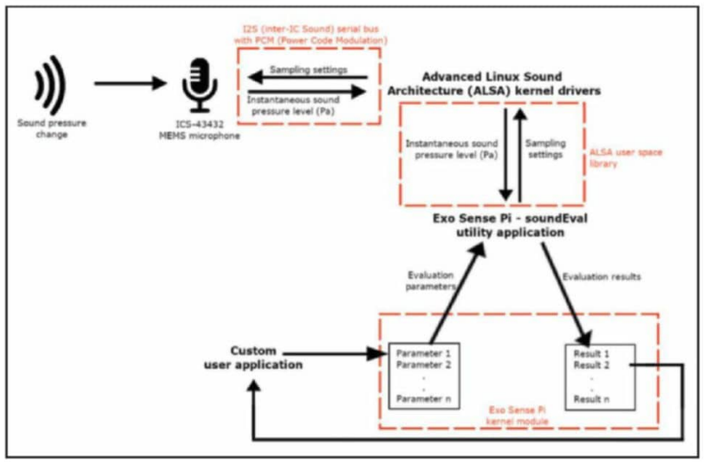


Figure 3: This is the basic software structure implemented inside Exo Sense Pi, using ALSA audio drivers.

About the Author



Ben Chen holds a bachelor and a master degree in "IT engineering" at "Politecnico di Milano." He has extensive working experience as a software engineer, mainly in developing automation applications for plastic production/packaging/wire drawing machinery. Ben is also a music lover and a maker.

Instantaneous SPL from Microphone

A crucial task is the reading of the actual sound pressure from the microphone. In order to perform this operation correctly, we used the following information from the datasheet:

Data format: ICS-43432 can be set to use 24 signed little endian digital format. In this case, the first bit is the sign, then the maximum digital value in output is 2^23 = 0x7FFFFF = 8388607

Sensitivity: It expresses a microphone's efficiency as a transducer (how well it converts acoustical energy to electrical energy). A microphone's sensitivity rating is due to its output voltage related to its measuring SPL. The standard reference input signal for microphone sensitivity measurements is a 1kHz sine wave at 94dB SPL (equivalent to 1 Pascal). ICS-43432 microphone has a sensitivity value of -26dB, compared to a 1kHz sine wave at 94dB or 1Pa SPL.

For digital microphones, sensitivity is a percentage of the full-scale output generated by a 94dB SPL. For a digital microphone, the conversion equation is:

$$Sensitivity_{dbFS} = 20 \times \log_{10}\left(\frac{Sensitivity_{\%FS}}{Output_{REF}}\right)$$

where, OutputREF is the full-scale digital output level. As written above, in the specific case of the ICS-43432 microphone, the sensitivity is -26dB, and the full-scale digital output level is 8388607, so our equation becomes:

$$-26dB = 20 \times \log_{10}\left(\frac{Sensitivity_{\%FS_{1Pa}}}{8388607}\right)$$

from which we can compute the exact digital value to represent 1Pa SPL:

$$Sensitivity_{\%FS_{1Pa}} = Output_{REF} \times 10^{\frac{Sensitivity_{dbFS}}{20}} =$$

$$8388607 \times 10^{\frac{-26}{20}} = 420426$$

Now it is possible to interpret any reading from the microphone and compute the instantaneous SPL in Pascal (Pa), using the following equation:

$$SPL[Pa] = \frac{value_{frommicrophone}}{420426}[Pa]$$

Software Architecture

The noise level evaluation process is complex because it involves a large set of software. On Exo Sense Pi and Raspberry Pi, the Linux operating system offers tools (e.g., the ALSA libraries) for sound card management or libraries to perform Fast Fourier Transforms (FFTs).

There are different ways to implement the same sound evaluation procedure using different software. **Figure 3** illustrates the basic software structure implemented inside Exo Sense Pi, using ALSA audio drivers.

The main idea behind this architecture is the creation of a core Linux software application, called soundEval, which main functions include: set up microphone configuration (sampling rate, audio data format, channels); retrieve instantaneous SPL from the microphone; and perform sound evaluation accordingly to evaluation parameters, such as time

```
pi@exosensepi:~ $ ./soundEval -h
 * Sound Level Evaluation utility for ExoSense Pi
 *
 *     Copyright (C) 2020-2021 Sfera Labs S.r.l.
 *
 *     For information, see the ExoSense Pi web site:
 *     https://www.sferalabs.cc/exo/

Commands:
   -t, --time TIME_WEIGHT          Set the time weighting mode:
                                   TIME_WEIGHT: fast|f|slow|s|impulse|i
                                      fast|f = fast weighting, t = 125 msec
                                               Used to replicate the human ear response
                                      slow|s = slow weighting, t = 1 sec
                                               The slow setting is good at *ignoring* short,
                                               fast sounds like car doors slamming or balloons popping.
                                               This makes slow weighting a good choice for environmental
                                               noise studies, especially for studies that span many hours
                                               or even days.
                                      impulse|i = impulse weighting, t = 35 msec
                                                  Usually used in situations where there are sharp impulsive
                                                  noises to be measured, such as fireworks or gunshots.
                                   If no correct time weighting mode is set, fast is used
   -f, --frequency FREQ_WEIGHT     Set which results of frequency weighting mode to print:
                                   FREQ_WEIGHT: a|z|c|all
                                      a   = A-weighting (Frequency weighting to adapt sound to human ear
                                            perception)
                                      z   = Z-weighting (No frequency weighting applied)
                                      c   = C-weighting (A nearly flat frequency response with the extreme
                                            high and low frequencies attenuated)
                                      all = print results of all 3 frequency weightings
                                   If no frequency weighting mode is set, only fast weighting mode is
                                   printed
   -i, --interval SECONDS          Duration in seconds of the evaluation interval. If not correctly set,
                                   the default value of 0 second is used and the interval evaluation is disabled.
                                   Without -c command. the prooram exits at the end of the soecified interval.
   -I, --interval-only             Interval only mode. If --interval is set, outputs regarding single evaluations
                                   periods (fast/slow/impulse) are hidden. Must be used with --interval command

pi@exosensepi:~ $ ./soundEval --time fast --frequency a --interval 60 -I
[*]     [2021-08-26 11:10:40.335] [Duration in ms = 60002.386501] [dB(A) = 66.53689]
pi@exosensepi:~ $
```

Figure 4: Here is an example of the usage of soundEval to evaluate the LEQ of 60 seconds, with the description of the parameters used.



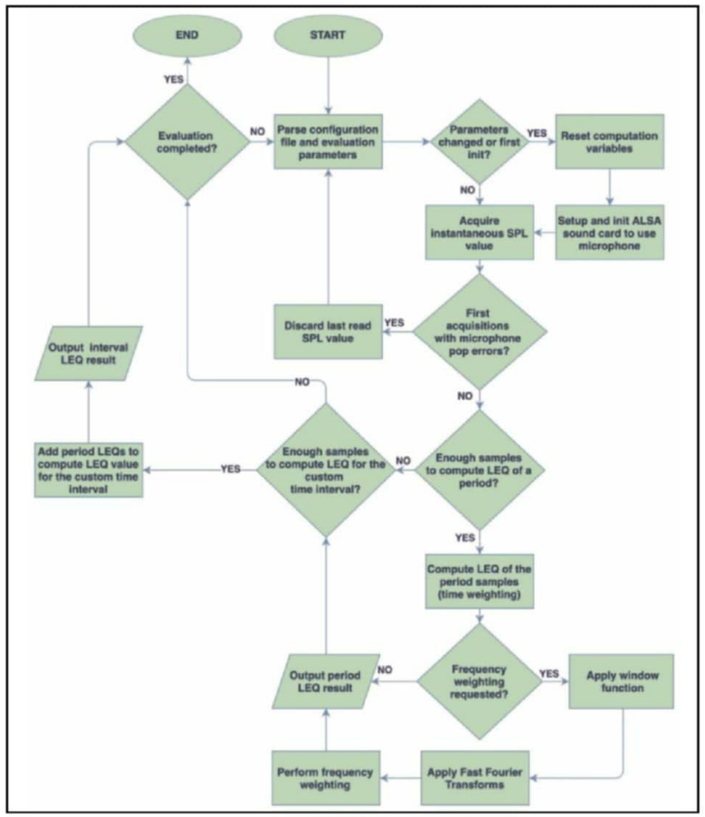


Figure 5: The simplified flow chart shows the sound level evaluation steps implemented in Exo Sense Pi.

	Fast time weighting	Slow time weighting	Impulse time weighting
A-weighting	$L_{AEQ,F}$	$L_{AEQ,S}$	$L_{AEQ,I}$
Z-weighting	$L_{EQ,F}$	$L_{EQ,S}$	$L_{EQ,I}$
C-weighting	$L_{CEQ,F}$	$L_{CEQ,S}$	$L_{CEQ,I}$

Table 2: This table shows the combination of different time constants and frequency weightings, which provides results of different classes of sound meters.

Resources

"Octave band," Wikipedia, Wikimedia Foundation, May 11, 2020, https:// en.wikipedia.org/wiki/Octave_band

"Pulse-code modulation," Wikipedia, Wikimedia Foundation, July 5, 2021, https://en.wikipedia.org/wiki/Pulse-code_modulation

"Spectral leakage," Wikipedia, Wikimedia Foundation, November 17, 2020, https:// en.wikipedia.org/wiki/Spectral_leakage

"Window function," Wikipedia, Wikimedia Foundation, August 20, 2021, https:// en.wikipedia.org/wiki/Window_function

weighting constant, frequency weighting type return evaluation results.

The soundEval application is designed to be run on any Linux OS (**Figure 4**). The Exo Sense Pi kernel module works merely as a device-specific interface between the end-user and the soundEval application, to facilitate the use of evaluation procedures by the end user, like the automatic configuration of the microphone in the Exo Sense Pi.

Exo Sense Pi and Sound Evaluation

The simplified flow chart with sound level evaluation steps implemented in Exo Sense Pi is shown in **Figure 5**.

Table 2 shows the combination of different time constants and frequency weightings, which provides results of different classes of sound meters.

Depending on the requirements of an application, it is possible to use the sound level evaluation as a classifier and a LEQ calculator.

One example of sound evaluation as a classifier is the explosion detection in a chemical materials storage facility. An LEQ evaluation with IMPULSE time constant and frequency C- weighting can detect explosions if the LEQ of a period of 35ms exceeds a threshold dBC value.

One example of sound evaluation as an LEQ calculator is evaluating the equivalent sound level generated by a music concert using an A-weighting frequency adjustment. After computing the LEQ for the entire concert duration, it is possible to compare this result with threshold values and check if the concert volume level was compliant with local regulations.

Since frequency weighting is involved in sound evaluation processes after applying the FFTs, it is possible to return the LEQs of a period, split up by frequency bandwidths, typically of 1 octave or a fraction of octave. With this capability, it is possible to perform operations, such as the creation of a graphic audio spectrum real-time display; a search for bandwidths in which the noise is mainly concentrated, for noise-canceling procedures; and a more detailed sound classification, which also considers the LEQs of each frequency bandwidth.

Conclusions

The soundEval application is an easy to use utility that implements the recognized sound level metering standards, based on an audio feed from a digital microphone. Besides its use in Exo Sense Pi, it could also be used with other hardware platforms, and adapted to process pre-recorded audio or network streams instead of a directly connected digital microphone

TRIAD
ELECTRONIC
TRANSFORMERS
• Original Equipment
• Replacement
• Amateur
TRIAD MAGNETICS, INCORPORATED
460 HARLEY KNOX BLVD., PERRIS, CA 92571
www.TriadMagnetics.com

All In One...

aptX HD Bluetooth Versatility
Digital Signal Processing
Make Your Own Sound



- ✔ Bluetooth 5.0 with AptX HD
- ✔ External Potentiometers
- ✔ DSP Processing
- ✔ Fully Customizable with SigmaStudio

Dayton Audio KABD-250 2 x 50W + DSP Amplifier Board aptX HD Bluetooth 5.0

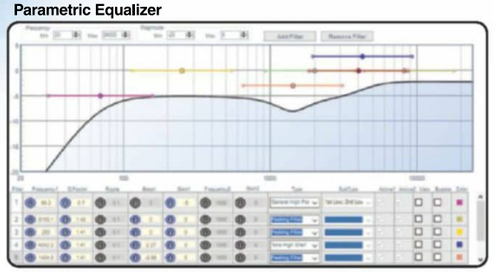
The all-in-one combination of Bluetooth 5.0 aptX HD streaming, advanced signal processing, and 2 channels of efficient class D amplification all in a compact form factor, makes the KABD-250 the perfect choice for creating an incredible sound system that is truly your own.

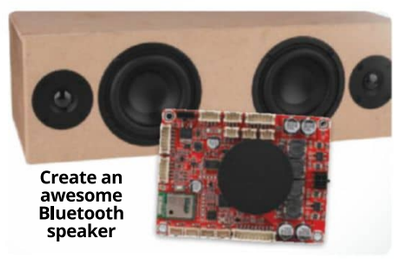
Combined with the ICP1 programming board, you gain access to the SigmaStudio DSP toolbox to create your own unique combinations of equalization, filters, delays, bass processing algorithms that the professionals use on their portable speakers, and so much more! Ports on the board even allow for up to 4 external knobs that can be easily programmed for on-the-fly control of volume, bass, treble, or any other filters you can think of. Use the stereo line output on the KABD-250 to connect another amplifier for a 2.1 or 4.0 system with full DSP customization.



Dayton Audio DSPB-ICP1 In-Circuit Programmer for KABD-250 Boards

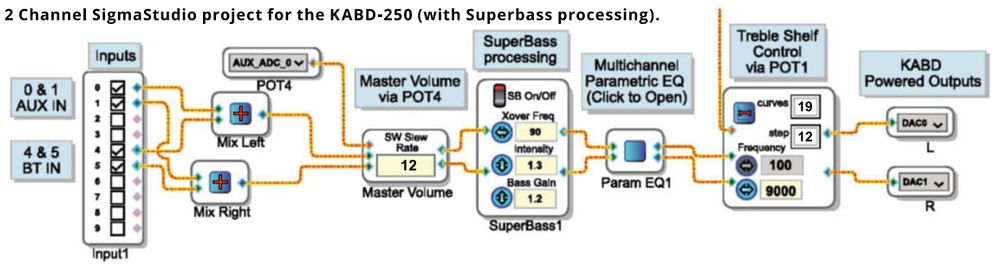
Parametric Equalizer





Create an awesome Bluetooth speaker

2 Channel SigmaStudio project for the KABD-250 (with Superbass processing).



Distributed by:

DAYTONAUDIO
daytonaudio.com

PARTS EXPRESS
AUDIO COMPONENTS & SOLUTIONS
parts-express.com 1-800-338-0531
Custom OEM inquiries welcome

Scan here to learn more about the KABD-250 Amp Board



A Critical Front-End for Innovative Applications

By
J. Martins

This article is an update on dedicated voice capturing and voice front-end solutions. From sub-miniature and MEMS microphones to complete voice-front-ends for voice recognition and voice interface applications, this is an intense area of activity for developers.

Voice applications are expanding, partly as a consequence of voice assistants becoming mainstream as part of the smart home, now broadening its reach to all sorts of mobile devices, connected automotive systems, and new, very specific out-of-home systems. Contactless voice interfaces serving the general public are being tested globally, with governments, banking, and transportation seriously considering all the possibilities given the recommended precautionary habits introduced by the pandemic. In most cases, public information and service terminals that support voice recognition and interaction also provide options for direct operator support, which means using the same voice front-end also has a communication terminal.

Simultaneously, the trend toward implementing voice recognition and voice commands on-device is quickly expanding—for both privacy reasons, but increasingly because manufacturers want to avoid the costs from connecting to voice assistants on the cloud, which can build overtime. As consumer electronics and appliance manufacturers see the possibilities for new exciting applications with machine learning and artificial intelligence on edge devices (edge-AI)—enabling improved features in combination with sensor-fusion, sophisticated algorithms and specialized trained models, they are also encouraged to offer voice control and voice interfaces as standard on those new designs. The new powerful core processing platforms designed for AI are also increasingly at the center of human-machine interface (HMI) systems and hubs for a multitude of networked systems and sensors, which include microphones and audio processing for different applications, from noise reduction and loudness compensation, to fully featured multiuser voice interfaces.

As voice transcription and natural language understanding engines are also evolving with edge-AI, and system designs become more ambitious, microphones are a critical component, frequently requiring multiple devices feeding multiple channels, and combined in groups or arrays. As an example, while working from relatively short distances, current premium true wireless stereo (TWS) earbuds use multiple microphones for voice communication and noise filtering, while also feeding sophisticated active noise cancellation and voice recognition engines—apart from always-on wake-word systems that can be triggered at any moment. All this, while meeting very strict ultra-low-power design requirements.

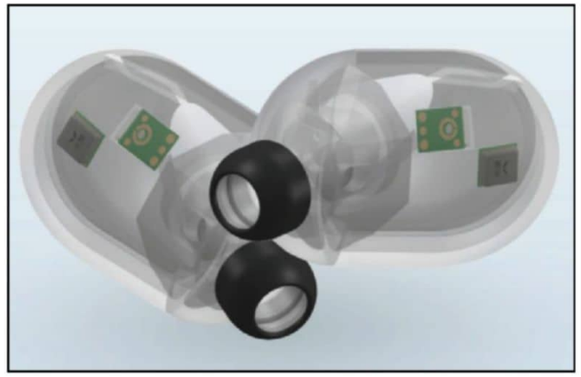
Far-field applications pose a different set of challenges, face similar cost constraints, and are often required to be built on existing generic platforms, which impose significant resources constrains, but that at least offer a comfortable development resource ecosystem.

We often find many different consumer electronics applications benefitting from significant advancements in MEMS microphones and voice front-ends optimized for these requirements in smartphones, TWS earbuds, and miniaturized applications in smart watches and other wearables, being applied to demanding far-field smart home designs, microwave oven applications, and even toys. After all, increasingly, the level of sophistication required for these projects forces engineering teams to resort to familiar development platforms and leverage existing experiences.

As an example, anyone developing hardware for Alexa Voice Service (AVS) support is likely to resort to one of the development platforms and Amazon-qualified "Software Audio Front-End" solutions recommended by Amazon. And many edge designs frequently leverage those very same resources, even if they were intended for always connected systems like smart speakers.

A Year of Groundwork

The research and development activities in this field have strongly intensified in 2021, with the number of employment opportunities also increasing exponentially. LinkedIn is exploding



Vesper's piezoelectric MEMS microphone devices are essential in low power applications, such as in true wireless earbuds designs.

with job announcements for talent in the field and many young engineers are even being encouraged to start their own businesses and offer services and support to multiple projects and clients.

Consumer market data for 2021 continues to be uncharacteristic, as behavior has been atypical. But with nearly two years of data since the beginning of the global pandemic, we can now start to trace and confirm some trends. One of those trends is the acknowledged observation that while significant work continues, the main voice assistants are stalling, with very little progress in accuracy and usability, and very slow progress in language support. In contrast, porting the accumulated knowledge in this field—and valuable trained models—to private, on-device solutions with more limited and highly targeted applications, is the dominant trend. After all, a trash can that recognizes a set of two commands in 40 languages is less of a challenge than an appliance with hazardous risks for users, involving thousands of possible interactions in just one language.

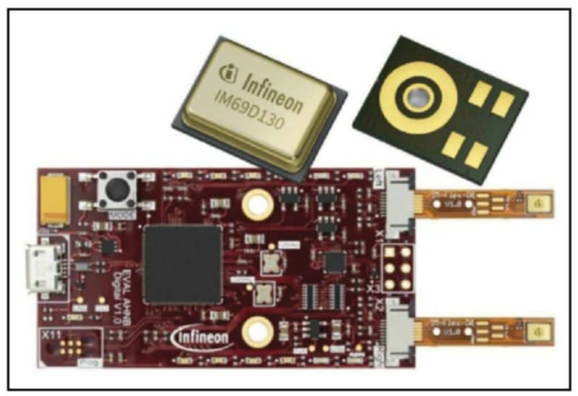
What has improved significantly in the past two years are the quality of the microphones used for these applications and the processing power that is available to support the algorithms.

Many of the application areas that indirectly benefit from these advancements—and can be considered most promising—such as dictation and translation, have not seen much progress in the form of actual products, given the global market circumstances.

An example that should be highlighted—as a trend that I believe will continue to be important for years to come—is that of voice recognition applied to voice communications. Motivated from consumer habits of dictating to their smartphone to send text messages, trust regarding the reliability of voice-to-text has increased enough that many users would not object of having their own real voices replaced by a computer-generated version. Essentially this already happens when smartphone users "speak" to message another user, converting voice-to-text, which in turn the receiver listens to by using the smartphone's own voice speak agent to read the messages.

With more users working from home and in very variable acoustic conditions, the option to use synthesized voices generated directly from voice input becomes an interesting concept—much in the same way as I can already use an animated emoji that is generated by tracking the motion of my own head, instead of showing my smartphone camera video for a call. It's not because we are all admirers of the "metaverse" concept, but simply because there are multiple advantages to completely eliminating communication challenges caused at the source.

This type of voice synthesis is being optimized for voice-overs, transcription, and translation, as *audioXpress* reported previously, and the level of accuracy is now good enough to support this usage. And the use of AI for audio processing and optimization is a related topic that *audioXpress* covered in its November 2021 Market Update discussing Audio Processing Powered by AI. What remains a critical component in the concept is the microphone and voice front-end.



The Infineon Audiohub Nano enables the evaluation of Infineon digital PDM XENSIV MEMS microphones, using Infineon's IM69D130 high-performance Dual Backplate digital MEMS microphones to deliver 105dB dynamic range and high output linearity up to 130dBSPL, with extended pick-up distance and sensitivity to both soft and loud signals.

Growth in MEMS and ECM Microphones

According to SAR Insight & Consulting, the market for MEMS and ECM microphones continues to see strong growth, with 10 billion units shipped in 2020 and that number expected to have been surpassed in 2021. This was driven by increased use of voice interfaces and multiple microphones per device, reinforced by the emergence of TWS earbuds and headphones, which are now the second largest market for microphones.

Smartphones continue to be the largest market for microphones and will remain so for the next five years. Other important market segments include speakers, computing, smart home, and automotive. Many new markets that are becoming significant include medical applications, robotics, toys, banking, and enterprise segments, SAR predicts.

As the market research firm adds, low-cost and robust MEMS microphones will continue to be used in new applications and cause downward pressure to ECM microphone applications. "The ECM microphone market is expected to continue to be eroded by the increasing use of MEMS microphones," Peter Cooney, the report's author, says. "Low-cost MEMS mics are available that compete with ECMs on price, and most new applications choose MEMS for their advantages, such as robustness."

In a related report, "MEMS Microphones Shares: Packaged Microphone Vendors, Microphone Die Vendors, Customers," SAR analyzed the market shares and customers. "The supplier landscape continues to evolve, with major vendors such as Knowles, Goertek, AAC Technologies, and TDK leading the market, and a whole host of others building their market share, particularly Chinese vendors selling into local markets," Peter Cooney adds.

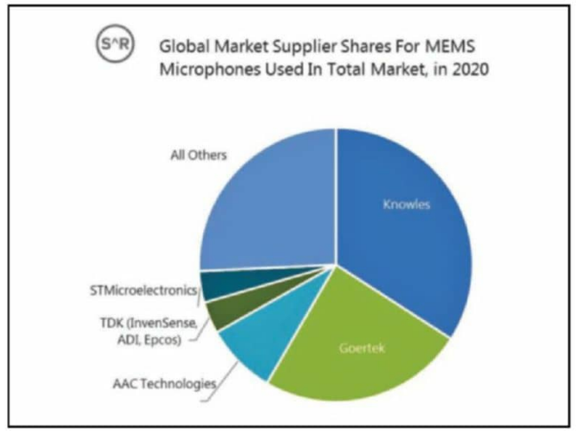
According to SAR, Knowles continues to lead the market in terms of packaged MEMS microphones shipped. Goertek has seen its market share rise rapidly from 2012 to be the second largest packaged microphone vendor. AAC Technologies' share has fluctuated through the years and, more recently, it has diversified its customer base. TDK continues to see success in the market, with a good spread of customers across multiple markets. Its acquisition of InvenSense (which earlier purchased Analog Devices' mic business) was a major win for TDK.

There are other microphone vendors in the market, which have varying success in the market. The market also continues to be affected by the introduction and growth of Asia-based vendors fueling expanding local markets.

Infineon became the number one MEMS microphone die vendor in 2016, and retained the top spot through 2020. It also started to produce its own packaged mics in 2017. Knowles is the second largest vendor of MEMS microphone die through its internal market. Others include Omron, New Japan Radio Company, TDK, and STMicroelectronics.

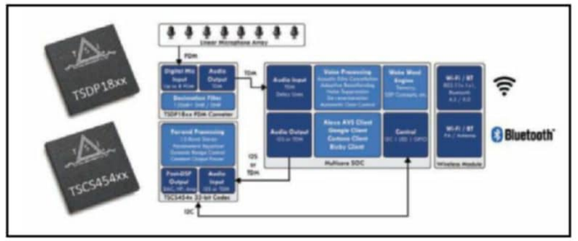
Apple remains the largest user of MEMS microphones, for its iPhone range, AirPods, and other products. Samsung comes a close second, again primarily for its smartphone range but also many other products (e.g., speakers and headphones). Many of the top customers are smartphone vendors but others are also significant, such as Amazon (speakers), Dell (computing), Harman (headphones and speakers), and Bose (headphones and speakers).

Yole Développement, is another market research firm that monitors this segment and confirms the growth trends, as functionalities (e.g., active noise cancellation and beamforming) are expanding the audio market. According to Yole, the MEMS microphone segment will reach a business volume representing $2.5 billion USD in 2026, growing at an annual rate of 10.5%. This increased business is motivated by microphones increasingly being integrated into all types of consumer devices to support smart assistant integration, and integration of audio processing and AI computing functions, the research firm adds.

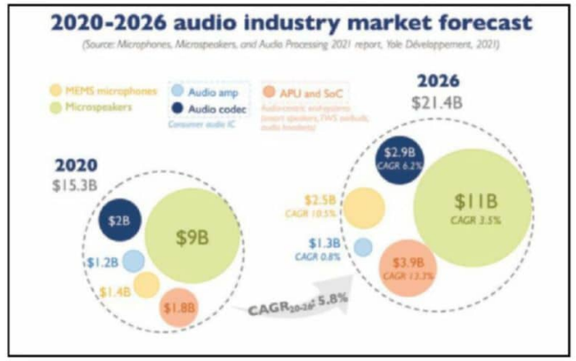


Global market supplier shares for MEMS microphones in 2020, according to data from SAR Insight & Consulting.

While Yole recognizes Goermicro (Goertek), Knowles, and AAC as the uncontested leaders on the microphone side, the French research company identified MEMSensing, Zilltek, and Gettop as part of a group of strong emerging Asian suppliers in the microphone space. MEMSensing combines both MEMS and IC product development, with integration of MEMS foundries, packaging and testing of MEMS microphones, while Zilltek is now offering miniaturized MEMS microphones for advanced ANC applications in true wireless



Tempo Semiconductor, Inc. offers ASIC solutions to support multichannel analog or digital inputs and outputs, including a 8:1 Digital PDM Mic Aggregator (Mic Array) that converts the PDM data into 16-bit or 24-bit Linear PCM in either I²S (1 or 2 channels), Left-Justified (1 or 2 channels) or TDM (1 ~ 8 channels) for voice activated designs.



The 2020-2026 audio industry market forecast according to Yole Développement's "Microphones, Microspeakers, and Audio Processing 2021" report.

earbuds—including a fully differential analog mic with 68dBA SNR and flat frequency response that quickly became a popular choice for ANC applications.

According to Pierre Delbos, market analyst at Yole, TWS earbuds are a fast-growing segment for these components, considering a typical ANC design can use up six microphones per pair of earbuds. "...TWS earbuds are experiencing a particular boom, both in volume and technology. From a sub-million market in 2015, the TWS earbud market is expected to reach 859 million units in 2026, getting closer to smartphone volumes each year, which are expected to be 1.4 billion units in 2026," he states.

The next most promising market applications for microphones, Yole also believes, is voice interaction and noise reduction in automotive applications, a promising field that will also enable new companies in the MEMS microphone space to finally reach large volume production and grab a share of the overall market.

Infineon

Latest Top Performance MEMS Microphones

Infineon Technologies AG continues to maintain a strong lead in the MEMS microphone market, expanding its range of packaged devices. In 2021, the company announced the launch of its latest XENSIV IM67D130A MEMS microphone, featuring a high acoustic overload point (AOP) of 130 dB SPL that allows distortion-free audio signals in loud environments, making it ideal for automotive applications both inside or outside a vehicle. The microphone also has an increased operating temperature range from -40°C to +105°C to enable various use cases in the harsh automotive environments, enabling advanced driver assistance systems and predictive maintenance applications.

This new device combines Infineon's expertise in the automotive industry with its technical lead in high-end MEMS microphones, making the XENSIV IM67D130A microphone qualified for automotive applications according to the AEC-Q103 specifications. This helps simplify the design-in efforts for the industry and reduce the risk of qualification fails.

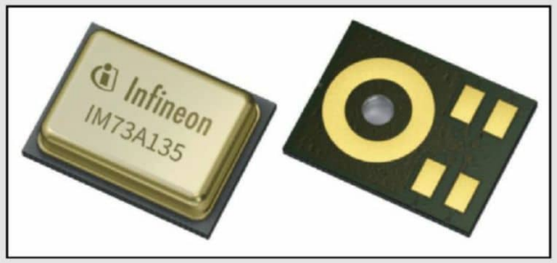
The new XENSIV IM67D130A microphone can be placed inside the vehicle for hands-free systems, in-cabin communication, and to feed all types of sound signals for active noise cancellation (ANC). The microphone is also suited for exterior applications such as siren or road condition detection. The high signal-to-noise ratio (SNR) of 67dB combined with an exceptionally low distortion level leads to optimum speech quality and superior speech intelligence for applications based on speech recognition. Its tight sensitivity matching also allows optimized beamforming algorithms for multi-microphone arrays.

New XENSIV IM73A135

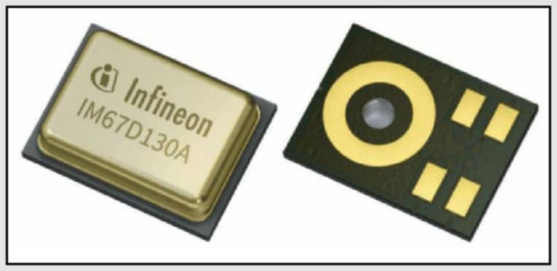
Earlier in 2021, Infineon introduced the XENSIV IM73A135 MEMS microphone, setting a performance benchmark for the company, with a best-in-class SNR of 73dB (69dB A-weighted) and a high acoustic overload point of 135dB SPL (128 dBSPL below 1% distortion). This high dynamic range microphone with a very small footprint (4×3×1.2 mm³) is based on Infineon's Sealed Dual Membrane MEMS technology, which allows for better handling during assembly.

The IM73A135 has a digital (PDM) interface with 6µs group delay at 1kHz, and allows designers to reach a level of high audio performance that was previously only achievable by ECMs while at the same time reaping the benefits inherent in MEMS technology (e.g., tight sensitivity and phase tolerances). Infineon's dual backplate MEMS technology is based on a miniaturized symmetrical microphone design, similar as utilized in studio condenser microphones, and results in high linearity of the output signal within a dynamic range of 105dB. And of course, the flat frequency response (28Hz low-frequency roll-off) and tight manufacturing tolerance allow close phase matching of the microphones, which is important for multi-microphone (array) applications.

www.infineon.com



The new XENSIV IM73A135 MEMS microphone enables designers to reach a level of high audio performance previously restricted to ECMs, while leveraging the inherent benefits of MEMS technology. This new high SNR and high AOP microphone is suited for active noise cancellation in headphones and high-quality audio capturing



The IM67D130A XENSIV MEMS microphone was announced in April 2021 and is available in a PG-LLGA-5-4 package.

Vesper

The Disruptor with Piezoelectric MEMS Microphones

Vesper, the pioneer in piezoelectric MEMS microphones and sensors that is increasingly empowering edge-AI applications in consumer electronics devices, continues to receive investments from multiple players, which already include Bose and the Amazon Alexa Fund among many others. The support is the result of the company's ability to innovate in MEMS microphones, showing strong potential in multiple markets and applications. In 2021, the Boston, MA-based company received the funding required to increase its manufacturing capacities for smart microphones and accelerometers, from tens of millions of units per year to hundreds of millions per year.

In April 2021, Vesper introduced its new low-power vibration sensor VA1200, the world's first analog piezoelectric voice accelerometer (a.k.a. the bone conduction sensor). This new device offers great possibilities in true wireless earbud designs to pick up the wearer's own voice through bone conduction—an option that manufacturers are increasingly combining with standard MEMS microphones, in order to improve voice communications and recognition even under the most challenging circumstances. Using the VA1200 in conjunction with a standard microphone and a sensor fusion solution, the application can achieve superior background and wind noise reduction.

Vesper VM3011 Piezo MEMS Microphone

The company's VM3011 intelligent digital microphone featuring Adaptive ZeroPower Listening, which entered production in late 2020, introduced a new power-saving architecture that enables a 10x extension of battery standby life, combined with best-in-class acoustic performance. Significant to manufacturers, the VM3011 integrates both the ASIC and the piezoelectric MEMS transducer in the same package and is an IP57 device, resistant to dust and moisture and doesn't require a protective mesh, which allows preserving key acoustic performance metrics such as sensitivity and SNR at the system level, ideal for voice-activated smart home designs and wearables.

The Adaptive ZeroPower Listening (ZPL) technology automatically "learns" the acoustic characteristics of the environment in real-time, allowing the system to ignore background noise and only wake in response to keywords or other desired acoustic triggers. By automatically adjusting the background threshold based on the acoustic level of the environment, this allows the system to hibernate over 90% of the time and extend battery life >10x.

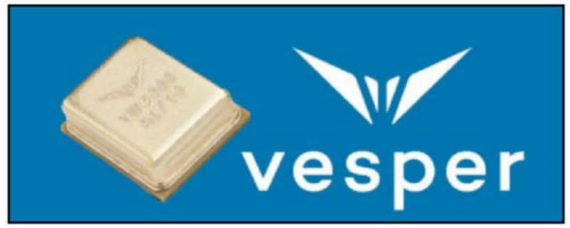
When combined with the ultra-fast startup of piezo microphones, the VM3011 enables unprecedented system power savings for voice-activated devices without compromising the acoustic performance. The VM3011 uses a bottom port and integrates both the ASIC and the piezoelectric MEMS transducer in the same package.

A digital PDM interface is available to stream the audio data while an I²C interface allows "on the fly" configuration of the Adaptive ZPL. The VM3011 uses the PDM clock speed to change device modes. The DATA pin outputs PDM output in normal mode operation. In Wake on Sound (WoS) mode, a Digital Output (DOUT) pin outputs a single bit "Threshold Exceeded" flag when the acoustic input is greater than the set threshold. When the required threshold is reached, the microphone sends an interrupt signal to the external codec or voice processor using the DOUT pin.

The microphone is constantly listening to the environment with just 10µA of current consumption. Adaptive Mode continuously measures the average sound level of the environment and adjusts the internal WoS threshold comparator until it converges to the threshold margin configured using the I²C register. This margin can be set to a range 6dB to 18dB above the average acoustic noise level.

This technology is ideal to power a new class of products, such as always-listening far-field TVs, always-listening smartwatches, home security cameras and doorbells, and many other systems.

www.vespermems.com



The new VM3011 MEMS microphone further extended Vesper's technology leadership in the low-power always-listening device market, and expanded applications for the company's patented piezoelectric MEMS transducer technology.

Alango Technologies

VoiceDefender Advanced Four-Mic Voice Processing Technology

Based in Israel, Alango Technologies is a technology company that should be on the map for any manufacturer looking at differentiating audio and voice products. Alango is strongly focused on voice communications, particularly in voice quality, intelligibility, and hearing enhancement. It is also a leading developer and licensor of front-end DSP technologies for voice communication and mobile audio.

But no matter how sophisticated the DSP algorithms can be, there's a constant need to adapt existing concepts to the specific needs of new product developments, particularly in the crowded and highly competitive consumer space. When looking specifically at consumer headphones, advanced features (e.g., active noise cancellation, voice assistant integration, and low latency streaming) are all in high demand. And stellar sound quality and battery life can't hurt, either.

Bluetooth headphones have come a long way since the first

A2DP stereo headset hit the market in 2004. But while today's headphones are leaps and bounds ahead in terms of sound quality, comfort, and battery life, they often fall short of our expectations when it comes to voice processing. The ability to make clear, noise-free voice calls from any environment is no longer seen as a "nice to have" feature—it is a "must-have."

But what about transmitted voice quality? Can headphones meet the challenges posed by noisy and windy environments (e.g., a bus, train, busy street, or café)? Hands-free communication with on-ear/over-ear headphones is convenient, but only if the person on the other end of the call—or the voice assistant—hears and understands what is communicated, even when speech originates from a noisy environment. And although boom microphones can improve voice pickup on headphones that use them, they are not necessarily suitable for every headphone design.

That is why Alango launched its VoiceDefender solution, employing microphone beamforming on four microphone signals (two per earcup) with a combination of end-fire and broadside beamforming to create a single enhanced output. As Alango illustrates, "With VoiceDefender, the user's voice is "defended" from ambient noise with a sphere of protection."

Alango's VoiceDefender virtually eliminates dominant noise sources and highlighting voice and speech signals, effectively taking augmented audio to a whole new level. And leveraging Alango's experience and solutions, VoiceDefender is a complete solution for hands-free communication that includes echo cancellation, multi-microphone noise suppression including wind noise reduction, equalization, and gains management technologies.

Compared to dual-mic beamforming on a single earcup, VoiceDefender adds an additional dimension of voice pick-up. A combination of end-fire and broadside beamforming is used to create an output for clear transmission of the user's voice even in high noise or windy conditions. With VoiceDefender, the clarity of voice pick-up in multiple environments can become a distinguishing feature of a product development, and a competitive advantage in a crowded market.

Additionally, a project implementing VoiceDefender can be also complemented with Alango's technologies for headphones, including Automatic Volume & Equalization (AVQ), ListenThrough, Voice Activity Detection (VAD), and Sound Effect Normalization (SEN), all of which are essential elements of an augmented audio experience.

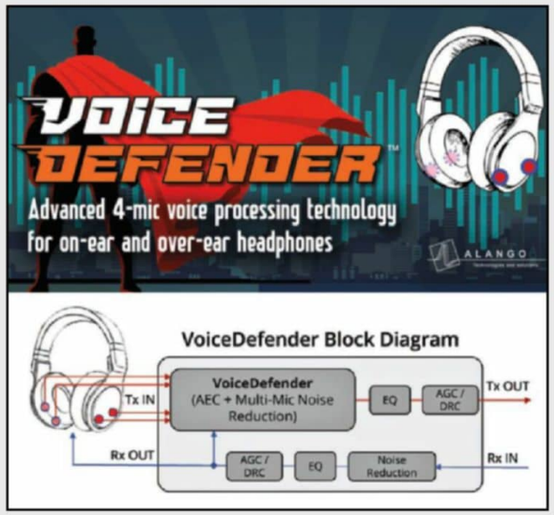
AVQ empowers headphones to offer a comfortable listening experience in conditions with dynamically changing background noise. Using the device's microphone, AVQ continuously analyzes both the level and spectrum of ambient noise in the user's environment. Accordingly, AVQ amplifies and equalizes the music to provide equal perceptual loudness, intelligibility, and sound coloration in conditions where noise level and spectrum change dynamically.

ListenThrough is a smart algorithm for headphones that adds environmental alertness during music playback. ListenThrough filters in important environmental sounds (e.g., alarm, car horn, train, etc.), which are then amplified and mixed with the music. Unlike simple "transparency mode" algorithms, ListenThrough does not let every sound in. It also filters out constant-level background noise, thus allowing the user to be aware of their surroundings while still maintaining the sound isolation provided by the headphones.

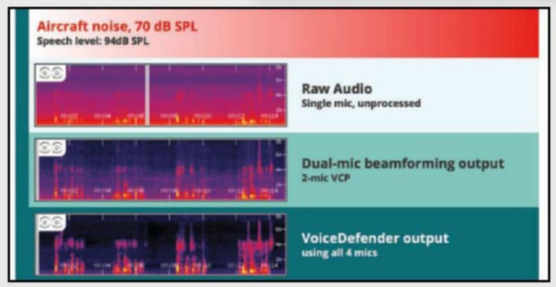
Alango's VAD technology reliably detects human speech in an acoustic signal. The technology is based on a proprietary, high-resolution spectral noise estimation algorithm operating in real time. VAD's sensitivity level is adjustable and ensures minimization of false positive detections in a presence of background noise and other non-speech sounds. In headphone applications, VAD enables always-on voice detection at low standby power by allowing MIPS-heavy signal processing to "sleep" until voice is detected.

And Alango Sound Effect Normalization (SEN) technology was developed to improve speech intelligibility in movies through de-emphasis of background sounds and music—reducing listening fatigue for the headphones user.

www.alango.com/voicedefender



Alango VoiceDefender is a new advanced voice enhancement signal processing solution, ready to be used by developers and manufacturers in new on-ear and over-ear headphone designs who would like to stay away from using ugly boom microphones.



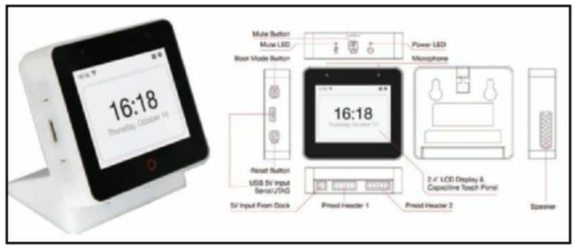
On its website, Alango offers multiple examples of VoiceDefender processing with aircraft noise, call center, automotive and transportation

Espressif Systems

AI Voice Dev Kit Targets Smart Speakers

Espressif Systems has announced the launching of its AI voice development kit, ESP32-S3-BOX, which provides a platform for developing the control of smart devices with offline and online voice assistants. ESP32-S3-BOX is well-suited for developing connected device applications with reconfigurable AI voice functions (e.g., smart speakers) and edge devices that achieve human-computer voice interaction directly. Interestingly, the ESP32-S3-BOX platform combines a touchscreen controller, various sensors, an infrared controller and a smart gateway, saving on development costs and time.

The ESP32-S3-BOX is based on Espressif's ESP32-S3 Wi-Fi + Bluetooth 5 (LE) SoC, with AI capabilities. In addition to ESP32-S3's 512KB SRAM, the ESP32-S3-BOX comes with 16MB of QSPI flash and 8MB of Octal PSRAM. The 2.4" display is a 320×240 resolution, capacitive touchscreen, and there is a dual microphone system, a speaker, and two Pmod-compatible headers which allow for the extensibility of the hardware.



And very important for development, the ESP32-S3-BOX also uses a Type-C USB connector that provides 5V of power input, while also supporting serial and JTAG debugging, as well as a programming interface; all through the same connector.

The ESP32-S3-BOX kit is designed to take advantage of ESP32-S3's AI acceleration capabilities, ideal for online and offline voice assistants, voice-enabled devices, HMI touch-screen devices, control panels and multi-protocol gateways, using Espressif's set of SDKs and AIoT solutions.

More specifically, the ESP-S3-BOX can run Espressif's Amazon-qualified Software Audio Front-End solution, Alexa for IoT SDK, the ESP-Skainet offline voice assistant, and the LVGL-based HMI solution, in addition to other SDKs, too, such as ESP-DL, and ESP-ADF. Espressif's complete AIoT platform, ESP RainMaker, can be used with the development kit for configuring GPIOs and offline commands, while providing control via phone apps and/or a voice assistant.

Espressif provides a number of technical resources relating to the ESP32-S3-BOX, including the hardware reference design and user guides.

www.espressif.com

ArkX Labs

EveryWord Ultra-Far-Field Solutions

Featuring Cirrus Logic's SoundClear and FlexArray technologies, ArkX Labs' portfolio for voice includes the EveryWord Audio Front End (AFE) Module, EveryWord Voice Module (System-on-Module + AFE), EveryWord Development Kit (which *audioXpress* detailed in its January 2021 edition), and EveryWord Software Stack. ArkX's production-ready voice capture solutions deliver far-field voice capture and a robust voice experience to consumers by capturing voice commands from three times the standard distance, around corners, in noisy and reflective environments, and without lowering playback volume. Additionally, the company's EveryWord technology provides the ability to identify and suppress speech from TV or other single-point noise sources present in the same space.



EveryWord technology does not require source-ducking for reliable interaction, provides linear, circular, square, triangular, or 3-D mic array geometries, and requires fewer microphones. The technology features ultra-low power battery operation for wake-on-word, and the flexibility for placement of microphones allows for in-wall, ceiling, or dashboard solutions. The company's 3D mic array (unlike linear beam-forming approach) enables fewer blind spots and increased performance while incorporating fewer redundant microphone arrays for coverage.

All ArkX solutions are Alexa-compatible and meet or exceed all requirements for the Amazon Voice Services (AVS) qualifications. In addition to Alexa, EveryWord is compatible with other platforms such as Google, Siri, Cortana, AliGenie, Baidu/Kitt.ai, Tencent, and Sensory. EveryWord voice solutions can also be customized for a company's ecosystem and applied to various types of products, from speakers and soundbars to appliances. The modules can be installed in hubs, ceilings, and in-wall for smart home or office applications.

ArkX Laboratories is a joint venture between product development pioneer Surfaceink and consumer electronics manufacturer Ark Electronics USA. The Silicon Valley, CA-based company offers advanced, high-performance far-field voice capture solutions, featuring Cirrus Logic and NXP technology, and is Amazon pre-qualified and production-ready.

www.arkxlabs.com

TDK InvenSense

New Design Options with SmartSound MEMS Microphones

In 2021, TDK introduced three new MEMS microphones as part of the SmartSound family of products for mobile and consumer connected devices. The new TDK T5919 is a high-performance PDM digital MEMS microphone and is the first with built-in particle ingress filter (PIF) for harsh environments. The new T3903 is a high-performance PDM digital MEMS microphone with very low power and wide dynamic range, while the T4086 is an analog output MEMS microphone in a tiny package, ideal for very space constrained applications.

The TDK T5919, T3903, and T4086 MEMS microphones are part of a fully compatible and multi-grade SmartSound portfolio of microphone products from TDK InvenSense. Each of these new MEMS microphones pushes the boundaries of microphone acoustic performance, providing advanced feature sets with very low power in small package footprints. All models are compatible with Sn/Pb and Pb-free solder processes, and are RoHS/WEEE compliant.

All models offer high acoustic overload point (AOP), high SNR, and wide dynamic range and are ideal for environments that shift from very quiet to very loud, such as far-field voice pick-up barge-in for Smart Speaker applications.

The new T5919 is the world's first PDM digital output microphone with built-in Particle Ingress Filter (PIF). This is a multi-mode, low noise digital MEMS microphone in a small 3.5mm × 2.65mm × 0.98mm bottom port package. The PIF filter is embedded in the microphone package, providing protection to particles and moisture. The T5919 supports multiple modes of operation, from High Quality to Low-Power (AlwaysOn), and Sleep, while maintaining high SNR in all operational modes. The microphone maintains 135dB SPL AOP in High Quality Mode and 120dB SPL AOP in Low-Power mode.



TDK's SmartSound family of MEMS microphones continues to expand gradually with improved features and specifications, meeting the most stringent requirements.

The new T4086 model is an ultra-small analog output MEMS microphone, designed to achieve high SNR and AOP at low power in a tiny 2.75mm × 1.85mm × 0.9mm bottom port package to support very space constrained applications/devices.

Finally, the T3903 is a very low power wide dynamic range PDM MEMS microphone, enabling the transition of microphones from analog output to digital output to support always-on applications. The T3903 is a multi-mode, low noise, bottom ported MEMS microphone with -37dBFS sensitivity, 66dBA SNR, and 133dB AOP in a 3.5mm × 2.65mm × 0.98mm package.

Lowest Power PDM Microphone with Acoustic Activity Detect

Announced earlier in 2021, the new T5838 SmartSound MEMS microphone offers a low-power mode of just 130uAmps and a high-quality mode of 330uAmps, while supporting a dynamic range of 68dN SNR, and 133dB AOP with 20kHz bandwidth.

This multi-mode MEMS microphone with PDM output added Acoustic Activity Detect (AAD) capability in an industry-standard footprint and was an important breakthrough in the InvenSense SmartSound product family. The T5838 offers an outstanding low current of 330µA when used for high-fidelity audio applications with an extended frequency response from 35Hz to >20kHz. And the T5838 adds Acoustic Activity Detect (AAD) capability, which is a new ultra-low-power edge processing feature where the microphone monitors the acoustic environment and wakes up the SoC or application processor when activity is detected.

It is available with three modes of operation starting at just 20µA total current draw from the microphone. It provides user programmability to apply various filters and thresholds for optimized performance for each application. It also supports 8kHz bandwidth output for "Always-On" systems requiring audio buffering for cloud processing.

www.invensense.tdk.com



Responding to market demand, TDK InvenSense introduced the new T5838 multi-mode MEMS microphone with Acoustic Activity Detect (AAD) capability.

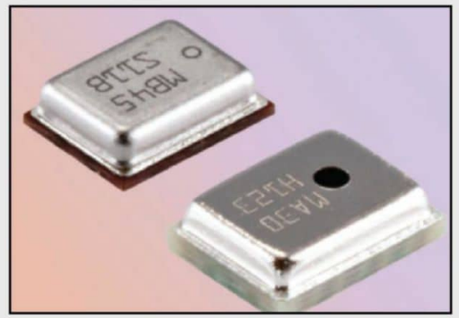
CUI Devices

MEMS Microphones with Digital I²S Output

Oregon-based CUI Devices' Audio Group announced the addition of two new digital I²S output models to its comprehensive line of MEMS microphones. The bottom port CMM-3526D-261-I2S-TR and top port CMM-4030D-261-I2S-TR are housed in extremely compact, low-profile packages measuring 3.50mm × 2.65mm × 0.98 mm and 4mm × 3mm × 1.25 mm, respectively.

Thanks to their digital I²S interface, these MEMS microphones eliminate the need for a codec, allowing for the microphones to connect directly to digital signal processors and microcontrollers in an end application.

The component savings and simplified integration allowed by adding a digital I²S interface make these MEMS microphones ideal for audio recording and voice capture in space-constrained designs such as smartphones, tablets, smart home devices, wearables, and more.



Both microphones are omnidirectional and feature -26dB sensitivity ratings, operating frequencies from 100Hz to 10,000Hz, and current consumption of 400µA. The CMM-3526D-261-I2S-TR further carries a SNR of 64dBA, while the CMM-4030D-261-I2S-TR offers a SNR of 59dBA.

As reflow solder compatible components, these digital I²S MEMS microphones afford designers additional flexibility where surface-mount assembly is required. Both I²S models are also well suited for array applications used to perform beam forming and noise cancelation functions due to their tight sensitivity tolerances of ±1dB.

www.cuidevices.com

Knowles

75 Years of Voice and Audio Technology Integration

In 2021, Knowles Corp. celebrated the company's 75th anniversary. From the first Apollo mission in 1969, where Neil Armstrong spoke into a Knowles microphone, Knowles pioneered many industry-firsts including balanced armature drivers for the first hearing aids in 1954, and the first MEMS microphones used in everything from the Mars 2020 Perseverance Rover to the latest true wireless earbuds. Today, Knowles products and solutions are in smartphones, laptops, wearables, smart speakers, or any device with voice control.

Knowles contributions have been integral to the evolution of hearing health solutions and the massive advancements in voice and audio applications experienced in consumer electronics over the last two decades.

Pioneering the MEMS microphone technology under the SiSonic brand in the early 2000s, Knowles created a new de facto standard for microphones addressing several significant challenges afflicting prior technologies. The SiSonic MEMS microphones produced in two state-of-the-art manufacturing facilities in Suzhou, China and Penang, Malaysia set the industry benchmarks for performance, quality, and scale and remain a preferred choice by leading OEMs around the world.



"The modern world depends on the technology that Knowles delivers. One could easily interact with over 20 Knowles components inside the devices that they use every day. Your cell phone, laptop, tablet, smart watch, headphones, smart speaker, car, and anything that has voice control all rely on Knowles technology. As the inventor of MEMS microphones and balanced armature receivers, Knowles has developed technology competencies and a high level of quality that has been forged through many years of leadership in the hearing health industry," says Jeffrey Niew, President and CEO of Knowles.

New Automotive-Grade MEMS Microphones

In 2021, Knowles announced two new SiSonic MEMS microphones for automotive applications. The new best-in-class analog and digital microphones—Knowles SPH1878 and SPH9855, respectively—were engineered to support the increasing demands of the automotive market for hands-free calling, advanced voice assistance and in-cabin noise cancellation for passenger comfort. Signaling Knowles' commitment to the automotive market, the microphones meet AEC-Q100/103 qualification requirements set by the Automotive Electronics Council for components used in the harsh automotive environment.

The new Knowles SPH1878 microphone features both single-ended and differential analog interface, while the SPH9855 features a PDM digital interface. The two microphones are designed for beamforming applications and are compatible with the European eCall communication system. Key features include low phase



distortion for superior noise cancelling performance, high dynamic range for enhanced call and sound quality, an extended operating temperature range for automotive applications, and improved manufacturing stability and traceability.

As the MEMS sensors, integrated circuits, and packaging for the microphones are all designed in-house, Knowles is in a unique position to offer best-in-class engineering support to brands and product design and development companies through its global field application team anywhere in the world.

www.knowles.com

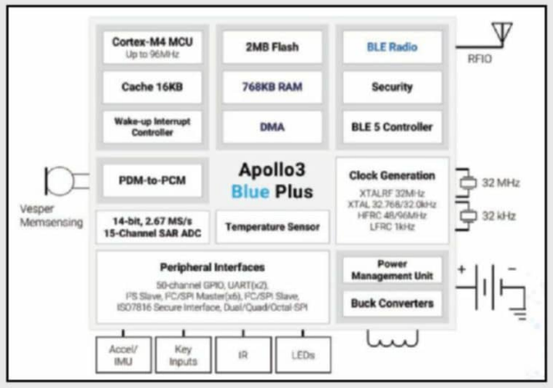
Ambiq

Ultra-Low-Powered Development Solution for Voice-Enabled Connected Devices

In May 2021, Ambiq, a leading technology provider of ultra-low-power microcontrollers (MCU), System-on-Chips (SoC), and Real-time Clocks (RTC), introduced the new Ambiq Voice-on-SPOT (VoS) Kit, designed for manufacturers to introduce voice-command into their IoT devices with faster time-to-market. The Ambiq VoS Kit offers a complete, always-on, always-listening, voice-to-cloud solution for local voice commands and/or voice assistant integration in battery-powered connected devices. Targeted applications include remote controls, gaming controllers, smart sensors, smartwatches, remote mics, and health trackers.

Aimed to deliver the complete ultra-low-power solution at both the MCU and the system levels, the VoS Kit integrates Ambiq's hardware and software with peripherals and third-party IP, including signal processing using DSP Concepts' Audio Weaver, Sensory VoiceHub, and Retune DSP VoiceSpot.

This VoS Kit is built on the Apollo3 Blue Plus MCU, and supports Always-on-Voice with application and cloud service driven options for one or two mics, signal processing, wake word/command detection, codec, and Bluetooth LE (BLE) communication. The next-generation Apollo4 Blue VoS Kit, targeted to be available in 2022, will further lower always-on power with two analog mic inputs, while increasing capabilities with up to eight digital mic inputs, 120dB SNR pulse density modulation (PDM) to pulse code modulation (PCM) converter, 192MHz Arm M4F MCU, highly precise clocking and advanced interconnectivity with hardware rate matching. This will expand Ambiq's approach for demanding voice input in phone calls, translation, and audio conferencing applications.

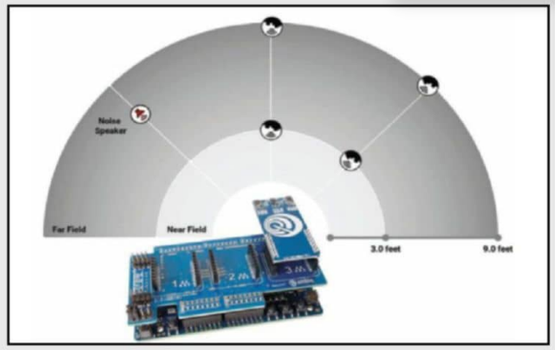


Apollo3 Blue Plus Voice-on-SPOT (VoS) Kit Block Diagram.

"Demand for voice-enabled products is at an all-time high and we've seen a lot of interest from brands looking to add voice to compact devices like smartwatches and remote controls. These applications of voice UI make perfect sense when you think about how often we use these products, but the size and power constraints present real design challenges. Ambiq is widely-recognized as a leader in ultra-low-power MCUs, and their VoS Kit is an attractive solution for product makers looking to overcome these obstacles and add reliable voice UI to compact devices," states Paul Beckmann, the founder of DSP Concepts.

Through its advanced Subthreshold Power Optimized Technology (SPOT) platform, Ambiq has already helped many leading manufacturers worldwide create products that can operate for days, months, and sometimes years on a lithium battery or a single charge.

www.ambiq.com



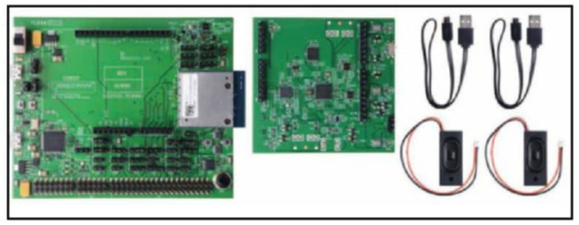
The current Apollo3 Blue Plus Voice-on-SPOT (VoS) Kit facilitates demos, evaluation, and the development of ultra-low-power audio and voice capabilities on the Ambiq SoC. The kit can be configured with or without pre-roll, with a Vesper microphone that keeps the system asleep until it hears sound above the ambient noise level.

OKdo and Qualcomm

Easy Development Platform for Amazon Voice Services

OKdo, a global technology company from Electrocomponents, is now supplying the Qualcomm Home Hub 100 Development Kit for Amazon Voice Services (AVS), a ready-to-use environment for voice-controlled applications in connected devices. Based on Qualcomm's highly integrated, multi-mode System-on-Chip (SoC), the QCA4020, the Home Hub 100 kit is provided with pre-integrated software support for AVS, to help kickstart the development of voice-controlled applications.

With built-in software support for AVS, including Wake-on-Voice (WoV), Acoustic Echo Cancellation (AEC) and far-field noise suppression, the Qualcomm Home Hub 100 Development Kit for AVS includes three MEMS analog microphones arranged to provide a wide angle of audio detection. The kit also features eight sensors to support various connected applications, including: an ambient light sensor; a 3D accelerometer; a 3D gyroscope; a 3D magnetic field sensor, a temperature sensor, a humidity sensor, a pressure sensor, and a PID sensor.



Further expansion is supported through a header that provides direct connection to the SoC's GPIOs and ADC channels. An Arduino-compatible header is also included to give developers even more flexibility.

The multicore, multi-mode QCA4020 SoC supports concurrent Wi-Fi, Bluetooth 5, and IEEE 802.15.4 (Zigbee, Thread) wireless connectivity, by integrating separate processing cores for the Wi-Fi and Bluetooth/15.4 interfaces. The application software also runs on a dedicated core; an Arm Cortex-M4F, which can be accessed using a number of development environments. A JTAG-over-USB interface is also included, for programming and debugging. Two micro-USB interfaces are provided for supplying power and connecting the development board to a host PC.

www.OKdo.com

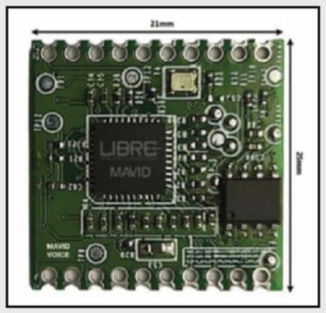
Libre Wireless

Knowles on MAVID Voice AI Platform

With voice interface and control quickly becoming a familiar method of interface for smart products, brands and smart product developers need scalable options to enable voice across their entire product lines. Leading embedded wireless voice/AI solutions provider Libre Wireless Technologies announced a strategic alliance with Knowles Corp. to jointly promote and market new solutions based on the Multiprotocol-Audio-Voice-IoT-Devices (MAVID) platform. These miniature-sized and low power mic-to-cloud edge voice/AI computing solutions now feature Knowles' micro-acoustic microphones and speaker technology, optimized for smart product development.

The MAVID family enables fully certified cloud-based voice solutions with Wi-Fi, Bluetooth, and Zigbee options, down to simple local voice control—no network or cloud required. Through the collaboration with Knowles, these Libre Wireless solutions are now available in completely optimized systems, delivering the latest in microphone and audio/voice processing technology and enabling a faster time to market. These solutions also help product developers deal with the complexities of voice and leverage Libre's acoustic lab and experience tuning and optimizing voice applications.

The two featured MAVID solutions designed in collaboration with Knowles include the MAVID 3M and the new MAVID Voice, a unique low-cost local voice solution enabling voice interface and control without the complexity of a wireless network or cloud setup. MAVID Voice greatly simplifies development, delivering the convenience of voice control to many new product categories. With Libre's SDK and voice firmware tools, custom wake words, languages, and voice control features can be customized and integrated into the system, and custom solutions can now be integrated in a matter of days.



The alliance partnership between Libre Wireless and Knowles leverages both companies' expertise in voice-activation control, and far-field speech recognition

With no need for Wi-Fi or BLE connectivity, there are no recurring charges for cloud services. All voice performance and tuning is provided by Libre allowing for a "light-touch" approach for brands and developers. MAVID Voice also allows support for device control peripherals including UART, SPI, I²C and GPIO, ideal for domestic appliances and devices with a limited set of controls. A proof-of-concept and demo platform is available on request.

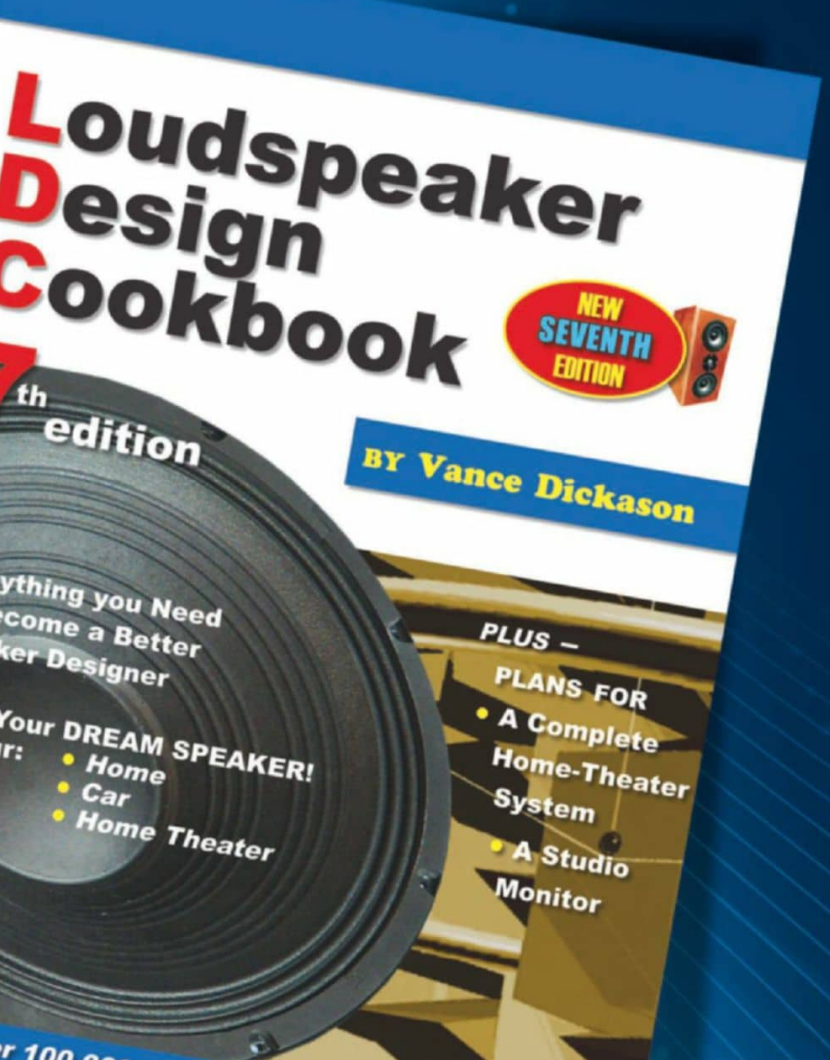
MAVID 3M is a low-cost IoT module supporting Wi-Fi, BLE, and AVS or local voice control. With MAVID 3M embedded within a product, a user can control third-party devices using voice, including from remote locations. Along with integrated wireless connectivity, the solution offers the necessary Voice DSP capability to deliver near- and far-field detection, also including advanced acoustic echo cancellation and noise reduction. This extremely low-power voice solution is ideally placed for battery operated devices.

For development with the MAVID 3M solution, OEMs and ODMs can take advantage of a comprehensive SDK with peripheral access, AWS Cloud APIs, allowing customization without a big investment and support. The standard SDK also supports over-the-air fleet updates and fleet provisioning options.

www.knowles.com | www.librewireless.com

The "Must Have" reference for loudspeaker engineering professionals.

Home, Car, or Home Theater!



Back and better than ever, this 7th edition provides everything you need to become a better speaker designer. If you still have a 3rd, 4th, 5th or even the 6th edition of the *Loudspeaker Design Cookbook*, you are missing out on a tremendous amount of new and important information!

Now including: Klippel analysis of drivers, a chapter on loudspeaker voicing, advice on testing and crossover changes, and so much more! Ships complete with bonus CD containing over 100 additional figures and a full set of loudspeaker design tools.

A $99 value!

Yours today for just $39.95.

Shop for this book, and many other audio products, at **www.cc-webshop.com.**

