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A Smart Future for Voice and Hearables

In this edition of *audioXpress,* we focus on two of the most important technology trends in today's audio industry. Truly wireless earbuds and voice assistants are two of the fastest growing product segments right now, and their effects on the industry will probably last for a very long time.

A new report just published by Juniper Research forecasts that hearables will become the new standard for personal audio in less than 5 years, and that more than 50% of all wireless headphones will incorporate Hearable Voice assistants, touch sensors, and other interface technologies.

It's just a natural transition to go for in-ear devices that are free of wires, as is the use of voice interfaces and digital assistants in everything, from today's smart speakers to smart TVs and appliances, and increasingly the complete home automation systems.

As distinct as the two product classes might seem today—where voice is still predominantly present in the so-called smart-speakers—the development of the essential technologies that enable these two product categories are sharing many essential components and will inevitably converge.

I don't believe it will make much sense to continue having a separate category called smart-speakers in the future, once all speakers become "smart." Today, we have this novelty of small speakers that are essentially being used to promote the use of voice digital assistants and incentivize the adoption of services from Amazon, Google, Apple, Baidu, Tencent, and others. In the near future, all speakers, portable, floor-standing, musicians' stage monitors, and even large-size line arrays for touring in the largest arenas will also be "smart." Meaning they will be connected, include a sophisticated computer inside, and will be equipped with both microphone arrays and signal processing capabilities, enabling real-time adjustments to any acoustic environment or usage scenario. According to the demands of the situation, those speakers will be able to adjust automatically to deliver the best possible reproduction, sometimes using shared resources available through the cloud, sometimes purely based on their own built-in "intelligence."

And I envisage a scenario where home users and audio engineers alike will not need to voice-interact with those systems much, since the systems themselves will talk to them when needed.

If today we associate Amazon Alexa or Apple Siri to services we need to "call" and ask something, very soon those AI-powered assistants will be talking to us instead.

And nowhere will that service be more useful than personal digital assistants "talking to us" on our hearables, and turning into really useful "augmented-hearing" assistants. Think about it. These connected devices will know we are looking for a restaurant and we have a meeting with an important client there. The assistant will guide us with voice navigation instructions when needed, will remind us of the latest information we need to have in mind for the meeting, and will anticipate if we want to hear Beethoven's "Symphony of Triumph" while walking/driving there. We will very rarely have to use OUR voice as an interface (not very useful or practical when close to other people) but instead the assistant's voice will talk to us precisely when we need it the most. Not shouting: Alexa, what time is it in Honk Kong? But instead having our digital assistant, reminding us gently, "You need to call Laura in Hong Kong in 15 minutes. This will give you time to review the sales figures. Here are the latest figures Tom just emailed you..."

All the technology pieces needed for these scenarios are exactly what's powering today's pioneering efforts in voice, machine learning, AI, digital signal processing, low-power embedded electronics, sensors, fast-charging batteries, wireless networks, and more. But also, low-power highly sensitive MEMS for those microphone arrays, extremely efficient audio amplifiers closely tied to signal processing, and highly efficient transducers that are able to beam sound exactly where needed and nothing else, avoiding reflections in rooms and including the ability to track and follow a listener in a room, if needed.

All the parts of the equation are needed.

J. Martins
Editor-in-Chief

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COLUMNISTS

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

Mike Klasco is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is the organizer of the Loudspeaker University seminars for speaker engineers. Mike specializes in materials and fabrication techniques to enhance speaker performance.

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

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

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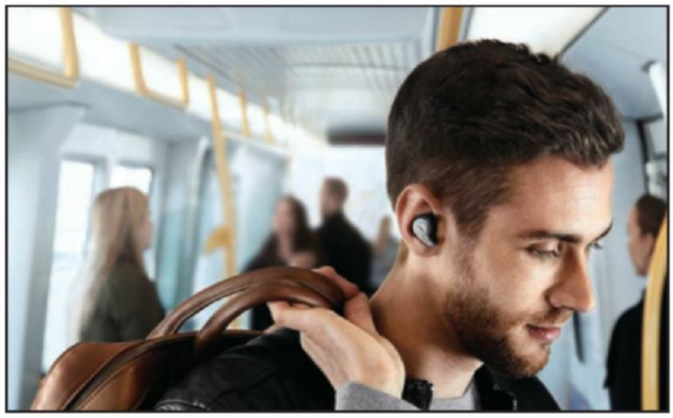

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COVER ART

Photo Credit: Cover photo courtesy of Earin. Pictured are the Earin M-2 true wireless stereo earphones using Knowles balanced armature drivers and featuring NFMI MiGLO® technology by NXP®

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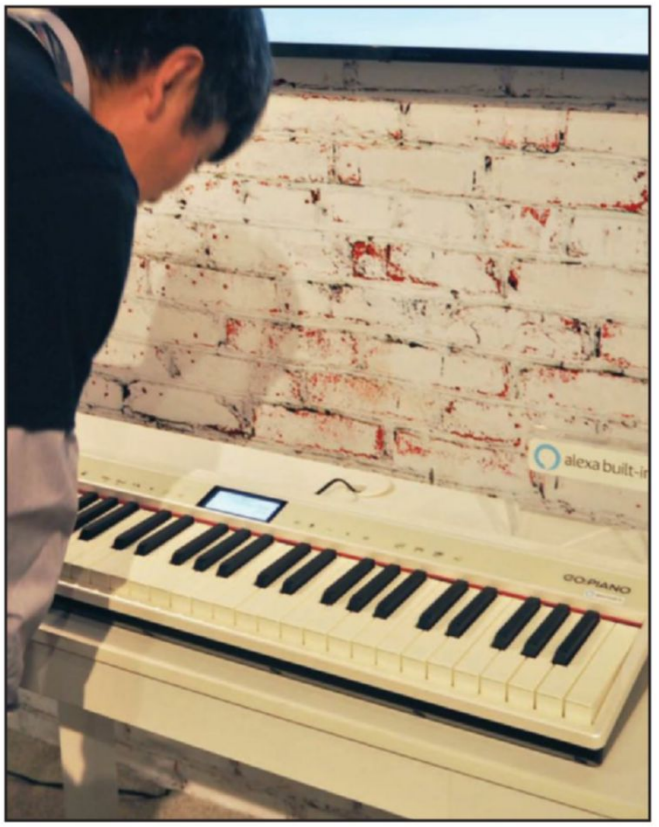

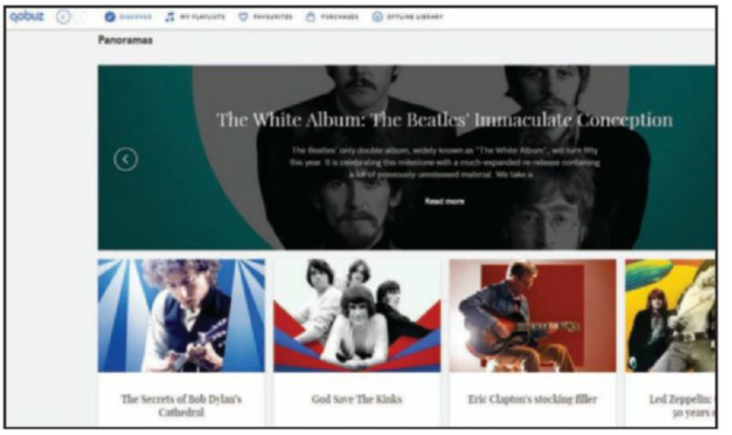

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Maximize Your Hearing

By
J. Martins
(Editor-in-Chief)

The headphones market continues its recent trend of solid growth in volumes and substantial growth in revenue. North America was the largest headphone market in 2018, accounting for more than 30% of global demand. True Wireless Stereo is the fastest growing category.

There's a lot of activity in the headphone, headset, in-ear, earbuds—wired and wireless—product segments, with the audio industry abuzz with new designs and expanding its product ranges in all directions. Reasons for excitement? Well, according to all the market research companies, the headphones product category is showing more consistent growth and there are many highly promising new opportunities, from true wireless stereo, to hearables and the convergence with hearing assistants and hearing aids. This Market Update tries to illustrate some of those key market trends and highlight a few important ongoing technology efforts.

A Growing Market

According the to the annual "Headphones Market Report" published by market research firm Futuresource Consulting, the headphones market segment continues to see solid growth, both in units sold and revenue.

In 2018, the worldwide headphone market has grown 4% in unit sales, while retail value continues to grow much faster, generating revenues worth $20.8 billion. Futuresource has even increased the longer-term outlook to 422 million units by 2022 (from previous projections of 400 million) as a result of growing demand from emerging markets.

Retail revenues are also forecast to grow by a compound annual growth rate (CAGR) of 11% (higher than previously expected at CAGR 5%) to generate $31.7 billion. "Growth in true wireless, voice technology, noise cancelling, and an overall improvement in the durability and quality of headphones will encourage growth in prices throughout the forecast period," Futuresource predicts. Emerging opportunities in areas such as voice assistants, voice translation, and the—about to explode—category of hearing enhancers or over-the-counter hearing aids could even improve this outlook.

As the Futuresource report confirms, consumers are demanding more features and willing to upgrade to new in-ear, wireless and true wireless models. Even the over-ear segment, fueled by demand for wireless is being propelled by noise-cancelling technology, showing that consumers are willing to spend more to get a better audio experience.

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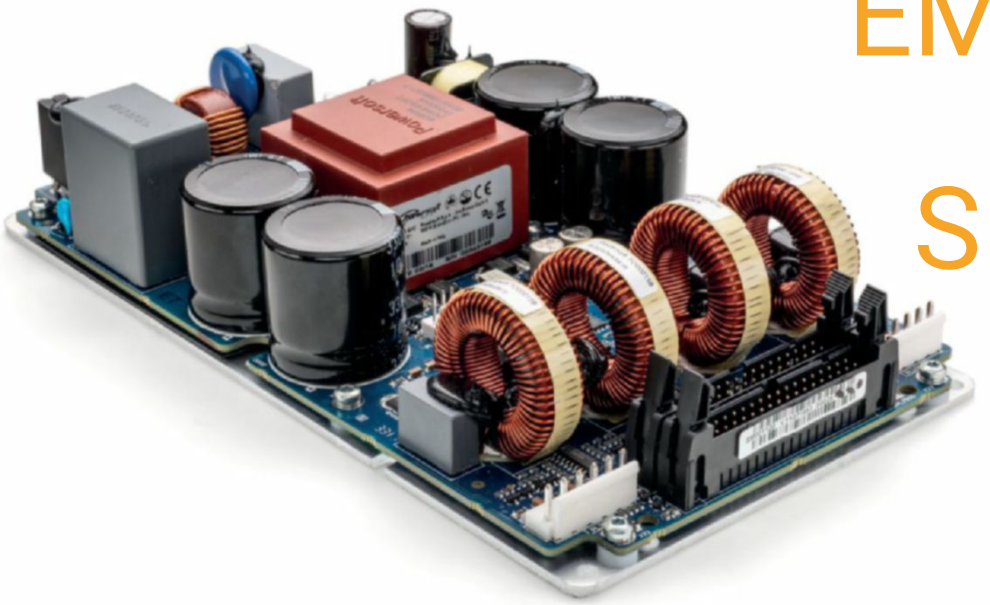

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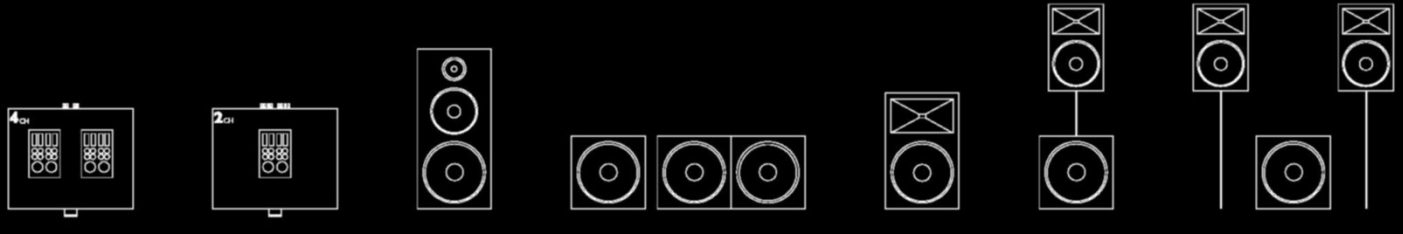
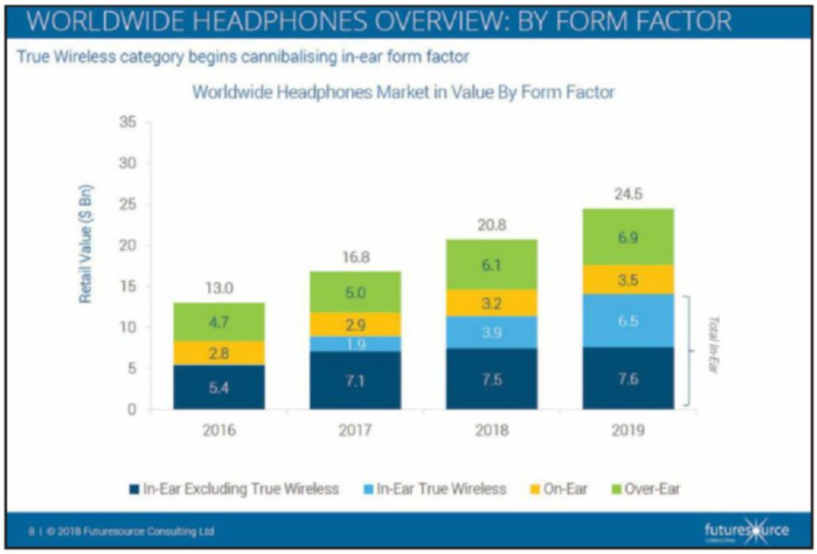
Application Examples



According to Futuresource, shipments of true wireless headphones doubled in volume year-on-year. Although Jabra and Bose have been nibbling away at Apple's lead, AirPods remain the true wireless giant and continue to dominate. Also, one in every 10 sets of true wireless headphones sold in the quarter were marketed for sports, a move driven by a range of brands looking at ways to differentiate.

The Hearables Opportunity

Futuresource Consulting also announced in its "Worldwide Hearables Report" that "the growing appeal of aftermarket headphones is opening up the new 'hearables' sub-segment, catering for those with hearing loss and alleviating much of the expense and stigma attached to hearing aids."



Consumers globally are willing to pay more to upgrade for better features, particularly wireless and more recently true wireless. In H1 2018, TWS accounted for 12% of wireless headphones and is set to capture more than a third of this segment by 2022. Revenues are set to reach $14.8 billion (up 44% year-over-year), accounting for 71% of global retail value. (Source: Futuresource Consulting)

Hearing aid manufacturers such as Widex are not sitting still. They are adding smartphone and television connectivity and more communication options, while making their products much more well-designed and attractive—even fashionable.

"Figures from the World Health Organization reveal 1.2 billion people have hearing loss and a further 375 million have disabling hearing," says Luke Pearce, a research analyst at Futuresource Consulting. "Hearing aids are typically expensive and many people feel embarrassed about wearing them. That's where the aftermarket headphones market makes its entrance. There's a lot of activity in the space, with consumers demanding more features, such as wireless and True Wireless, noise-cancelling technology and sports-oriented products. As brands explore ways to differentiate and compete, they're beginning to feel out the hearables opportunity."

As Futuresource explains, headphones are increasingly used for communication, smart home control through voice assistants, and a range of other always-in activities and brands are exploring products that can remain in the ears throughout the day. Furthermore, some noise cancellation features are loosely based on the same digital noise-reduction principles found in hearing aids. For headphone manufacturers, this presents a significant opportunity to capitalize on these synergies and launch into this now-attainable space.

Hearing aids are typically expensive and the stigma of wearing them has put many people off utilizing them to improve their hearing capabilities, cost being a particular concern in emerging markets where hearing loss is a growing problem. The traditional hearing aid and personal sound amplifier companies have been focusing on physical design improvements over the years and are cognizant to the needs of modern users in an ever-increasingly connected world that would like to sync their hearing aids with other devices, in particular smartphones to receive telephone calls or listen to music.

Equally traditional headphone vendors and other consumer electronics segments (e.g., the smart watch/wearables sectors) are increasingly focusing on fitness and wellness, a growing segment getting more attention from government and consumers alike.

Headphone vendors recognize the importance of ears in reliably determining health issues and at the same time understand how they can utilize their audio amplification technology and

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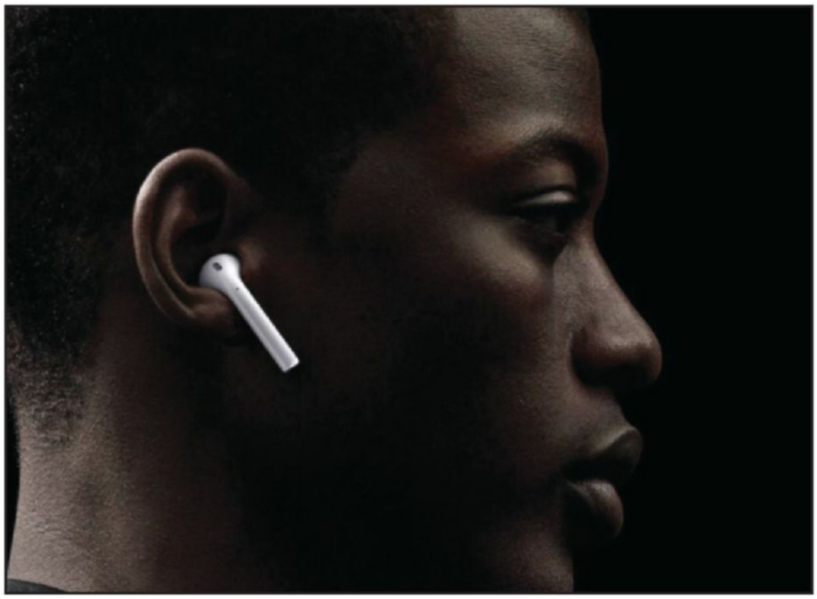

As always, Apple gets the fundamentals right, first. With the AirPods, Apple focused on what people needed from wireless earbuds and made it work 100%. They didn't overpromise, kept it simple, and delivered. The AirPods get consistently great reviews from every user. No wonder Apple commands the segment and owns 83% of this market.

developments and expertise in true wireless capability to address the wellness space and the growing demand for cost-effective hearing loss solutions.

With hearing aids still a highly regulated medical market in many parts of the world, developments such as the US Senate mandating that the US Food and Drug Administration (FDA) establish an "over-the-counter" hearing aid category for adults opens up the sector to many new entrants, including mainstream headphones companies.

"Hearing health is becoming a serious public health issue across the globe," says Pearce, "and with bodies such as the FDA making noises about deregulation, a range of solutions may become available to tackle this problem. The role of headphones makes logical sense, though there are still many complex and technological barriers for manufacturers to overcome before a well-rounded solution can prevail."

Offering improved audio quality when compared with Apple's Airpods, RHA Bluetooth 5 TrueConnect true wireless earbuds are an example of the audio industry's approach to an increasingly popular segment.

Strong Growth for True Wireless Stereo (TWS) Earbuds

According to the latest report from SAR Insight & Consulting—"FOCUS, True Wireless Earbuds: Market Growth, Market Shares, Technologies Used, Components Used"—more than 40 million true wireless stereo (TWS) earbuds with a retail value of more than $6 billion were shipped in 2018, and rapid growth is expected to continue.

TWS earbuds are driving innovation and creating a growth market. Total market revenues for the TWS earbud market are expected to reach almost $10 billion in 2019, outstripping all other headphone markets, according to SAR Insight's research.

As SAR Insight anticipates, the market is expected to rapidly grow over the next five years accounting for more than 60% of all wireless stereo headphones shipped by 2023. The biggest driver of this growth will be TWS's role as a companion product to smartphones, and the increasing trend of including voice-activated assistants.

"Apple has led the market since introducing the Airpods in 2016, which dominate market shipments today," Peter Cooney, Principal Analyst & Director at SAR Insight & Consulting, confirms. "Smartphone OEMs lead the market and accounted for an estimated 90% of volumes in 2018. These vendors are expected to continue to dominate, supplying TWS as companion devices to their smartphones. Total market revenues are expected to reach almost $10 billion in 2019, and smartphone vendors are expected to account for the vast majority of this."

"SAR expects that Wi-Fi and cellular technologies will start to be used in TWS, primarily to enable a direct connection to the cloud. However, there are still major issues such as efficiency and radiation that need to be overcome for these technologies to be implemented."

TWS Earbuds as Personal Sound Amplifiers

In October 2018, there was a very intriguing announcement from the US FDA allowing the "marketing of a new device, the Bose Hearing Aid, intended to amplify sounds for individuals 18 years

and older with perceived mild to moderate hearing impairment (hearing loss)." As *audioXpress* commented on the story (see Resources), this is the first hearing aid authorized for marketing by the FDA that enables users to fit, program, and control the hearing aid on their own, without assistance from a health care provider.

A few months later, we still don't know the exact device, what it does, or what the form factor is. However, we do know that the over-the-counter (OTC) hearing aid regulations should not be in effect in the US before August 2020. People quickly jumped to the conclusion that the FDA announcement referred to the existing Bose Hearphones, a neckband design, which the company itself states in its product page that "fall into a category of products called 'Personal Sound Amplification Products' (PSAPs)."

Whatever it might be, by approving this new product from Bose, the FDA is paving the way for this intermediate category of "personal sound amplification products" or PSAPs, with the personalization tuning features that users will require to use them as "self-fitting hearing aids," assuming they will not directly compete with medically prescribed devices.

Skullcandy recently announced its first truly wireless earbuds and opted for the affordable level ($129.99) in a distinctive design, still using a Bluetooth 4.2 platform. The earbuds offer a full day's worth of battery life, 12 hours—up to 6 hours within the earbud and up to 6 hours in the charging case.

In 2018, Australian company Nuheara received approval to deliver its IQbuds BOOST "smart personalized hearing" true wireless earbuds as hearing solutions to adults and children with mild to moderate hearing loss. The company even received a hearing aid supply contract (£138 million) from the UK's National Health Service (NHS), commencing in April 2019 and for an initial two-year period. This contract followed the announcement in March 2018 that Nuheara had received Australian government approval for supply of hearing devices, under the country's Hearing Services Program (HSP).

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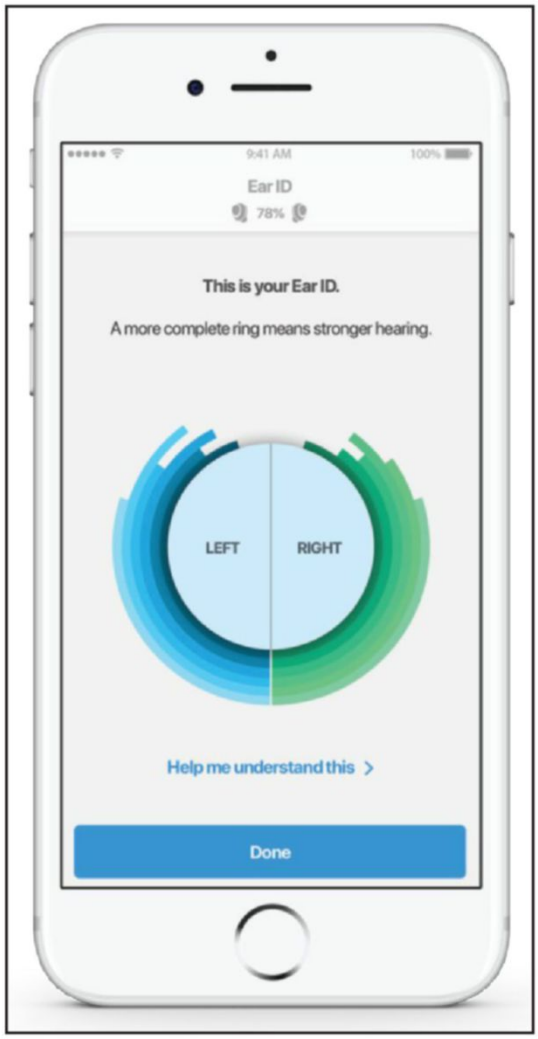
Promoted as an assistive listening device, IQbuds Boost leverage Nuheara's developments for the IQbuds, adding a clinically validated audiometric hearing assessment to calibrate the earbuds to the user's unique hearing profile.



The Libratone Q Adapt. In-ears designed with a Lightning connection supporting adjustable noise cancellation powered by the iOS device—and extremely affordable. Of course, Lightning designs could do so much more.

With an annual budget or more than 500 million Australian dollars, the project is one of the largest government-funded hearing health care programs in the world, providing eligible Australians with free and subsidized hearing devices and services.

The Nuheara IQbuds BOOST includes Ear ID, a clinical-grade hearing assessment that automatically calibrates the earbuds to each user's unique hearing profile. The technology calibrates the device to meet personalized hearing needs, using the globally renowned NAL-NL2 prescription procedure known for fitting expensive hearing aids.

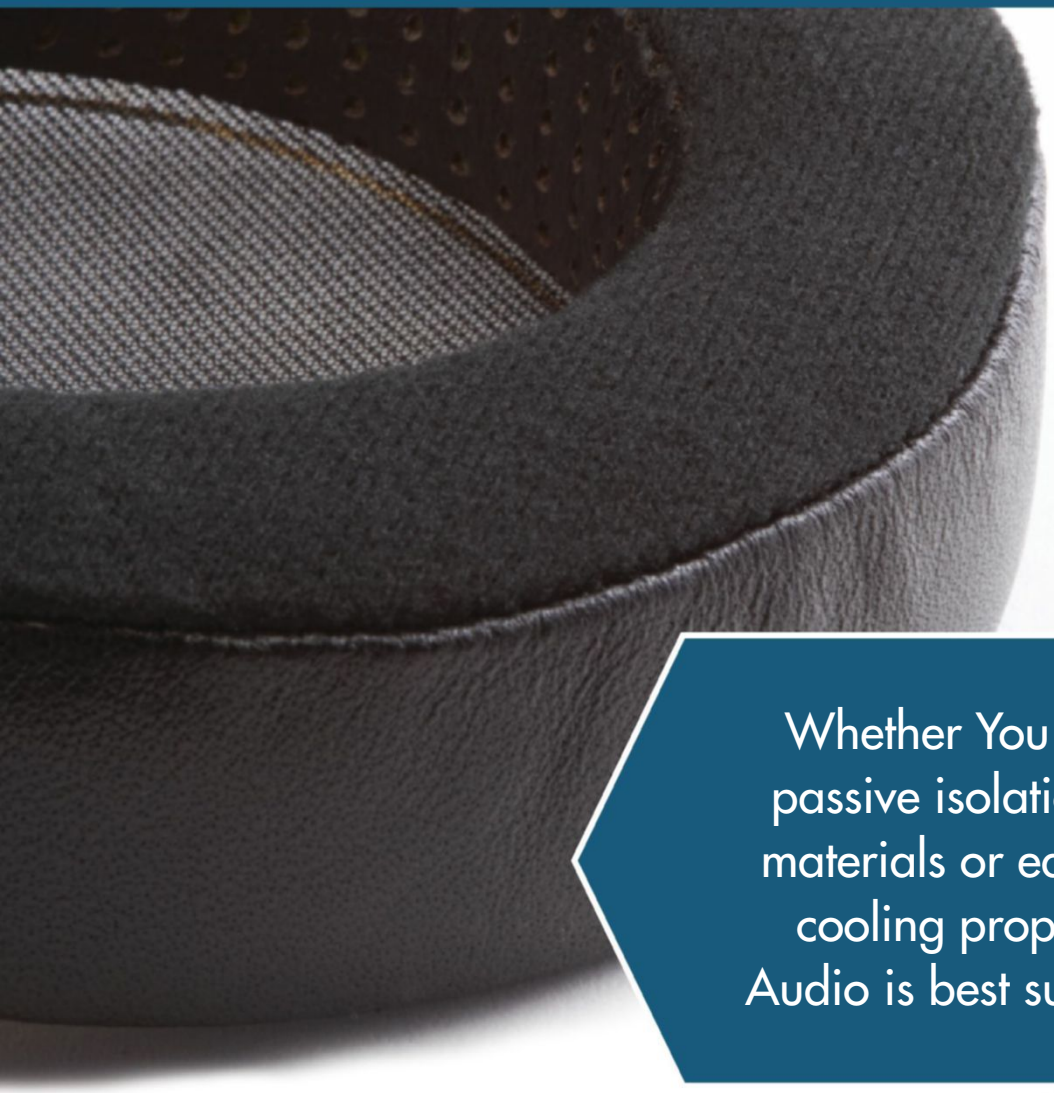
At the end of 2018, Nuheara was forced to raise funds to increase inventory levels of its IQbuds BOOST earbuds, selected by the NHS, and also to increase sales and marketing activities to launch new products including IQstream TV (an adapter for TV) and IQbuds MAX, a new advanced version featuring Active Noise Cancellation, which the company is developing in cooperation with Austrian company, ams AG. Already at CES 2019, Nuheara confirmed the launch of IQstore, "the world's first subscription-based hearing services app store designed to provide life-long hearing support solutions." The IQstore features two apps designed for tinnitus relief and AUTOCROS for those with unilateral hearing challenges, allowing users to choose supporting features to their IQbuds devices.

Worlds Collide

Anyone who closely follows the industry knows how technology developments on this front have been copious, with important innovations from all the leading hearing aid giants, such as Starkey, ReSound (GN Group), Signia (former Siemens), Widex, Oticon, and Phonak/Sonova adding completely new features (e.g., wireless smartphone connectivity and more sophisticated sensors). The consumer electronics industry has also amassed an impressive set of features that allow hearing measurements, preset-based frequency tuning, noise-cancelling, awareness amplification, and other personalization features. But new players in this area (e.g., Bragi, Nuheara, Earin, Jabra, Optoma, and others) also created completely new platforms that are now effectively very powerful embedded computers able to run machine learning and neural engines on the cloud and on the edge, and support personal voice assistants and voice interfaces.

All this, while still being used for high-quality music listening, with all the (now streamed) related connected services. When those systems start enabling practical text-to-speech conversion and

Superior Grade OEM Headphone Ear Cushions



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If manufacturers don't offer USB-C headphone options, the market will buy more adapters such as this one from Moshi, certified by Google for the Pixel 3 models.

translation, they can become an entirely new product category, as companies such as Bragi, Waverly Labs, Timekettle, Mymanu and others, including Google with its Pixel Buds, have been attempting.

Those two worlds are bound to collide, and all the companies know it and are working on strategic plans that leverage the best technologies from both sides of the equation. Also, the whole world is not going to wait for the OTC transition in the US. With Starkey being the only hearing aid company that's US based while most of the leading manufacturers are based in Europe (three in Denmark!) things will evolve in different ways globally.

In many European Union countries, hearing aids are heavily regulated, but they are also fully subsidized by public health schemes in most countries, or supported directly by health insurance companies, making the "medical approach" for OTC much more complicated. In those markets, the products will evolve purely targeting the generic consumer electronics approach, without explicitly advertising the medical applications (we can see that happening already with smart watches and sensor-equipped sports bands). That doesn't mean they couldn't become extremely popular very quickly if the benefits are considered effective—and that could lead to a reversal in regulation, incentivizing their acquisition and use. Very possible and likely.

Encouraged by the possibilities, mobile device accessory maker Belkin decided to enter the headphone business with new in-ear Rockstar wired earphones that have Lightning and USB-C connectors.

In other regions of the world—and the poorer countries—the self-fitting hearing aids will probably be welcomed as a competitive, affordable alternative to something people cannot typically afford. The volumes those countries can represent are simply massive, and it is the reason why companies in China are watching the space very closely. Also, for the big global consumer technology companies, this is far too attractive an opportunity, with far reaching implications on their expanding ecosystems of hardware, software, and services—we bet Apple will quickly jump at this.

The (Low) Power Problem

While there's no doubt that many of the previously mentioned developments are happening in truly wireless devices where data connectivity is present, and the form factor is miniaturized and battery-dependent, there is a middle ground that remains to be explored and presents a much larger opportunity for new headphone categories.

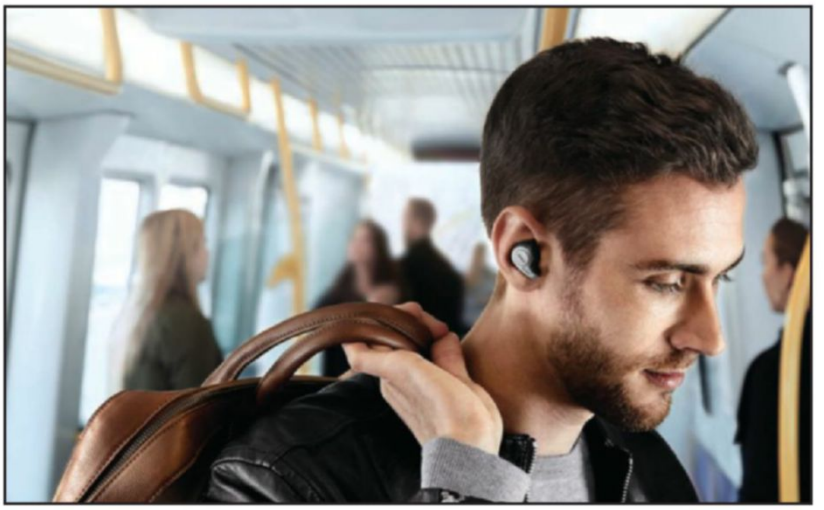
Consumers need to get familiarized with all the possibilities added to headphones, and the industry still needs to sort out how to combine all the features in devices with batteries that last a full working day (at least 8 hours) and still support an operational range from connected devices of at least 30 meters (98')—as offered by Bluetooth power Class 1 range. Neckband designs might seem part of the answer to those specifications, but now that consumers have experienced the freedom allowed by TWC, they are increasingly a tough sell.

While the TWS evolution process is ongoing, there's room for other form-factors and probably much more profitable design choices for generic feature-rich consumer headphones/earphones. An intermediate step is to combine a wired connection using Lightning and USB Type C connectors. While it's true that the removal of the headphone jack from smartphones was essentially the main motivation for headphone companies to invest in wireless, there is also increasing demand for new wireless features with noise-cancelling,

personalization, and voice assistants among the most requested. These are all features that pay a heavy toll on the duration of the battery, which is the reason why a digital connection also makes sense.

There's every reason for manufacturers to explore the potential of connecting to other devices via Lightning or USB-C interfaces, providing digital audio and power to the headphones, as well as externally supporting even more sophisticated personalization, noise-cancelling and voice interface features. Not only can they design better sounding and more lucrative models, but they can differentiate in features, without the limitations of built-in batteries. Those advantages are already present in a few models of Lightning-connected headphones such as the Audeze's SINE, Philips' Fidelio M2L, Bowers & Wilkins' P9 Signature, Libratone's Q Adapt, Pioneer's RayZ Plus, and Sennheiser's Ambeo Smart Headset, to mention just a few.

And it's not just for Apple iOS and Lightning. The same advantages can now be leveraged for USB-C. Android smartphone brands are adopting USB-C and removing the headphone jack, and yet, the headphone manufacturers are being slow to react. And as the requirement for USB-C connected solutions also increases, other companies will simply sell accessories. As an example, Moshi just announced a second-generation USB-C Digital Audio Adapter with Charging, certified by Google for the Pixel 3 models. Developed under Google's Made for Google partner program, users can listen to music while charging their Pixel 3 or



The Jabra Elite 65t can be considered state-of-the-art true wireless stereo (TWS) earbuds struggling to become hearables, even benefiting from the extensive experience in wireless audio and hearing aids from the GN Group—owner of Jabra and ReSound, among other brands.

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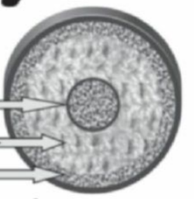

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Pixel 3 XL at the same time. The adapter retails for $40 and features an improved amplifier and digital-to-analog converter (DAC), which outputs high-resolution audio at 24-bit/96 kHz, and is able to drive even more demanding headphones. Now, there's an obvious convenience in having headphones with USB-C that offer the same and much more, without requiring an adapter.

Also, given the complexity of TWS designs, companies will increasingly rely on existing "fast-to-market" development platforms, making the designs essentially very similar and with limited features. In a market where Apple continues to reap the benefits with its AirPods, manufacturers are simply trying to throw in every argument they can in an effort to attract consumers.

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No wonder Amazon and other online retail platforms have been recently invaded by cheap TWS options advertising Bluetooth 5, fast-charging, graphene drivers, voice-controlled AI and more, obviously limited to less than 3 hours of use.

Of course, other manufacturers are trying to find a way to explore the possibilities of the latest SoCs to pick and choose the features they will highlight—even if in the end not all of them will be able to work as advertised. An example, using Qualcomm's latest chipset for premium wireless headphones was announced by Mavin, a California brand, and promoted on Indiegogo. Without much surprise, the Marvin Air-X true wireless earbuds, which were promising 10 hours with one stand-alone charge, and up to 50 hours with its included charging case, seemed too good to be true and didn't even reach one-third of its funding goal. It seems that, after multiple disappointments, people finally are now cautious with overpromises on crowdfunding platforms.

This doesn't mean that there aren't technology solutions on the way to solve the TWS dilemma of features versus battery capacity, as Qualcomm already has shown with its QCC3026 Bluetooth Audio System-on-Chip (see additional content).

And There's Hearables

On the consumer electronics side, we have truly wireless stereo earbuds, and on the other side of the equation, we have TWS earbuds expanding into Personal Sound Amplification Devices (PSADs) and hearing enhancers that will soon be available over the counter. We can also look at the promise of hearables, which are (or should be) a different category. How different?

For a very simple perspective, we can look at the example of an American company as ZVOX Audio (see additional content). The company is working in the field of intelligibility with sound for TV and soundbars, and decided to be among the first to grab the OTC opportunities. ZVOX Audio has designed an affordable solution for voice enhancement, targeting a very specific market of older people. It is a curious example of the diversity of approaches this segment will see in the near future.

The term "hearables" has so far been associated with the evolution of consumer-oriented in-ears toward something that could be health-related, not necessarily medical, but combining the benefits of biometric sensors, data connectivity and some kind of augmented hearing, as consultant Nick Hunn wrote in his document "The Market for Hearable Devices 2016-2020," published in November 2016.

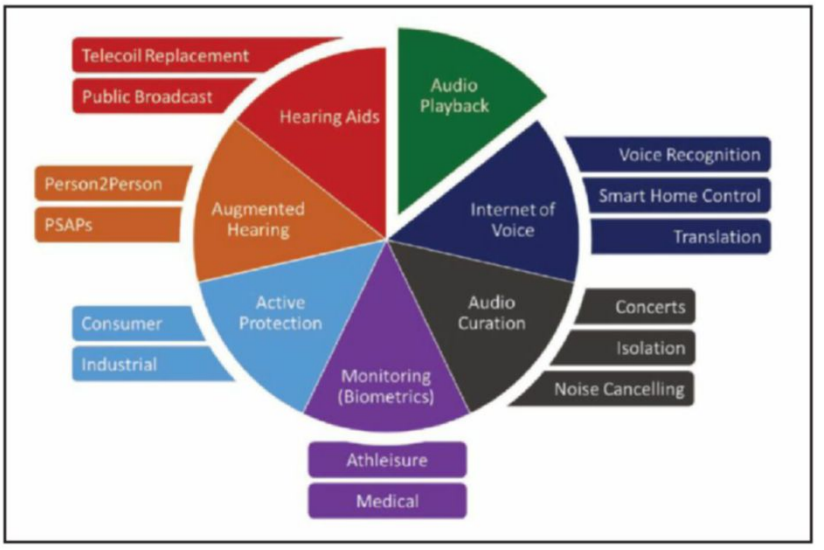
Not surprisingly, since Bragi successfully launched its Dash Pro "hearables," the company also decided to collaborate with the leading US audiology company Starkey, launching The Dash Pro tailored by Starkey Hearing Technologies. Bragi itself holds an important patent portfolio in this area, and the company makes no assumption that there will be confusion between the two markets, as Bragi's founder and CEO, Nikolaj Hviid stated in the interview he did with *audioXpress* at CES 2018 (see Resources).

In that interview, Hviid shared his wish that medically oriented hearing aid companies would enter the "hearing enhancement" space, proudly stating "Hearing aids are not a luxury item, because it is a tool for those people. They need it and it's very important to them. But there's this huge unsatisfied group of people that have an issue that they need to deal with. I even see many people that would buy our product, will eventually go and buy a hearing aid. Because they will realize that what we can do for them is OK, but that they actually need more. What we do will also help making hearing aids more accessible."

That's why it's important to address the market for hearables from its own intrinsic perspective and key selling arguments, outside of the "hearing enhancement" applications. As Hunn clarified in the mentioned 2016 report, a "hearable" entails much more than wireless connectivity, to include sound personalization and enhancement features, but also, audio curation, voice detection and processing, possibly access to translation services and what he summed up as "audio augmentation," as main drivers for market development and expansion of hearables.

Augmented Hearing was also the topic for a presentation given at the IBC 2018 show, with G-Audio Labs VP of Business Development Adam Levenson, titled "Augmented Reality Audio: The Next Generation of Hearables." In this session, Levenson presented his forward-looking glimpse into the future of Augmented Reality (AR) audio.

As a starting point, Levenson asked the audience the same question: What are Hearables? And he defined it as a product at the crossroads of headphone and hearing aid technologies and AR. In his presentation, he referenced Poppy Crum, Chief Scientist at Dolby, who said hearables are at the "convergence between entertainment, lifestyle, and hearing health," and David Cannington, the co-founder of Nuheara, who said, a hearable is a device to "control the elements of your physical environment," to "orchestrate your soundscape," and to provide an "on-the-fly personalized hearing experience."



The market segmentation for hearables according to analyst Nick Hunn (Image courtesy of WiFore Wireless Consulting, www.wifore.com)

Levenson goes on to detail that, apart from the basic functions of music streaming and input level adjustment from connected devices, hearables should support audio enhancement features, with personalization based on a hearing test, noise cancellation, and noise reduction. But more important, hearables should have "smart capabilities" and support speech amplification, listening directivity, translation, voice assistant, and biometric tracking—all things that we've seen in multiple products so far.

In his presentation, Levenson stated, "Imagine having the ability to adjust the mix of your daily listening experience with a technology that reduces ambient noise, amplifies and focuses speech, adjusts EQ to match your unique hearing profile, and interactively layers music with your voice assistant. This is AR Audio, and the latest

Adam Levenson, VP of Business Development, G-Audio Lab, believes that hearables hold a lot of potential for augmented reality (AR) and his company is looking at offering unique solutions from an engineering perspective. "I think the whole area of AR in audio is inevitable. How it's going to look, how this is going to become a consumer product is still very much unknown."

The result of a multi-year engineering effort, combining artificial intelligence (AI) and advancements in sensors and hearing technology, Starkey's new Livio AI platform combines tracking brain and body health and advanced environmental detection to create what the company calls "Hearing Reality Technology."

Hearables on the market are already tapping into this potential. But this is just the beginning. The future of AR Audio is Artificial Intelligence (AI), advanced DSP, and procedural sound."

In his opinion the "Augmented Reality Audio" approach, powered by machine learning (ML) algorithms, DSP, and binaural rendering, will enable new features like Situational Responsiveness, Focused Listening, Selective Hearing, and positioning of Virtual Objects. And this should be supported in automatic response to the "environment classification" and not something manually adjusted by the user.

In the next few years, we will see both consumer electronics and hearing aid companies exploring this hearables potential. As Bragi has shown with its Dash Pro, and Starkey Hearing Technologies has announced with its Livio AI platform, introduced in 2018, this will be both a multipurpose and completely-new augmented concept.

When Starkey's President Brandon Sawalich said, "we are not in the hearing aids business, we are in the communication business," that is a clear sign where things are heading. Their Livio AI platform explores all the embedded processing power and AI capabilities and goes as far as pitch new language translation possibilities. Starkey calls the concept "healthables" and talks about a new "hearing reality."

ZVOX VoiceBud VB20 Full-Featured Over-the-Counter Voice-Enhancing Hearing Device

ZVOX Audio, a Massachusetts-based audio manufacturer, announced the availability of its VoiceBud V20, the company's first personal audio device. As the company believes the category will be booming in the next 18 months as more companies enter the over-the-counter (OTC) personal sound amplification product business, ZVOX designed a very affordable device equipped with dual-microphone NoiseBlocker technology and smartphone app control using high-performance American-made electronics components.

The VoiceBud VB20 is designed for people who have trouble understanding voices in some circumstances, and features a combination of advanced technologies that currently cannot be found in devices selling for less than $600. The VB20 is being introduced with an MSRP of $299.99. Although priced similarly to single-microphone hearing devices without wireless app control, the VB20 is FDA registered, and offers an array of features normally found in much more expensive devices.

All electronics components, microphones, and transducer in the VB20 are made in the United States by two of the leading companies in hearing technology.

www.zvoxaudio.com

Innovative Interface Technology from ams Reduces Constraints of True Wireless Earbuds

Austrian sensor specialist ams introduced POW:COM, an innovative interface technology, which enables Power and Communication between a true wireless earbud and a charging cradle over a two-wire connection. A true wireless earbud previously required as many as six pins to provide both power transfer and communication between the charging cradle and the earbud, compromising the mechanical design of a small form factor that fits comfortably in the user's ear.

The ams POW:COM interface technology, for the first time, enables simultaneous data communication with 1kBit/s net data rate and over 150 mA charging current supported, far more than applications need today.

The POW:COM interface is implemented with the ams AS3442 host IC in the cradle, and the AS3447 client IC in each earbud. The sophisticated POW:COM protocol developed by ams provides for power transfer at 5 V, I^2C communication, interrupt signals and up to five GPIOs via a single transmission line between each pair of AS3442 and AS3447 devices.

The communications capability provided by the POW:COM interface supports an enhanced user experience of the charging cradle. For instance, the earbud can request the battery information from the charger cradle and display it on a mobile app. The POW:COM interface also allows the earbud manufacturer to implement other useful features, such as auto-recharge, automatic pairing of the earbuds with the user's phone on opening the charging cradle's case, and accessory firmware updating while charging.

The AS3442 and AS3447 POW:COM devices are available now in production volumes, and some manufacturers have already included the ams POW:COM devices in their earbuds.

An evaluation board for the POW:COM interface parts is available on ams online shop. For sample requests or for more technical information, visit www.ams.com/Power-Communication-Interface

Qualcomm QCC3026 Bluetooth Audio SOC

Launched in 2018, Qualcomm's QCC3026 is a flash-programmable Bluetooth Audio System-on-Chip (SoC) designed to reduce power consumption by up to 50% compared to previous generation entry-level flash devices. The new SoC is also engineered to help manufacturers simply and quickly develop and commercialize feature optimized truly wireless headsets, and extends the Company's portfolio of low power Bluetooth audio solutions following the announcement of the flagship QCC5100 Bluetooth SoC series earlier in 2018.

At its lower price point, the QCC3026 is able to deliver a compelling performance and low power consumption with the use of enhanced Qualcomm TrueWireless Stereo technology. The new SoC allows for a truly wire-free listening experience using enhanced Qualcomm TrueWireless Stereo protocols together with an improved radio that allows a robust overall wireless connectivity, from handset to earbud as well as between the earbuds themselves, important both when making calls and listening to music.

The solution enables an easy pairing experience and balanced power distribution between earbuds for lower overall power consumption and longer usage time, while supporting consistent, high-quality audio over Bluetooth with Qualcomm aptX audio technology. For digital assistant applications, support for voice services is available via button-press activation, relaying the audio stream and voice control

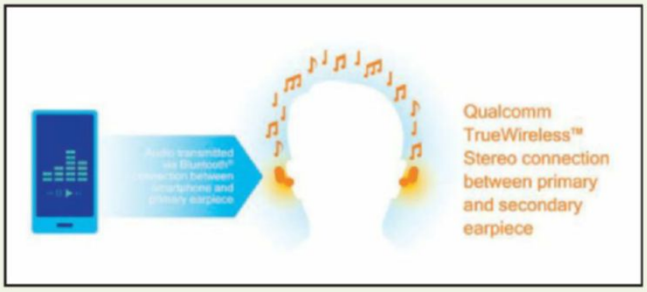
Qualcomm's QCC3026 low-power Bluetooth Audio SoC already supports Bluetooth 5 and features Qualcomm aptX audio technology and Qualcomm TrueWireless Stereo Plus, eliminating the need for cross-head Bluetooth transmission by simultaneously connecting the mobile device to both earbuds.

capabilities to a handset to process and execute commands.

The solution also allows for suppressed background noise and echo feedback with Qualcomm cVc Noise Cancellation Technology. Fully tested Qualcomm TrueWireless Stereo applications can be immediately deployed on the SoC with a wide range of features that can be readily configured or customized.

Reference designs based on the QCC3026 SoC are already available from several ODMs to help manufacturers commercialize entry-level and mid-tier wireless headsets quickly and cost-effectively.

www.qualcomm.com/products/qcc3026



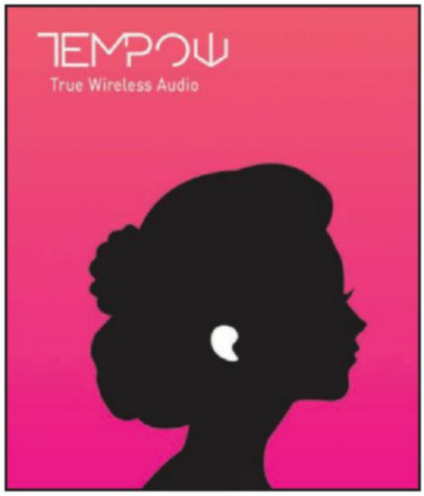
When paired with the Qualcomm Snapdragon 845 Mobile Platform for smartphones, Qualcomm TrueWireless Stereo Plus can help reduce power consumption by up to an additional 10%. It supports a more simple pairing experience when connecting earbuds to the mobile device and helps reduce latency.

A New Technology for True Wireless Earbuds Compatible with Any Bluetooth Chipset

CEVA, a leading licensor of signal processing for connected devices, along with French wireless audio pioneer Tempow, recently unveiled a collaboration to bring next-generation True Wireless Stereo (TWS) to the Bluetooth earbud mass market. The resulting joint IP solution delivers synchronized left/right audio over standard Bluetooth dual mode connected links with short latency and without the power consumption penalty associated with many of today's TWS implementations.

CEVA's Bluetooth Dual Mode IP was enhanced to work seamlessly with Tempow's patented True Wireless Earbuds technology, eliminating the need for power-hungry inter-earbud wireless links.

Tempow is a software research and design company focused on pushing the limits of Bluetooth technology. Its first product, the Tempow Audio Profile (TAP), is an updated Bluetooth protocol allowing any Bluetooth chip to stream audio on multiple Bluetooth audio outputs simultaneously. It is a 100% software solution, and works with any brand of Bluetooth speaker on any chip. Tempow has also been working in specific software solutions designed to expand Bluetooth capabilities for home theaters and smart home assistants—TAP TV and TAP Smart, respectively.



But Tempow's hardware-agnostic solution also lets any smartphone manufacturer create wireless earbuds with near-zero latency, increased battery, and high fidelity, in alternative to proprietary or non-standard Bluetooth solutions. The company has established global partnerships to bring its Tempow-enhanced Bluetooth solution for True Wireless earbuds, named Tempow True Wireless (TTW).

As Tempow details, other than TTW, there are only two ways to power wireless earbuds with Bluetooth. The first is to create a chained connection between the sound source, the first earbud and the second earbud, which requires either a non-standard Bluetooth link (e.g., Qualcomm True Wireless Stereo) or a Near-Field Magnetic Induction (NFMI) link. The second is Apple's solution implemented in the popular AirPods, which connects the first earbud to the sound source through a classic Bluetooth audio link (A2DP) and then requires the second earbud to sniff the first data link. This is only compatible with its Airpods, which inherently limits product diversity in the consumer marketplace.

The TWS Challenge

Audio streaming via Advanced Audio Distribution Profile (A2DP) is a proven strength of Bluetooth, but with the advent of wireless earbuds, new challenges have arisen. With traditional Bluetooth stereo headphones, the headphones are seen as a single end-point.

With earbuds, however, the left/right sink points are now physically separated and the Bluetooth A2DP paradigm is challenged to stream audio with the necessary degree of synchronization. To date, a number of different solutions have appeared to address this, such as audio-forwarding and eavesdropping, but these solutions often result in significant battery drain on the already power-restricted earbuds.

Tempow and CEVA are leading the

charge on a new, optimized approach, which eliminates the need for the inter-earbud secondary wireless link and works with any Bluetooth AD2P-enabled phone on the market. Within the phone, a straight forward software update is needed to install the Tempow Audio Profile (TAP). Within the earbuds, CEVA and Tempow have collaborated to implement a joint solution, which delivers the essential high quality audio synchronization over standard Classic Bluetooth BR/EDR links.

True Wireless earbuds are currently one of the top product categories in the headphone market both in terms of volume and price, with several brands increasing their efforts to launch their own devices to compete with Apple's Airpods. "Several smartphone companies have tried to compete with Apple on true wireless earbuds—Samsung with its IconX, Huawei with its Freebuds, Google with its Pixel Buds, and countless others—but they've all fallen short on delivering an exceptional listening experience due to poor audio quality and stability," the company explains. "Until now, no other company has been able to replicate the Airpods' user experience because Apple has created a closed ecosystem with both hardware and software under the same roof—an inherent advantage over companies that focus exclusively on hardware and outsource software."

"There's absolutely no reason that only one company should be able to deliver an amazing wireless earbud experience," says Vincent Nallatamby, founder and CEO of Tempow. "In many ways, we are thankful that Apple took the first step in showing the world the possibilities of Bluetooth-enabled wireless earbuds, but we believe our solution can democratize

the experience and help other companies push the boundaries even further."

As the company confirms, the new Tempow True Wireless solution doesn't require any retransmission of data between earbuds and saves substantial bandwidth, permitting the use of high-fidelity codecs such as LDAC or HWA for improved sound quality, and reducing the audio-video latency for gaming and watching video on mobile devices. This, while allowing to extend battery life by up to 50%—one of the critical factors in the design of true wireless earbuds.

"TTW is completely hardware agnostic, works with any smartphone, with any chipset, and is not cost prohibitive for hardware manufacturers," the company adds.

www.tempow.com | www.ceva-dsp.com

Dialog Semiconductor Demonstrates Bluetooth Low Energy True Wireless Stereo Solution

Claiming an industry-first technology to address the challenges associated with power consumption in true wireless stereo (TWS) systems, Dialog Semiconductor introduced an audio over Bluetooth Low Energy (BLE) proof-of-concept running on its widely adopted SmartBond SoCs, the first implementation to solely make use of BLE. This technology, which Dialog unveiled at Bluetooth World 2018 in Santa Clara, CA, allows for best-in-class audio synchronization between left and right channels, and addresses the challenges associated with power consumption.

Dialog Semiconductor demonstrated a Stereo High Fidelity (HiFi) quality audio streaming over BLE technology, enabling an improved implementation of TWS audio streaming. Dialog's development runs on its widely adopted SmartBond SoC, using only BLE, and allowing for best-in-class audio synchronization between left and right channels. An updated implementation shown at CES 2019, already provided the ability to broadcast the same audio stream to multiple TWS headsets while delivering best-in-class audio latency and perfect synchronization between left and right channels on each headset.

As Sean McGrath, Senior VP & GM of Dialog Semiconductor's Connectivity Business Group explains, "This proof of concept demonstrates Bluetooth low energy's potential to deliver the solution developers need, while overcoming the limitations of existing technology that have traditionally held devices back from their true potential."

"By solving the issues of power consumption and synchronization, we are showing how Bluetooth low energy could address two of the biggest complaints from True Wireless audio users. Manufacturers would have the ability to deliver True Wireless Stereo audio products that can

simultaneously stream HiFi quality audio while delivering long lasting battery life for their products in the consumer and professional wireless audio market."

Bone Sensor Headset

At the recent CES 2019 show, Dialog Semiconductor also demonstrated a new True Wireless Stereo Headset, which utilizes Bone Sensor Voice Activity Detection to activate or deactivate wireless audio signals just by relying on the bone vibrations of the headset's wearer. By detecting when the wearer is speaking simply by picking up the sound that vibrates off their bones, the demo highlighted how efficiently it can filter out outside audio and create a high-quality, far more accurate noise-cancelling headset experience.

www.dialog-semiconductor.com

Dialog Semiconductor's demonstrations of Bluetooth 5 and TWS over Bluetooth low energy at Bluetooth World 2018 in Santa Clara, CA.

Cypress Bluetooth Improves Connections for Wireless Earbuds and Hearables

Stereo synchronization has proven to be a difficult challenge in Bluetooth earbuds, as even the slightest connection loss can result in a frustrating user experience. Cypress Semiconductor introduced a Bluetooth audio development solution based on the new Wireless Audio Stereo Synchronization (WASS) application and the CYW20721 Bluetooth and Bluetooth Low Energy (BLE) Audio microcontroller (MCU). The solution provides stereo synchronization and robust connections for differentiated performance in wireless earbuds.

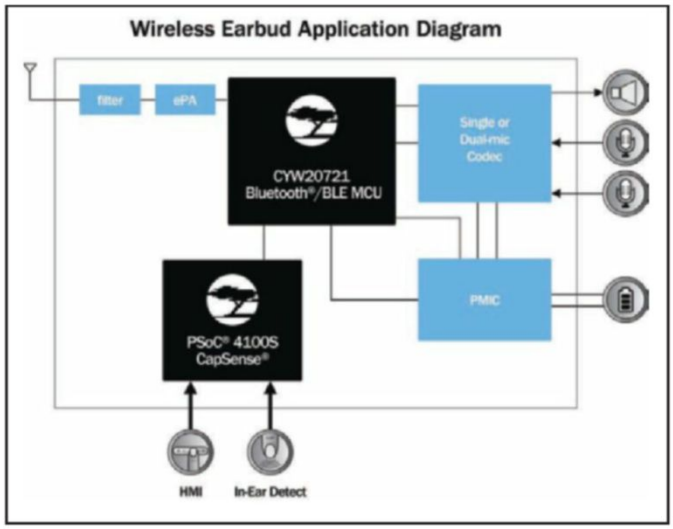
The MCU features a link budget that is 6 dB better than competing chips, resulting in up to twice the range. The extended range and the WASS application combine to enable superior cross-body performance, giving users uninterrupted audio on their earbuds with their smart device in their back-pocket or on their wrist. The MCU also operates at nearly 50% lower power consumption than existing chips, enabling twice the playback time or the use of smaller batteries in sleeker form-factors.

The novel WASS application is available in Cypress' WICED Pro software development kit (SDK), the latest version of its all-inclusive WICED SDK that provides developers with a turnkey solution to get products to market quickly.

The highly integrated CYW20721 chip and WASS software combine with multi-mode Bluetooth/BLE MCU to support essential additional consumer features such as voice commands and cloud-based voice services.

Cypress' WICED Pro development platform also supports an ecosystem that includes multiple audio codec partners, allowing for price and performance differentiation, including echo and noise cancellation. The Wireless Internet Connectivity for Embedded Devices (WICED) SDK enables cloud connectivity with libraries for cloud services such as Amazon Web Services, IBM Bluemix, Alibaba Cloud, and Microsoft Azure, along with services from private cloud partners. The WICED SDK also supports iCloud remote access for Wi-Fi-based accessories that support Apple HomeKit, which enables hub-independent platforms that connect directly to Siri voice control and the Apple Home app remotely. The WASS software further benefits from advanced wireless error correction algorithms in WICED Pro, offering high-fidelity connections.

www.cypress.com/wireless



Cypress Semiconductor Wireless Audio Stereo Synchronization (WASS) wireless earbud application diagram.

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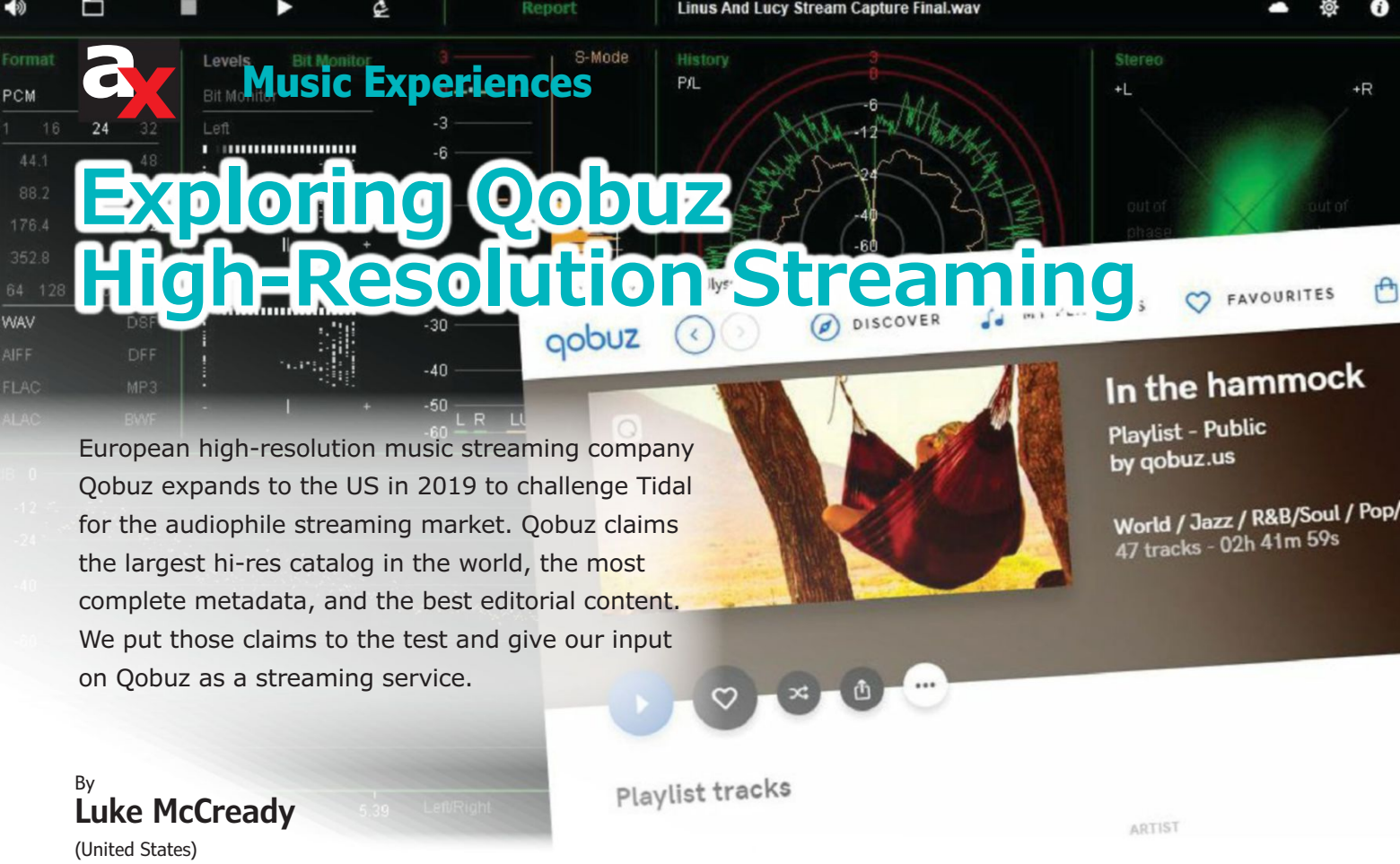
Exploring Qobuz High-Resolution Streaming



European high-resolution music streaming company Qobuz expands to the US in 2019 to challenge Tidal for the audiophile streaming market. Qobuz claims the largest hi-res catalog in the world, the most complete metadata, and the best editorial content. We put those claims to the test and give our input on Qobuz as a streaming service.

By
Luke McCready
(United States)

It's 2019, and I don't think I could ever give up music streaming. I don't miss purchasing each and every CD I listen to, or the countless hours I spent ripping them, or the never-ending mess that is organizing metadata. But I'll tell you what I do miss: the sound. When I stream music from a lossy streamer or YouTube, each song gets its point across. But it's not the full experience. That's fine for listening while cooking or driving, but not for system calibration, and not for an old-fashion listening session. This is where hi-res streaming enters the conversation. Now you don't have to choose: You can have the convenience of streaming, as well the quality of lossless CD audio and higher, using one of two services.

High-Resolution Streaming

You're probably familiar with Tidal: the streaming company that made a splash in the audio community in 2014 when it brought lossless CD-quality streaming to the US, UK, and Canada. Enthusiasts were again pleased when Tidal announced in 2017 that it would begin streaming hi-res audio. But there's another contender that you might not have heard much about: Qobuz. The Paris, France-based company aims to outclass all the other streaming services on three fronts: sound quality, documentation, and editorial content.

I recently chatted with Dan Mackta, Managing Director of Qobuz US. On the company's history, Mackta said, "Qobuz was started in France in 2007 as a hi-res download store by classical music fanatics. They felt that serious music fans were not being addressed in the online music space." And in the intervening years, "it evolved into a full-service, all-genre streaming service."

Though it was founded a full seven years before Tidal, Qobuz may not have reached your ears because it was initially focused on Europe. But that changes in 2019 with an expansion to North American shores. Some consider this a dangerous move: German company HighResAudio said that it considered expanding to the US, but opted not to because of music licensing fees. "None of the exciting streaming service[s] are profitable … it's a huge loss. So, why would anyone try to enter the US market, especially operating it from Europe?" HighResAudio is not wrong. Spotify has never had a profitable year since it was founded in 2006. And Tidal, the only hi-res streaming service in the US until Qobuz's 2019 launch, is in a pattern of losing tens of millions each year.

I spoke at length about this with David Solomon, the Chief Hi-Res Music Evangelist of Qobuz. Solomon said, "Our big plan is we're going to augment our profitability with downloads promoting the audio industry, which will promote us back, and then promoting artists who will promote back. [We're] doing a whole bunch of stuff now that's never been done before." It should be noted that the US has never had a hi-res streaming service primarily marketed to audiophiles. And Qobuz has plans to outdo Tidal. When I asked Mackta about what makes Qobuz different, he reiterated the three advertised strengths of Qobuz: "The largest available library of streaming hi-res music, focus on quality of metadata, and context."

The Qobuz Difference

Let's briefly explore those three areas of improved service. For sound quality, Qobuz believes in delivering music to your ears in the highest resolution possible. For much music, that is CD quality at 44.1 kHz/16-bit. Beyond CD quality, you can find 24-bit music in its catalog, including higher sample rates of 96 kHz and 192 kHz, as heard in the studio. Qobuz emphasizes discovering and cleaning up metadata, and filling in the story of how a piece of music came to be and which people made it happen. Additionally, Qobuz includes digital booklets for millions of albums in its catalog, showcasing lyrics and additional art, as carefully created by the artist. And for editorial content, the staff at Qobuz puts together articles dubbed Panoramas as an exploration into artists fundamental to shaping their genres (see **Figure 1**).

The first Panorama was published in late 2016, exploring the music and cultural significance of The Beach Boys' *Pet Sounds*. Since then, new Qobuz Panoramas have appeared almost weekly, delving into the history of legendary conductors, the fusion of afrobeat, the shaping of 90's British shoegaze rock, and the telling of film compositions in the making. Panoramas generally include music videos and mini-documentaries, and a handpicked list of songs that demonstrate the styles and forms laid out in the article. As compelling as these benefits seem, Qobuz is certainly in a crowded marketplace. Let's take a look at the other players.

The Competition

The typical streaming services are difficult to differentiate: Spotify, Google Music, and Apple Music all offer individual plans for playing unlimited, ad-free music for $9.99/month and family accounts for $14.99/month. Streamed music is typically 320 kbps MP3, considered high quality for lossy streaming. These services each have catalogs with more than 40 million songs, each offer algorithm-generated radio stations off of artists and playlists, and each offer downloads for offline listening. Deezer offers similar features and solo/family plans with the addition of Deezer HiFi, its $19.99/month service to stream full CD-quality music in a lossless format. Qobuz matches Deezer's service tiers and pricing, and then some, with Qobuz Premium providing 320 kbps MP3 for $9.99/month, Qobuz Hi-Fi offering 44.1 kHz/16-bit FLAC streaming for $19.99/month, and Qobuz Studio unlocking hi-res FLAC streaming up to 192 kHz/24-bit for $24.99/month. Each tier can be purchased annually at the cost of 10 months instead of 12. And the final tier, titled Qobuz Sublime+ at $299.99/year, includes the 192 kHz streaming of the Studio tier as well as discounts in its download store, reportedly 40% to 60% off. And Qobuz streaming is available via its web player and dedicated apps for Windows, macOS, iOS, Android, and Kindle. For those in the US, the only option for hi-res streaming above 44.1 kHz has been Tidal, stepping up from its lossy 320 kbps AAC streaming at $9.99/month to Tidal HiFi at $19.99/month. This service tier includes music at higher sample rates and bit depths delivered via an MQA stream.

Opting Out of MQA

MQA is short for Master Quality Authenticated, a sort of assurance to the listener that the content being delivered is the highest quality available. MQA is able to "fold" higher sample rate audio, significantly cutting down on file size. The result is a file that can be unfolded to a higher-res file by the right software and hardware. MQA is technically lossless compression that some experts believe comes with audible consequences, and many in the audio industry are hesitant to opt-in, viewing the licensing of MQA as giving too much power to one company, causing the cost of all music and audio equipment to increase.

For better or worse, Qobuz does not rely on MQA. I probed on this point: a man of few words, Mackta simply stated, "If customers demand it, we will consider it." Solomon added, "I have zero issues with MQA. I think it sounds good. [But FLAC] is just the way it's always been done. It's easy. You don't have to go through any steps to unfold this or fold that."

Forgoing MQA, the Qobuz approach to hi-res is about transparency and quantity. With Tidal, each hi-res album is merely branded "Master," and whether Master in each particular case means a purist 192 kHz/24-bit or a paltry 44.1 kHz/24-bit, there's no way to be sure. Qobuz, on the other hand, marks the sample rate and bit depth of every

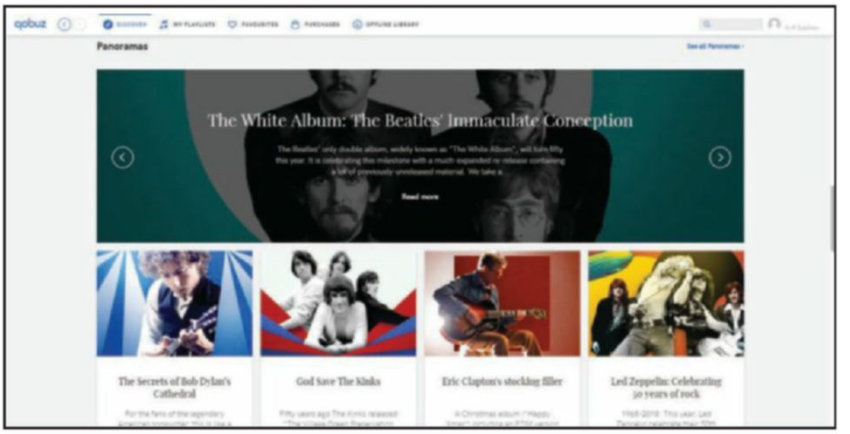


Figure 1: Qobuz Panoramas delve deep into the making of influential music.

album. As for quantity, Qobuz already has double Tidal's number of hi-res tracks, and that number increases almost daily. Qobuz also makes it easier to stream hi-res music. You can stream 192 kHz through the Qobuz desktop app, the mobile app, and even the web player with no additional hardware or software required. Meanwhile, Tidal restricts its streams to 96 kHz using the desktop and Android apps, with a 44.1 kHz limit for iOS and the Tidal web player. The only way to get 192 kHz out of Tidal is through the use of an MQA-capable DAC.

The Bandwidth of High Resolution

You pay heavily for the convenience of Qobuz with bandwidth, however. Streaming an album at 192 kHz uses about 42 megabytes of bandwidth per minute of audio. And halving the sample rate to 96 kHz still requires 23 megabytes per minute.

Qobuz Data Tier Bandwidth per Minute:
192 kHz/24-bit—FLAC—41.9 MB |
96 kHz/24-bit—FLAC —23.4 MB |
44.1 kHz/16-bit—FLAC —8.0 MB |
320 Kbps—MP3—2.5 MB
Total Bandwidth per minute by Quality Tier:
Master—MQA—12.9 MB | 44.1 kHz FLAC 6.8 MB
320 KBPS—AAC—2.6 MB | 96 Kbps—AAC+—0.8 MB

The numbers are intimidating. Streaming 192 kHz audio on 4G LTE, I'd chew through my monthly 2 gigabyte data bucket in less than an hour. And with daily hi-res streaming added into a month's worth of downloads, cloud backups, and a whole household's video streaming: data overage charges could become a regular occurrence even at home. But Qobuz is looking to the future: "Space is getting cheaper and bandwidth is just absolutely multiplying," Solomon said. "Bandwidth is wide enough that nobody has any problems streaming stuff. 5G is coming in about 10 minutes. A gig up, a gig down, that's gonna be the basic plan that's selling for $30/month."

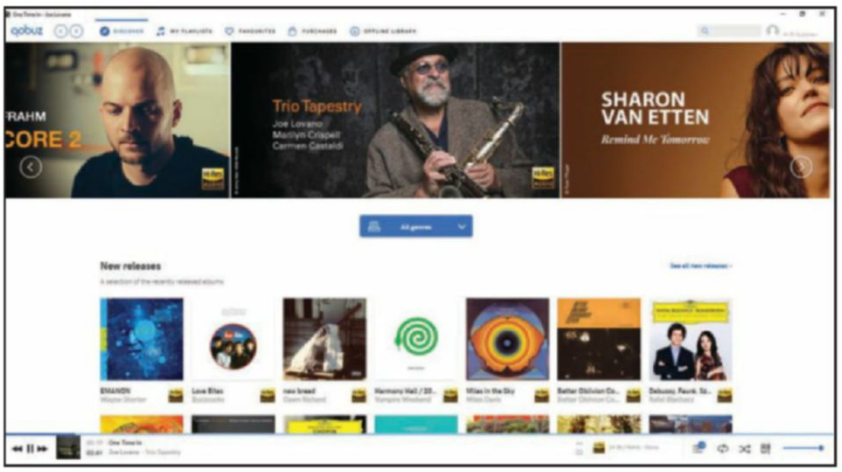


Figure 2: The Discover tab is the Qobuz hub to finding new music.

Judging Streaming Services

We've looked at sample rates and bandwidth, but how does Qobuz sound? In a word, excellent. Listening on my powerful home system in a treated room, I'm very pleased with the sound. To be fair, Tidal also sounds excellent. Can one differentiate between sample rates? I believe yes, and it's easiest when looking for specific cues in the audio. But these are small differences for the enthusiast, not the casual listener.

However, there's more to like or dislike about a streaming service than the raw quality of its files.App functionality and ease of use are important.Music discovery plays a big role to many. And we haven't yet explored Qobuz's claim to improved metadata. I would traditionally include depth of catalog in any streaming service review, but the good folks at Qobuz thoroughly stressed that Qobuz US was still lacking a significant portion of its catalog in the beta version I reviewed. Eric Benoit, the Marketing and Operations Manager at Qobuz US, assured me that many missing albums were still being populated. Qobuz in Europe offers more than 40 million songs in its catalog, and I would expect nothing less from Qobuz US now that the beta status has officially been lifted.

The Desktop Interface

The Qobuz interface for desktop and web player is laid out in four tabs: Discover, My Playlists, Favorites, and Purchases. Discover consists of a genre filter, music newly added to Qobuz, curated playlists, and Panoramas (see **Figure 2**). My Playlists functions as you'd expect. And Favorites houses all the music for which you've clicked the heart icon.

The Qobuz desktop app adds a fifth tab for Offline Library. Any music throughout the Qobuz catalog can be downloaded, a convenient feature for those traveling. While expected for mobile, Qobuz is unique in offering downloads on the desktop. It's worth noting a few Frenchisms that remain in the Qobuz interface at the time of review. For example, Qobuz metadata lists the "Composeur" of each album, and the quality of a current stream may be listed as "16-Bit/44,1 kHz - Stéréo". Also, the application measures used storage in Mo and Go, not MB and GB.

The Mobile Interface

The mobile app has a lovely simplicity to it. The bottom always-visible menu has four tabs for Discovery, My Qobuz, Offline Library, and Search. Discovery is laid out just as on the desktop app, and My Qobuz contains subpages for My Playlists,

Favorites, Purchases, and the app settings. And the Offline Library tab has a download animation while downloads are active, as well as a number indicating how many remaining tracks are queued to be downloaded. I especially like the interface for showing what's currently playing: a tab called Listening with album cover art, file quality, playback controls, and track metadata. A quick swipe to the left shows all tracks queued to play, and another swipe left shows a history of tracks played regardless of play queue (see **Figure 3**).

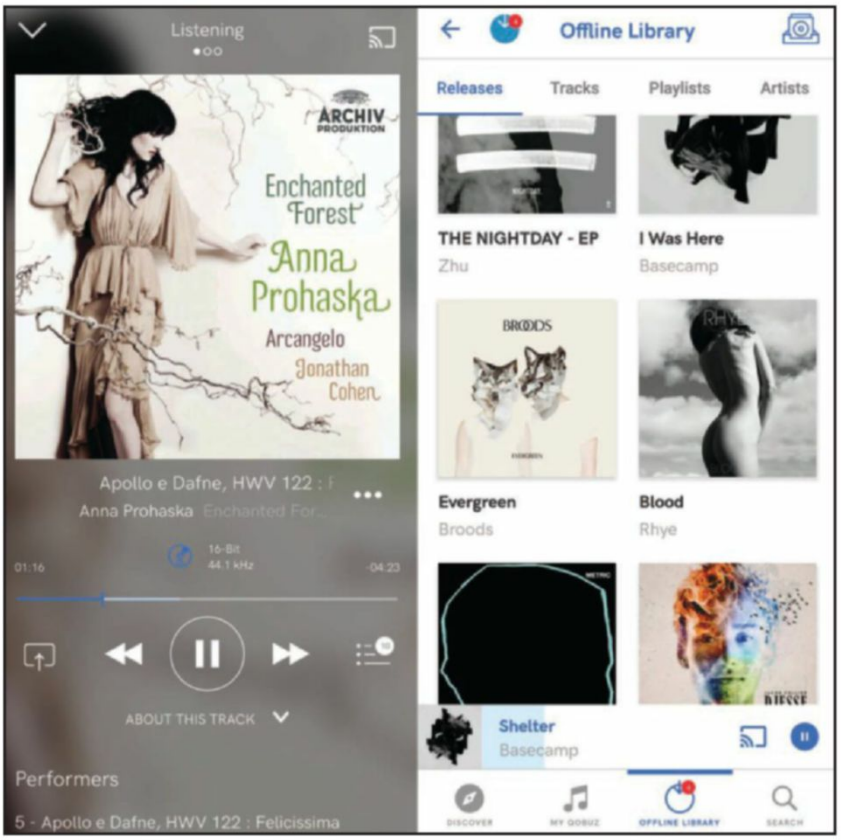


Figure 3: To the left, the Listening tab. To the right, the download library.

The Nitty Gritty

Playback quality is governed by the maximum you choose (see **Figure 4**). On mobile, you have the ability to set separate quality caps for streaming on Wi-Fi and mobile networks.

In the desktop app settings, you have the option to take exclusive control of the selected audio device. Using exclusive control, I watched my DAC display its automatic switch to different sample rates as I played different tracks, barring Windows Core Audio from resampling my Qobuz stream. The mobile app does not cache audio more than a handful of seconds in advance, but the desktop app caches the entire

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track at once. The caching is a nice feature because it visibly builds your offline library if you check an option in the app's settings, adding all cached tracks to the offline library.

You'd be right to be concerned about Qobuz monopolizing your storage space as it automatically caches all music played. Under music playback settings, you can set the maximum cache size from

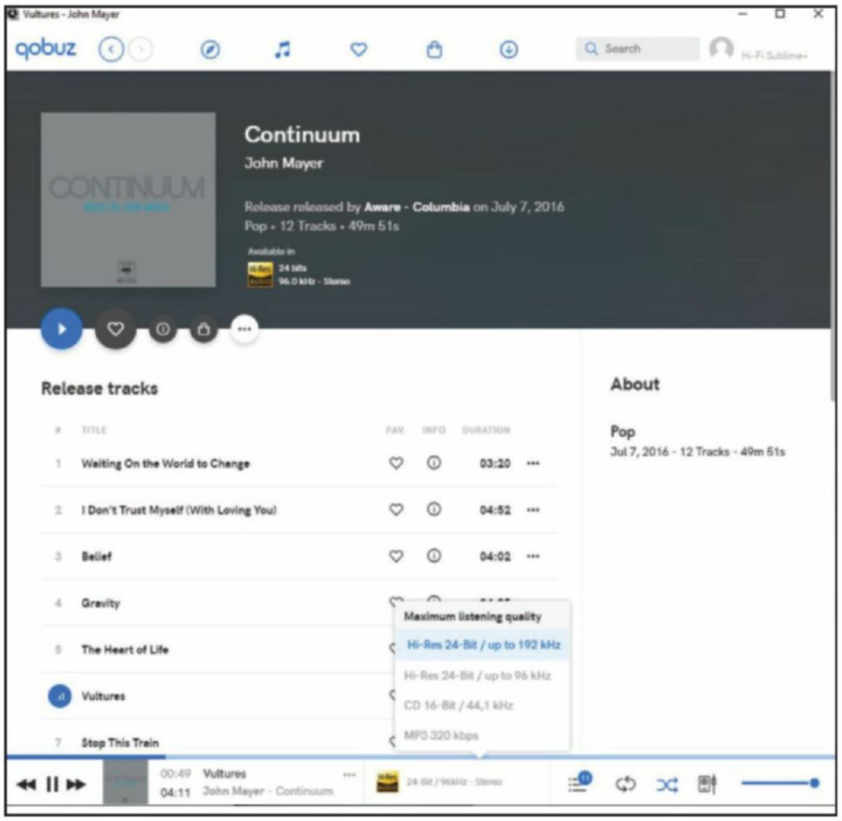

Figure 4: Use Qobuz quality tiers to set your maximum quality; see your current playback quality at all times.

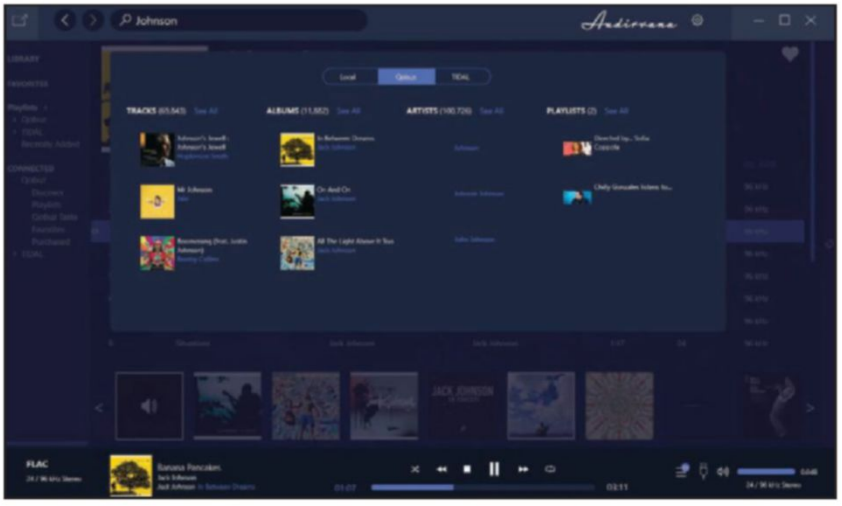

Figure 5: The search function in Audirvana Plus has separate tabs for Qobuz, Tidal, and locally found results.

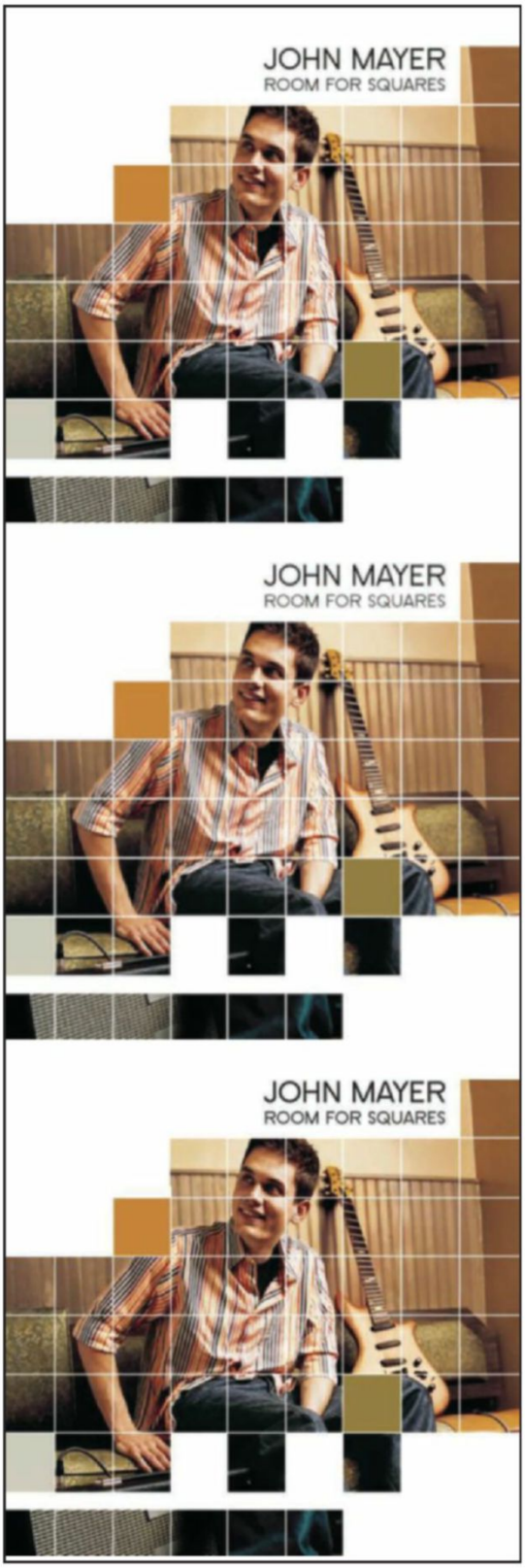

Figure 6: From top to bottom, album art in Qobuz, Google Music, and Tidal, scaled to the same size. In this comparison, Qobuz has a touch more detail than Google Music, and Tidal suffers the most from visible compression.

500 MB to 100 GB. As a bonus to those with multiple hard drives, Qobuz allows you to set a new location for cached music to reside. The music download location can also be changed in the settings, and you can monitor how much space it is using. Qobuz also supports linking to Facebook and Last FM, in case you want the world to know which music and Qobuz articles have been occupying your time. Qobuz lists keyboard shortcuts to better navigate the interface, control playback, and adjust streaming quality on the fly. And there's even the option to route your stream through a proxy.

Third-Party Support

In addition to its native software apps, Qobuz has partnered with a number of other companies to make its streaming available to more devices in more homes. By my count, 39 different hardware manufacturers from consumer to hi-fi support Qobuz hi-res streaming. Also, Qobuz supports software streaming through specialized playback programs, including Amarra Luxe, Audirvana Plus, and Roon. I asked Solomon about the benefit of these programs, and he said, "You can make almost anything play bit-perfect if you really know what you're doing, if you disable all the Windows crap that happens in between the signal and your DAC; all the Mac crap that happens in between the signal and your DAC. All of that is removed with the use of [these] specialized programs."

I explored Qobuz through Audirvana Plus with good results. Hi-res albums are marked in the interface, and I found album reviews by Qobuz staff and even my favorite albums marked in Audirvana (see **Figure 5**). However, Qobuz's extra metadata was not visible.

I also explored Qobuz through Roon. Roon has a lot of unique features it adds beyond acting as a portal to Qobuz: It has a unified search for music that shows both Qobuz and Tidal results together, and songs from both services can exist in the same playlist. Additionally, it acts as a music hub for your other stereos and devices. At $119 annually, Roon adds a lot of functionality to Qobuz that might be considered irrelevant to some and mandatory for others. Qobuz streamed just fine to my Chromecast Audio up to 96 kHz. In addition to Google Cast, Qobuz supports MusicCast, AirPlay, and Bluetooth.

Usability

Both the desktop and mobile Qobuz apps have excellent ease of use. Features are laid out intelligently, and extended metadata is easy to access. I compared latency streaming 320 kbps as an even benchmark across services. Loading a new track in Qobuz takes on average 1¼ to 1½ seconds. This feels noticeably latent compared to other services, with Tidal and Google Music consistently loading new tracks in ¼ to ½ second. The latency is also felt in the interface, where it typically takes three full seconds to load an artist page. Mackta said, "Right now, AWS (Amazon Web Services) hosts and serves Qobuz to consumers around the world." Whether the fault lies in hosting or optimization, at least for

About the Author

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now, Qobuz feels more latent than Tidal and Google Music. The search functionality in Qobuz is generally robust, though there are a few hiccups. When searching for "Clapton," Qobuz first recommends the artist *Clapton* with 0 albums and a description matching Eric Clapton. The second result is for the artist *Eric Clapton* with 172 albums and a different description of the same man. Tidal and Google Music don't have this confusion. The Qobuz search is intelligent enough to show the artist *P!nk* when searching for "pink," listed before *Pink Floyd*, yet after the artist *White Noise Meditation*, *Pink Noise*, *Zen Meditation,* and *Natural White Noise* and *New Age Deep Massage*. Neither Tidal nor Google Music get hung up on the meditation "artist."

Art and Metadata

In Qobuz, album art is always 600 × 600 pixels. This feels small compared to Tidal's 1280 × 1280 pixel album art, but fine in comparison to Google Music at 512 × 512 pixels. Depending on the album, Tidal has the obvious advantage of resolution. But for other albums, Qobuz or even Google Music appear sharpest. After all, the largest stored image means nothing if it was scaled up from the smallest source. Overall, there is no clear winner for album art quality (see **Figure 6**).

Qobuz and Tidal both have pretty deep liner notes for each and every song. However, Tidal also includes liner notes per album, which Qobuz doesn't. Further, Tidal includes links to featured musicians' artist pages, making it easy to tell when a guest horn player on a song is also an artist in his own right with albums of music to explore. Tidal and Qobuz sometimes have a review next to an album, sometimes not. The presence of a review on both can be hit or miss, even on major releases. Both Qobuz and Tidal can be subject to the same minor metadata mistakes, perhaps missing one of several backup singers in a song. Both Tidal and Qobuz seem beholden to the same sometimes-obscured information (e.g., guitarist vs. participant). Google Music's approach, by comparison, offers generally more complete and accurate information by relying on the crowd-sourced input of *Wikipedia*, visible with a single additional click.

1. Sébastien de Brossard, Dictionnaire de musique.
2. Denis Diderot et Jean le Rond d'Alembert, Encyclopédie ou dictionnaire raisonné des sciences et des métiers, tome 2, p. 622.
3. *Grove Dictionary of Music and Musicians*, « Cantata, 5. Decline of the Cantata and Cultivation of the Cantatille. »
4. *Mercure de France*, juin 1747, tome 2, p. 137.
5. Louis-Nicolas Clérambault, *Préface du 2e livre de Cantates françaises* (1713).

10

EVA ZAÏCIK MEZZO SOPRANO

« . . . présence sur scène et maîtrise de la voix, timbre mordoré, aisance dans les passages rapides et diction remarquable, mêlant des aigus assurés et des médiums très expressifs à un phrasé sans accroc. » Olyrix, novembre 2017

« Grande noblesse, Timbre somptueux, style souverain, de la Mezzo-soprano Eva Zaïcik » le progrès. fr, octobre 2017

Figure 7: A digital booklet for the album *Venez chère ombre* by Eva Zaïcik with a photo of the recording.

Qobuz seems subject to additional metadata mistakes, however. For example, though Qobuz has deep liner notes for Evanescence's album *Synthesis Live*, the name for each song is followed by "(Live) (Live)". Google Music and Tidal don't share this redundancy. Similarly, for the new album "boygenius," Qobuz lists each track with "[HD]" after the name, though the music is standard CD quality. Again, Tidal and Google Music display the song names without the tag. And artist Sara K., though listed properly in Google Music and Tidal, shows as K. Sara in Qobuz. In my testing, I didn't notice any examples of Qobuz having more accurate song titles than the competition. Mackta of Qobuz shared, "The metadata is initially delivered along with the music and artwork by the record label or rightsholder. When errors are brought to our attention, we can manually make fixes. In addition, we are partnered with several third-parties to enhance the metadata and information we are able to provide."

As for digital booklets, Qobuz claims there are millions in its catalog. Yet, despite opening some 300 albums in Qobuz, less than 10 had digital booklets. When pressed on this topic by a user in a "Hello Qobuz!" Q&A session, Qobuz responded: "It has been 6 years now that we have fought, there is no other word for it, with publishers and distributors to ensure that [digital booklets] are delivered to us. But if many are still missing, this is not our fault. If you only knew the time that we spend chasing this up!" And that was back in 2015. Judging by its press release claims over time, Qobuz is having success in finding and adding booklets. But it comes across more as a pleasant surprise for some albums than a feature across most albums (see **Figure 7**).

Overall, Qobuz comes up short in metadata, especially considering Tidal prides itself on exclusive releases and music videos, not metadata. But this trend may reverse given more time.

Music Discovery

In addition to the superb Panoramas I described earlier, Qobuz leverages its Ideal Discography and Qobuzissime awards to draw attention to significant historical albums and unique albums by emerging artists. Qobuz awarded albums are recognized in a special place, and most have original reviews by Qobuz staff. If you're after an education in musical history, Qobuz has a lot to share. However, there's also the dark side of artist discovery on Qobuz: There's no form of algorithm-generated radio whatsoever. And this is important because, for many listeners, algorithm-generated radio is the single best way to find new artists. Broad categories are fine for those merely after "classical" or

"blues," but it doesn't offer much for those after more specific or obscure genres. My personal listening tastes continue to evolve to more niche subgenres, guided by Google Music's superb radio algorithm. This just isn't possible in Qobuz.

Tidal offers limited algorithm-generated radio with mixed results, but Qobuz's complete lack of radio feels even more claustrophobic: Sure, the catalog is large, but it's lost to you if you don't know specifically what to search for. Fortunately, Qobuz has confirmed that algorithm-generated radio is "on the roadmap," but there's no telling of when it will be released or how effective it will be. Furthering the artist discovery problem, recommended artists in Qobuz are weak, and recommended albums are worse. Some of the suggestions couldn't be more wrong, tying Fleetwood Mac and a choral album as recommended listening to the electro-pop artist Broods. And when listening to an album by pop star P!nk, my first two suggestions are for a German folk/orchestral cross-over album and a jazz crooner singing Christmas songs.

Beyond custom radio, Google Music has well over a thousand pre-made stations spanning diverse subgenres, allowing you to dial into specific styles even if you've never heard of them before, and each playlist feels bottomless. By comparison, Qobuz includes a few curated playlists, but the playlists are short, sound scattered, and seem entirely dependent on a listener's tastes. I'm left with the distinct impression that Qobuz is not currently fit for assisting in artist discovery. It hasn't bridged me to a single new artist that I'm interested in exploring (see **Figure 8**).

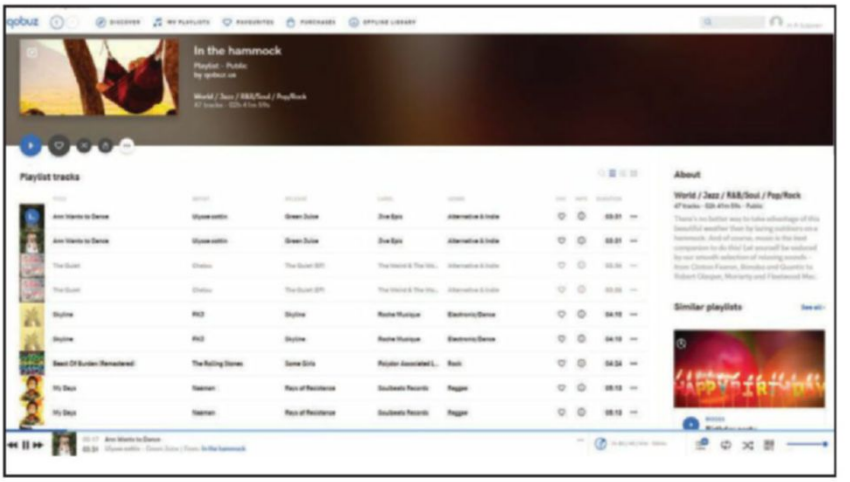

Figure 8: A Qobuz curated playlist. Discounting the curiously doubled tracks, there are only 23 unique songs.

Analysis of Files

Qobuz assured me that the quality of its stream is identical to the quality of its purchased downloads. I put that to the test with a 192 kHz file from the Qobuz download store and the same file streamed and captured at 192 kHz. MusicScope by Xivero, an

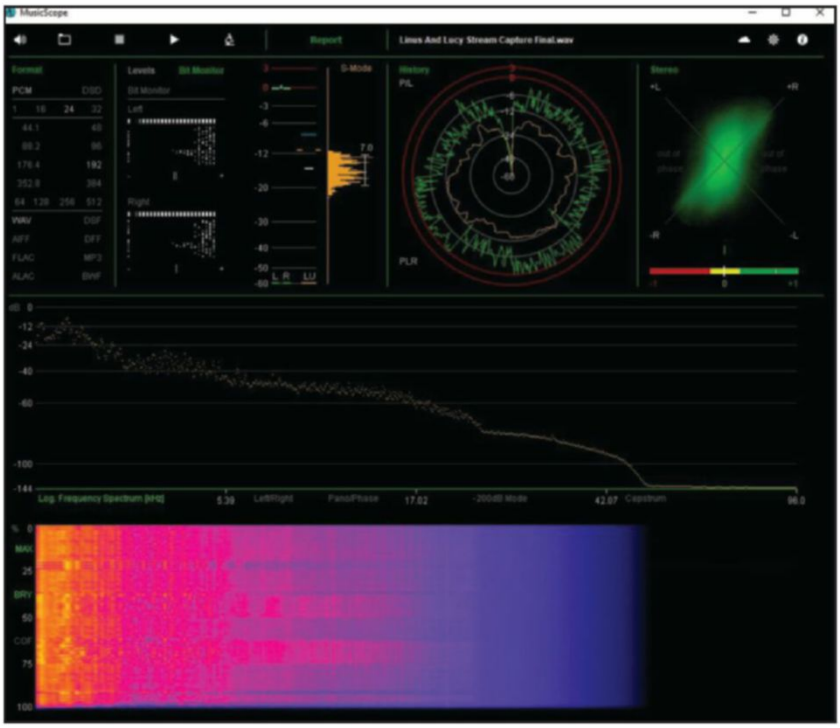

Figure 9: MusicScope analyzing a 192 kHz stream captured from the Qobuz desktop app.

excellent tool for learning more about your audio files, confirmed that both files contained the full 24-bits of data, and that there is high-frequency content north of 45 kHz that wouldn't exist in a file upscaled from a CD-quality source. A null test confirmed that the Qobuz download and stream sound the same (see **Figure 9**).

The State of High-Resolution Streaming

Let's take a step back. Tidal has more than 1 million hi-res tracks available for streaming, but it masks what hi-res means for each track. And we have Qobuz with more than 2 million hi-res tracks, though many are barely hi-res at 44.1 kHz/24-bit. And the vast majority of each's catalog remains CD quality. One might come to the conclusion that hi-res music just isn't that available. I've heard speculation that music license holders fear unlocking the possibility for hi-res music piracy. No one can deny that capturing the audio output of a computer is easy, but Qobuz isn't concerned. "If you look at piracy numbers, say 10 years ago and now, they're almost at zero," Solomon said. "It's still going on—I'm not naive. But the numbers suggest that people almost totally lost interest, at least in music piracy, because [music is] so available." Tidal has many hi-res albums that Qobuz doesn't, and Qobuz has many hi-res albums that Tidal doesn't. But overall, hi-res availability is growing. Solomon added, "It's all about ingestion. If you look at our site now, and then you look at it in a month, there will be another 25,000 cuts on there. Yesterday, we just signed a deal with Channel Classics, and now we've got the whole Channel Classics collection in hi-res on Qobuz that we didn't have the day before."

I asked Solomon if Qobuz is working harder than Tidal at growing its hi-res library. He said, "We've got two million and they've got one million, and we're just starting. If you're only going to have one service and you're truly into hi-res, we're the only game in town. You'll never be able to catch up with us because we just keep adding whole catalogs. We're about to add Blue Coast [Records]. Nobody's got Blue Coast. The only way you've ever been able to get Blue Coast is to buy them, and we're gonna be streaming them."

The Value Proposition

We've explored the positives and the negatives of Qobuz. Its metadata needs work, the interface feels sluggish, and Qobuz has dubious skill in pointing you to new music. But where Qobuz excels is sound quality, hosting the largest hi-res streaming library available, and the added bonus of a music history curriculum for those who choose to explore it. I see one target audience that values these features above others: the audiophile. Fortunately, Qobuz already has this group in its sights. When I asked about Qobuz's future plans, Mackta said the goal is to "gain widespread adoption of the streaming service by the audiophile community in the US and grow from there."

I don't see Qobuz offering value with its $9.99/month service tier. It lacks the algorithm-generated radio of its competitors, and it offers no fringe benefits, like Google Music's ad-free YouTube inclusion, or Spotify's Hulu inclusion. Qobuz's $19.99 plan merely ties Deezer with audio quality, and for the same price, Tidal offers hi-res. It isn't until the $24.99/month Qobuz Studio tier that the value begins to make sense. This service offers twice as many hi-res tracks as Tidal, and these tracks are delivered to more devices, with transparency on the files being streamed. This is exactly what hi-res streaming has been missing. Given time, I expect its metadata to improve and its curated playlists to expand.

Properly implemented algorithm-generated radio could completely overturn Qobuz's music discovery limitations. And the company has demonstrated a thirst for collecting hi-res music as Qobuz aims to set itself apart from Tidal. Even in its current state, I believe Qobuz to be the best option for the audiophile who knows what he wants to listen to and cares more for playback quality than any other aspect a streaming service offers.

Resource

J. Martins, "Master Quality Authenticated (MQA): Redefining the Source for Music," *audioXpress*, June 2016, www.audioxpress.com/article/master-quality-authenticated-mqa-redefining-the-source-for-music

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Enabling Higher Quality Audio Capture for Smartphones

The rise of user-generated content is placing more and more demands on the capture capabilities of today's smartphones. Yet while imaging and camera technologies have improved significantly over the past few years, audio recording features have been lagging behind. Until Nokia's OZO Audio.

By
Jyri Huopaniemi and Kai Havukainen
(Nokia Technologies)

User-generated content is big. Over half of all consumers globally share their videos through mobile. Millennials spend 30% of daily media use viewing user-generated content (5.4 hours)—almost as much as print, radio, and television combined, which comes in at 33%. Yet while imaging and camera technologies have improved significantly over the past few years, audio recording features have been lagging behind until recent advances in spatial audio processing.

Introducing Immersive Audio Capture

Recent advances in spatial audio processing have seen the audio capabilities of today's smartphones improve significantly. For smartphones with two or more microphones, we can now capture and process spatial sound, enabling faithful reproduction of the spatial sound field and salient audio cues that are present in the recording.

But audio innovation doesn't stop there. The combination of accurate spatial audio analysis with smart multi-microphone audio processing algorithms enables an entirely new set of spatial audio features to be introduced, including advanced audio focusing and audio zoom.

Here, we look at the key factors and technologies behind making modern smartphones into true audio recording devices and showcase the latest application examples.

Innovating Audio Capture through Algorithms

Mobile and consumer devices come in many different shapes and sizes, from large smartphones to dedicated audio and visual capture devices, such as cameras and video recorders. The technology is incredibly advanced. Today, we practically have home recording studios that can fit in our pockets. But these capabilities have yet to be unlocked or—at best—are largely unusable.

How do we unlock these new audio capture experiences? First, microphones must be placed in such a way that ensures optimal audio capture. There must be at least two microphones present in the device to enable these capabilities—but three or more is better. Put simply, the more microphones, the more use-cases and audio fidelity accuracy you can achieve.

But simply slotting in more microphones by themselves is not enough. Intelligent placement of microphones on devices can overcome the most challenging form factors (e.g., slim, modern smartphones). Embedding high-quality spatial audio does not mean we have to re-invent the wheel.

The design of advanced audio capture systems requires a coordinated implementation, combining the design of product acoustics and spatial audio algorithms. Product design involves many requirements that manufacturers need to consider

and seldom can the number of microphones or their placement be selected freely. Even though a modern smartphone design raises a very challenging form factor for spatial audio capture, the design can be optimized for excellent performance in all conditions, without a need to rely on external microphones.

If optimizing hardware is one side of the coin, then the software is the other. Creation of true-to-life immersive audio recordings can be delivered through OZO Audio smart algorithms, which are carefully tuned to provide the best performance for any hardware design.

This approach means OEMs and ODMs don't need to go back to the drawing board when it comes to the form factor. The algorithm does the heavy lifting and the design can be adapted to the desired form factor and microphone locations through a specialized algorithm tuning process (see **Figure 1**).

True-To-Life Immersive Spatial Audio

For the Nokia OZO Audio portfolio, the basis of all our algorithms is in the analysis of the multiple microphone signals. These allow us to simultaneously enable high-quality spatial reproduction, as well as being able to provide audio focus and zoom capabilities.

To maximize the performance of the audio capture technology for a device of any shape and size, we can utilize several alternative design methods. The methods include rapid prototyping techniques enabling acoustic measurements or modern virtual prototyping techniques modeling the product acoustics. The key variables addressed in the design are related to the device's dimensions as well as the microphone locations, along with other acoustic details. This workflow enables early evaluation of the target performance and efficient comparison of alternative designs without a need to delay the performance evaluation until mature target product hardware becomes available.

So, what are the audio capture capabilities we can expect? There are three primary capabilities that radically transform user-generated content—spatial audio capture, audio focusing, and audiovisual zoom. These features and the processing flow are described in Figure 1.

The aim of spatial capture is to faithfully reproduce the full spatial extent of the captured sound scene. Human hearing is omnidirectional—we are able to hear sounds from all directions—and it is able to distinguish sounds coming from front and back, left or right, up and down. To deliver true-to-life experiences, this must be replicated.

Traditionally, spatial recording is achieved by using expensive and dedicated multi-microphone systems and using audio formats that enable modifying the captured audio into desired use case, such as Ambisonics. By utilizing two or more microphones, as mentioned above, we can capture and analyze the content's spatial features and replicate human hearing capabilities.

This is Audio 3D—superior spatial sound capture. Audio 3D captures and delivers a natural sound experience within one degree of accuracy, similar to how the human ear works. With Audio 3D, consumers capture the full richness, depth, direction and detail—without missing a moment, even if it's out of frame.

Focusing on the Sounds that Matter

But spatial audio capture and playback is just one application. By combining with artificial intelligence machines already present in a device, we can unlock further audio innovations that radically transform the content experience.

Audio focus is one of these—allowing users to focus the audio in a direction that matters when capturing content, similarly as we do for video. An audio focus feature enables users to capture the direction of sound during recording. This gives them the power to focus on only the desired sound from any direction and attenuate all other sounds and background noises.

Imagine you are in a crowd—a very noisy one—in a busy tourist destination. You are recording what you see and are narrating the video you capture to later share on social media. Usually, your voice is

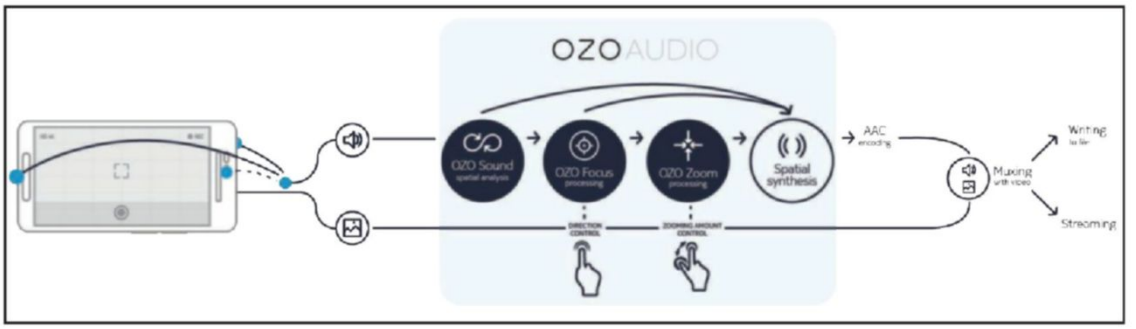


Figure 1. This diagram explains how OZO Audio processes captured audio.

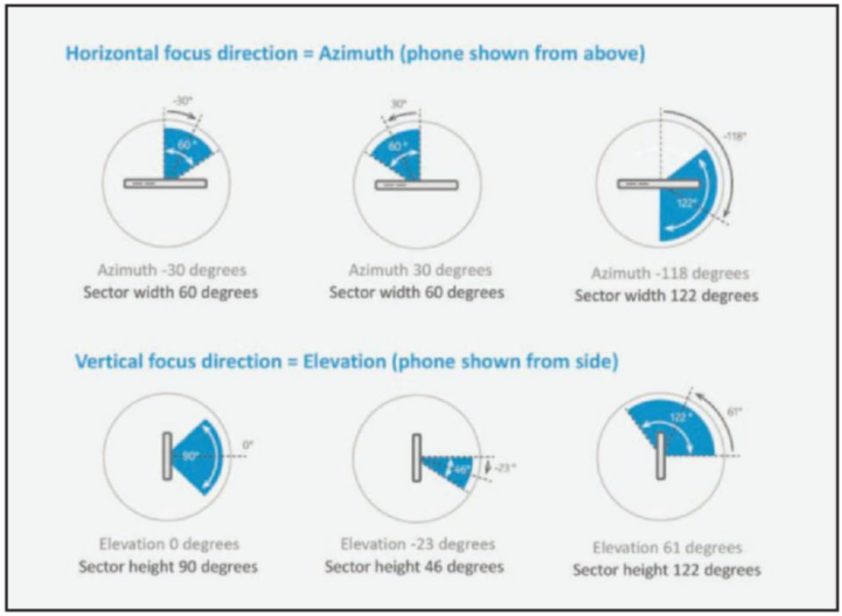


Figure 2. Audio Focus capabilities include audio directional controls. Parameters can be changed dynamically while recording.

competing against all the distracting surrounding noises—it is drowned out. But with audio focus technology in your device, you can focus on your voice and eliminate the background noises you don't want to feature. Your voice comes in crisp and clear, matching the high-definition video you capture.

We can achieve this in consumer devices through advanced multi-microphone audio capture algorithms that analyze and process the audio signals within the device software. For the user, this is delivered through a user interface in native or third-party video recording applications, allowing them to control the audio focus capability, either manually or automatically.

This thinking led us to develop Audio Focus, which minimizes distracting background noise and captures sound from a specific direction, whether the user is shooting a selfie or a street musician.

Audio Focus makes switching from rear-camera video recording to selfie recording simple and effective. It enables the user to adjust audio to a specific part of the screen and select what matters, and—with audio focus parameters—the direction and sector size, too. By integrating with artificial intelligence algorithms in the device, Audio Focus also enables users to maintain audio focus on moving people or objects and automatically follow a sound source with audio focus parameters controlled by object tracking (see **Figure 2**).

Realizing Audio-Visual Zoom

The ability to zoom video is now a given for any image capture device or app. We are all used to being able zoom through pocket cameras, digital single-lens reflex (DSLR) cameras, and through our smartphones. It is an essential capability. But the one thing that this function is missing is being able to match the audio to the zoomed area of the video.

Audio-visual zoom realizes this capability and resolves the disparity between what we see and what we hear. When you zoom the area of video during recording an object of interest, the audio zooms in line with this. It allows you to reduce unnecessary background noises and retain the high fidelity of the visual and audio content in view.

Imagine a busker playing on a busy street. You want to capture this moment, but are a little distance away, so you zoom the video. Before, this would result in you capturing the subject of the video, but still hearing everything else around you.

But with audio-visual zooming, when you zoom the video, you hear what you capture—delivering a superior content experience. It gives the user even greater control, allowing them to hear what they want to hear. This is why we integrated the Audio Zoom functionality into the OZO Audio portfolio. Audio Zoom features intelligent audio zooming that enables the user to dynamically adjust audio to the area of zoomed video. As the picture zooms in on the subject, so can the audio (see **Figure 3**).

The Importance of Universal Playback and Sharing

There is a rapidly increasing amount of user-generated content being created and consumed on smartphones. This means there is a new requirement for OEMs and ODMs to ensure that all the above spatial audio functionalities can be universally played back and shared across any device or platform. If one user can capture great audio but is not be able to share it with anyone else, then there is no point to the technology.

Universal playback and sharing means that a user can send the video or audio to anyone else and they can also experience true-to-life, immersive

About the Authors

Jyri Huopaniemi leads product and technology strategy for the Media Technologies organization in Nokia Technologies, the licensing arm of Nokia. Jyri joined Nokia in 1998 and has 25 years of experience in developing and productizing breakthrough audiovisual and software technologies. He has actively contributed to multiple fields of digital media, including audio and imaging technologies, mobile platforms and virtual reality solutions. Jyri received his doctoral degree in 1999 in electrical and communications engineering on the topic of 3-D audio and virtual acoustics, and he has published more than 60 peer-reviewed technical articles in international journals and conferences.

Kai Havukainen is a senior product manager for audio solutions in Nokia Technologies. Kai holds a master's degree in signal processing (2003) and has nearly 20 years of experience at Nokia. He has been working in various roles, including audio engineering, business development, technology licensing, and sales.

experiences—even if they do not have the same spatial audio software running on their device. It means that these experiences can be shared across any social media platform and heard through any device—regardless if it is a laptop, tablet, or smartphone.

When designing OZO Audio, we ensured this was a key design requirement. This is why OZO Audio also utilizes common audio standards (e.g., AAC), so content can be played on a range of popular devices and shared on social media.

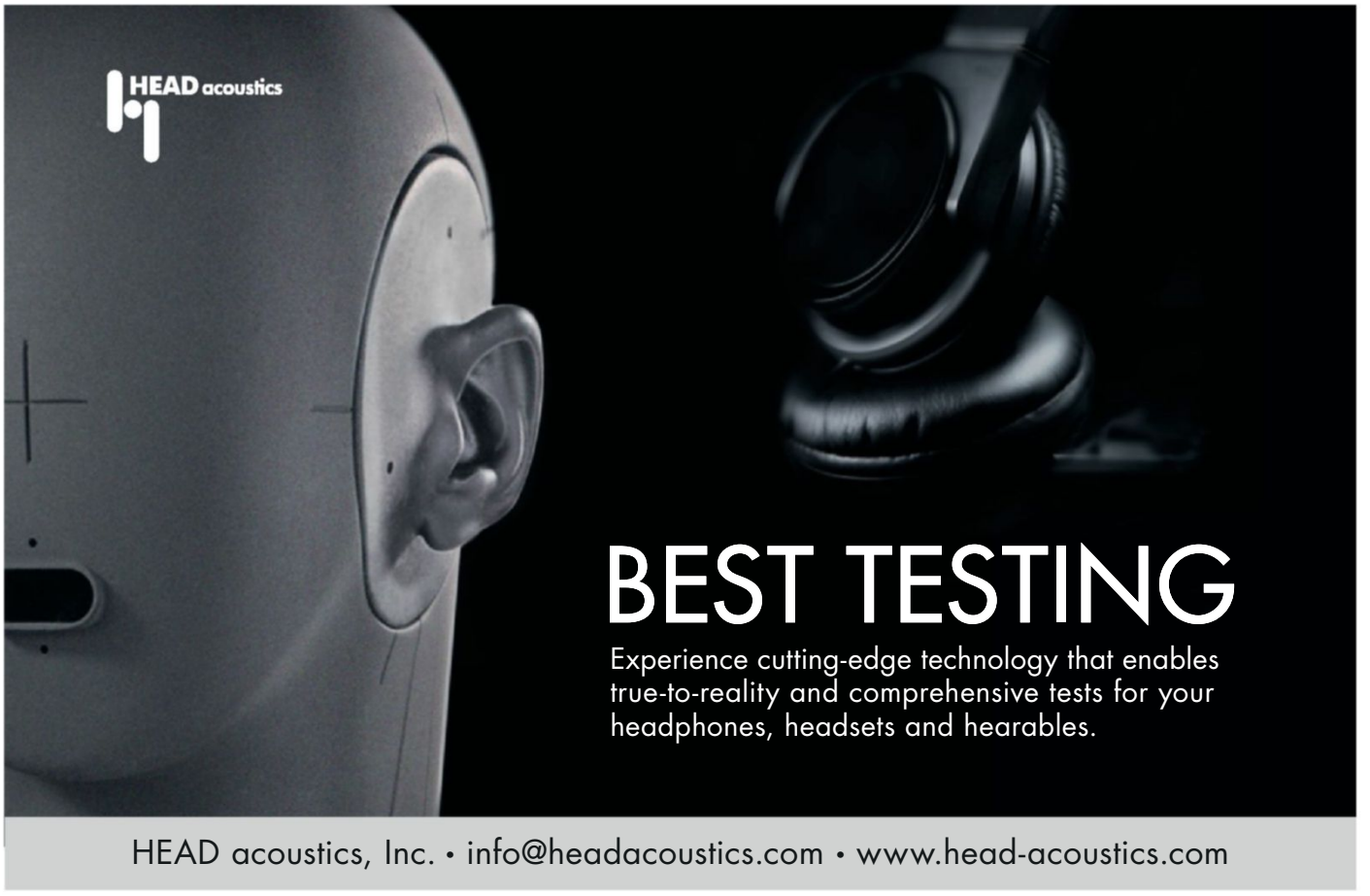
Re-Defining Audio on Consumer Devices

People are telling their stories with technology, and they want devices that allow them to capture the full richness of their lives—both in pictures and sounds. OZO is an evolution in audio, imaging and video technologies. It is born from our proven track record in innovation, using industry-leading technology to enable anyone to seamlessly capture and share the reality of the moment with unprecedented accuracy and ease. OZO enhances the way we see, hear, and feel our human experience.

The OZO Audio portfolio transforms how people capture and share their moments through intelligent audio, imaging, and video technologies. Sharing our lives and fostering interaction and community has become more important than ever, and we need the ability to do it in a way that helps us connect more naturally—and in a more authentically human way—with others.

The Nokia Media Technologies group offers licensing of innovative audio and video technologies to manufacturers of consumer devices, providing a key opportunity for device differentiation for its customers. It is time we unlock and let consumers enjoy amazing sound capabilities with their mobile devices, bringing audio up to the same quality we expect from video.

Figure 3. This is a UI demonstration of how Audio Zoom automatically and dynamically adjusts audio focus to the area of the zoomed video.



Testing Smart Devices

This article addresses the challenges and solutions for testing the wider class of "smart" connected, voice-activated and sensor-equipped devices, proposing solutions to implement basic acoustic tests and real-world tests of greater complexity.

By
Steve Temme
(Listen, Inc.)

Smart devices such as smart speakers, hearables, wearables, robots and voice-controlled car entertainment systems are notoriously complex to test. They have numerous interfaces from Bluetooth to cloud-based/Wi-Fi, and contain much signal processing, both on the record side (e.g., beamforming and background noise filtering) and on the playback side (e.g., loudness and active noise cancellation). This means that their characteristics change according to "real-world" conditions such as the physical environment and background noise. Furthermore, their multifunctional nature means there are many aspects of the device that may need to be tested, ranging from voice recognition to music playback or even operation as a hands-free telephone. Due to their complex nonlinear use cases, these devices often need to be tested at different levels and different environmental conditions (e.g., different physical setups and background noise, different signals etc.).

Although as yet there are no standards for testing smart devices, we can borrow principles and test configurations from many other audio devices and use existing standards such as the European Telecommunications Standards Institute (ETSI) and The Institute of Electrical and Electronics Engineers (IEEE) background noise and telephone test standards. Flexibility of the test system, and experience with testing a wide range of acoustic devices is critical to enable a device to be completely characterized. In this article, we will discuss how to implement basic acoustic tests and then discuss some of the more complex real-world tests that may be carried out on smart speakers and other smart devices, along with the techniques and standards that may be used. Finally, we present a check list of the test-system functionality you should look for when choosing a system to fully characterize a smart speaker or other smart device.

Basic Acoustic Tests

Basic acoustic tests essentially measure the device under static conditions to measure microphone array and speaker characteristics (e.g., frequency response and distortion). While the measurements themselves are similar to standard loudspeaker and microphone measurements, injecting the stimulus and extracting the recorded signal present challenges. Unless you happen to be the device's manufacturer and have access via a debugger or other remote access tool, or are testing speaker Bluetooth performance only, you have a true black box test solution. This is generally the case when comparing the performance of different speakers, for example, for reviewing or competitive benchmarking. Without knowing what is inside the device under test (DUT), you need to get signals in and out of it, and the only way to do this is like an actual user—via the cloud using voice commands.

To further complicate matters, the test signal needs to be in the cloud to enable playback. Each manufacturer's ecosystem is different in how it plays back from the cloud. Some enable you to upload your own recordings (although you should bear in mind that these will probably be compressed). Others require them to be on a media streaming platform (e.g., Spotify). For microphone testing, some systems such as Alexa enable you to access recordings you have made; others do not for security and privacy reasons, which makes microphone

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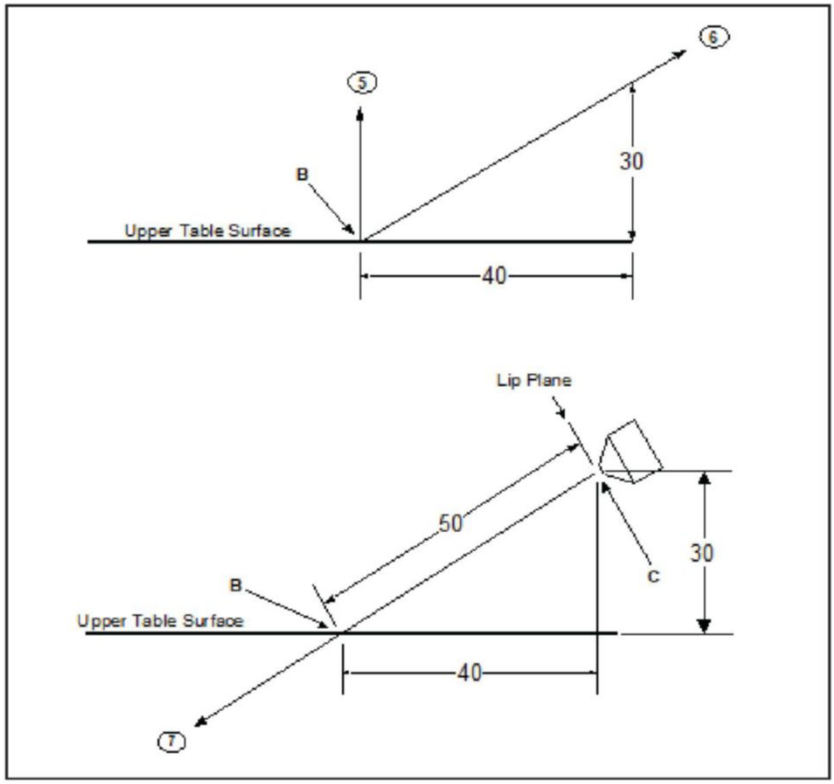


Figure 1. This is the IEEE 1329-2010 standard test setup with a table (dimensions in centimeters).

testing challenging. Although the actual physical testing setup is very similar from device to device, for each one it is necessary to understand out how to wirelessly route the signal. Furthermore, each device needs activating with a different wake word, has a different delay compensation, and records for a different amount of time after it hears the wake word. This, therefore, needs figuring out (largely by trial and error) for each speaker that you need to test.

Let us first discuss the physical setup. At this point there are no test standards specifically written for smart speakers. However, their practical use case closely resembles speakerphones, so the physical geometry recommended by IEEE 1329-2010 is a good starting point. Following this standard, tests would be conducted inside an anechoic chamber with the DUT placed on a table (see **Figure 1**).

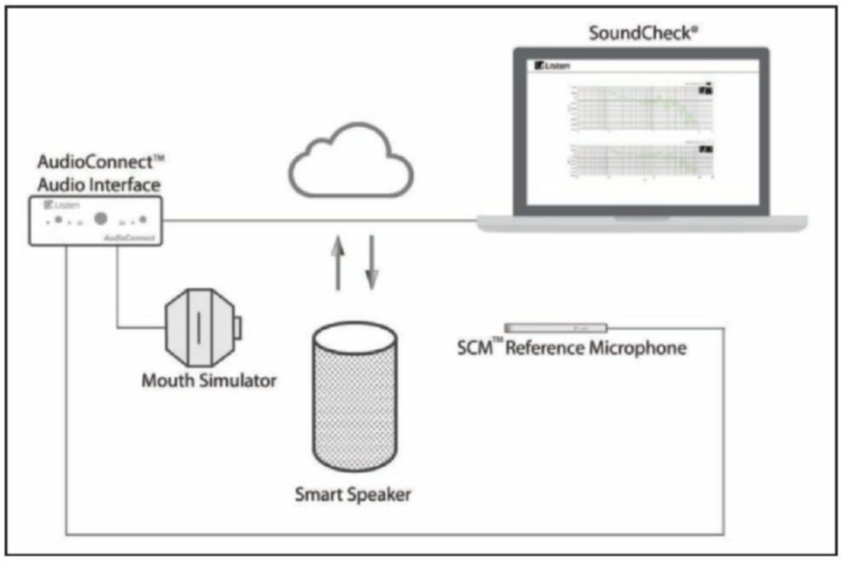


Figure 2: This is a viable test configuration for the basic loudspeaker measurement of a smart speaker.

The setup includes a speaker or artificial mouth to play the voice activation commands, a reference microphone to record the signal, and of course, the DUT (see **Figure 2**).

Either a mouth simulator or loudspeaker may be used to play the voice commands, but generally loudspeakers are preferred—measurements are made at a distance to represent a voice command and loudspeakers typically can play louder and are easier to equalize as they have a flatter frequency response.

Let's now discuss the steps necessary to create a test. It is ideal to automate the test as much as possible, so wake words and test stimuli should be pre-recorded and uploaded so that they can be triggered by the test system.

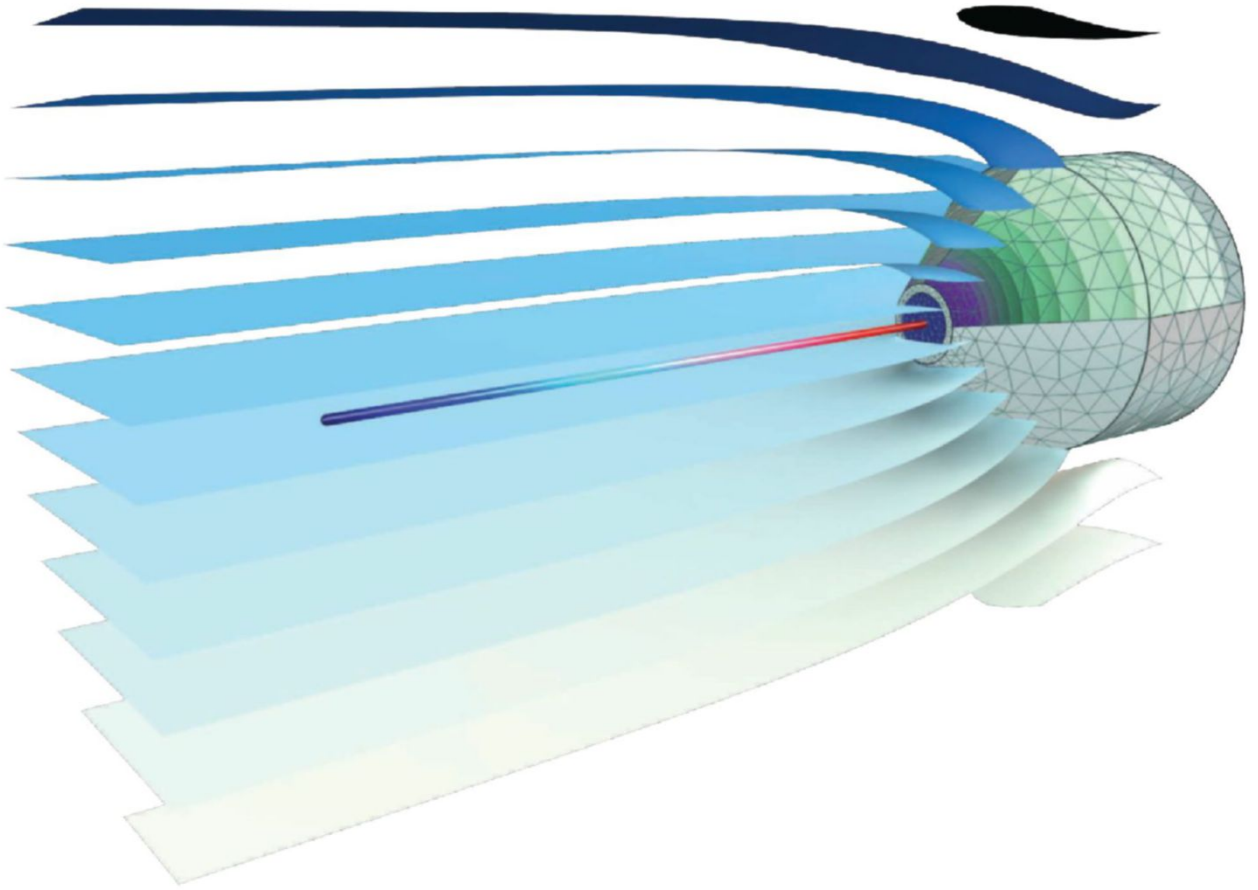
First, we need to record the activation signal. This might be something like "Alexa, play test signal 1." Naturally the system's appropriate wake word should be used. If a wide range of speakers from different manufacturers is being tested, it makes sense to record the wake words separately from the command so that any combination can be selected.

The test stimulus also needs to be created and uploaded to the cloud. Bear in mind that it might be compressed when it is uploaded, which could introduce some distortion. You can verify that the MP3 encoding itself does not degrade the test signal by encoding and decoding the test signal, and using transfer function analysis between the original signal and the encoded version. We'll go into test signals in more depth later, but for basic speaker performance testing we tend to use a standard Stweep (stepped swept sine sweep).

Once the stimuli and voice commands are created and uploaded, the test sequence simply needs to play the wake word followed by the activation command via the speaker and capture the response with the reference microphone. Stimulus and analysis are then compared as in a traditional measurement to objectively qualify characteristics such as frequency response and distortion.

However, it is not as simple as it looks—although the good news is that with the right test system, the complicated analysis happens "under the hood" so there is very little that the user needs to do other than make sure their test system has been told to analyze it correctly.

Hearing aids can't solve the cocktail party problem...yet.



Visualization of the total acoustic pressure field around and inside an elastic probe tube extension attached to a microphone case.

Many people are able to naturally solve the cocktail party problem without thinking much about it. Hearing aids are not there yet. Understanding how the human brain processes sound in loud environments can lead to advancements in hearing aid designs.

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One of the main challenges that test system manufacturers have faced in developing accurate smart speaker tests, is overcoming sampling rate error. Traditional measurement systems rely on the DUT having a synchronous input and output. As a category, smart speakers are intrinsically open loop, which means that although the device can record or playback a signal there is no synchronous signal path. This introduces the possibility for sampling rate error.

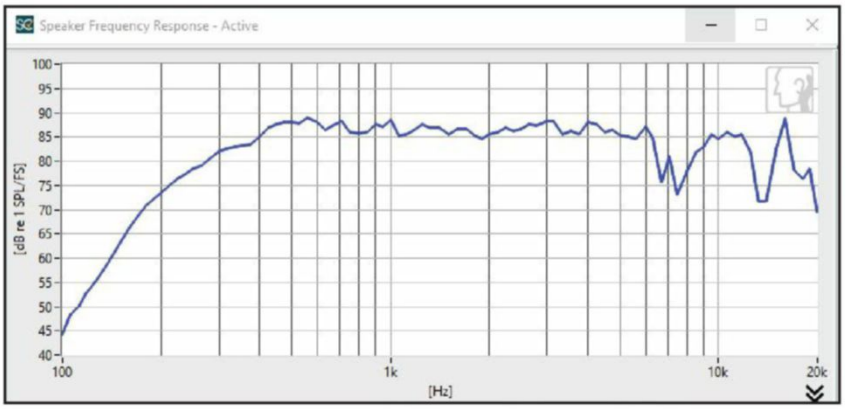


Figure 3: This graph shows the frequency response of a smart speaker.

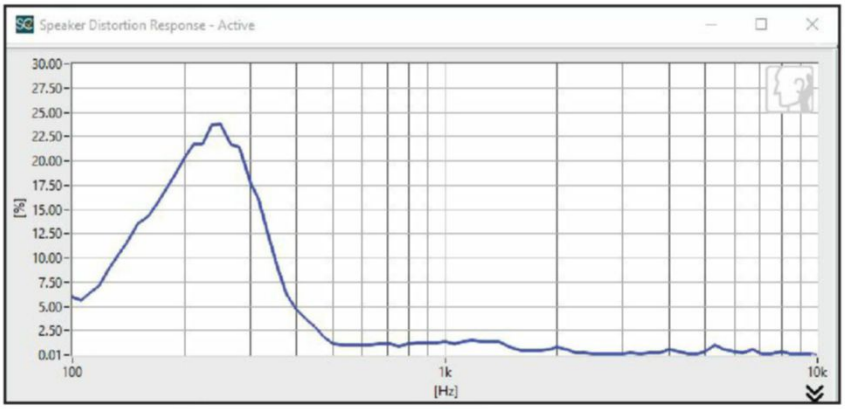


Figure 4: This graph shows the distortion response of a smart speaker.

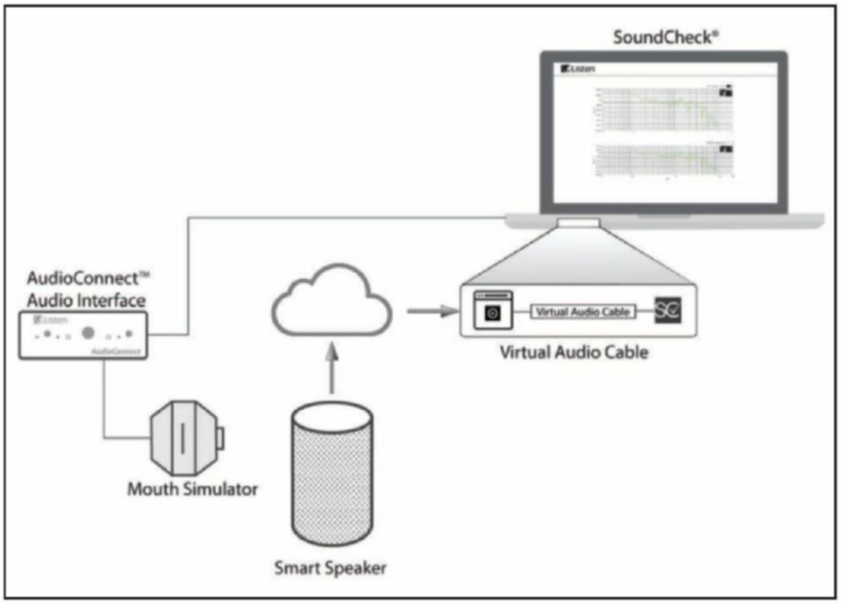


Figure 5: Here is a viable test configuration for a basic microphone measurement of a smart speaker.

In other words, the device may record a signal to a file with a sample rate of 44.1 kHz, but it may have been recorded at a slightly different sample rate (e.g., 44.09 kHz) due to skew in the actual rate of the crystal clock used to drive the sampler. A similar error can occur when playing back test signals. That is, the test stimulus may be sampled at 44.1 kHz, but due to error in the playback sample rate the file is actually played back at a slightly faster or slower rate. This sampling rate error results in the component tones of the test stimulus being shifted to either a higher or lower frequency. This shift can then lead to measurement errors due to loss of coherence between the stimulus and response signals.

To overcome this sampling rate error, an algorithm is applied, which searches the beginning of the response waveform for a "trigger" that is used to provide a reference point for alignment and shifting of the stimulus and response signal. This trigger may be a steady-state sinusoid at a pre-set level and frequency, or for more robust performance that is less susceptible to false triggers, a log chirp (SoundCheck offers both). The signal is then shifted to DC using a heterodyne filter and all other frequencies are filtered out. The output of the heterodyne filter includes the phase information, which is ultimately used to estimate actual playback or recording sample rate of the response signal. With this information, the entire response waveform is resampled to the correct stimulus sample rate prior to analysis. This frequency shift step corrects for sampling rate error in the device and makes testing them straightforward.

With the test system taking care of all these, frequency output and distortion can be displayed in exactly the same way as for a wired speaker, or a speaker connected via a Bluetooth interface or directly via a debug port (see **Figure 3** and **Figure 4**).

Of course, it is also necessary to measure the microphone array in the speakers. The setup for this is very similar, except that in this case, we need to play the wake word and the test stimulus through the mouth simulator, then download the test stimulus from the cloud and route it back into the test system using a virtual audio cable (see **Figure 5**). Once the signal has been brought into the test system this way, as with the speaker test, the appropriate calculations are applied to overcome frequency shift and error, and it can be analyzed as normal (see **Figure 6 and Figure 7**).

Advanced, or Real-World Testing of Smart Devices

While the basic speaker test previously described is ideal for measuring frequency response and distortion, it barely scratches the surface in terms of the breadth and depth of tests that one might want to carry out on a smart device in the R&D lab. Consider that smart devices are generally nonlinear and their response probably varies according to where it is in the room, what the background noise is, how loud the background noise is, whether it is being used to play music or to communicate voice, and more. These considerations alone bring up a multitude of tests that could yield valuable information about the device's performance. While the specific tests that are carried out will vary according to the manufacturer's design objectives, we can discuss some of the test methodologies that we might use and some of the industry standards that provide a useful starting point.

A significant difference from basic tests is that advanced tests generally require the use of speech or music as test signals. This means a test system that can support such test signals is essential, and it is also important that the system can measure the

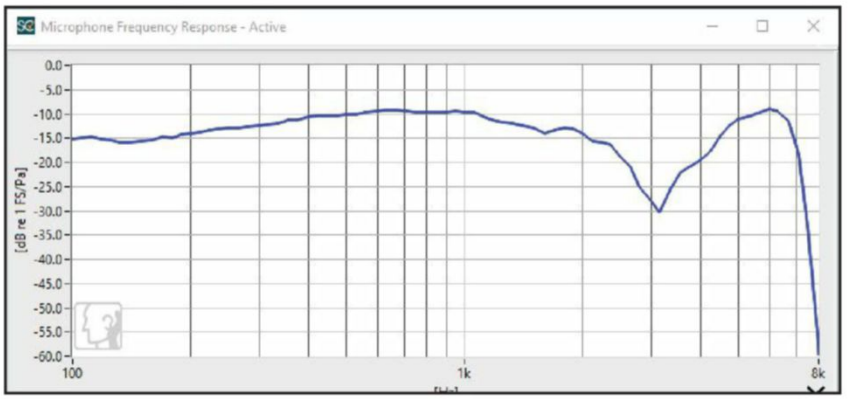


Figure 6: This graph shows the frequency response of a smart speaker microphone array.

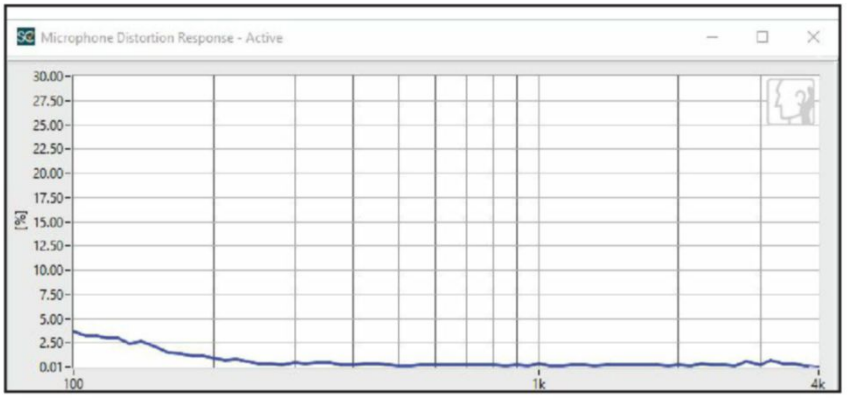


Figure 7: This graph shows the distortion response of a smart speaker microphone array.

active speech level of speech-based signals to ensure calibration and standardization as commonly used in telephony measurements.

Voice Recognition

Speech recognition tests are important for evaluating the device's ability to understand voice commands. The physical setup is the same as in the basic test outlined above.

One option is to create a series of specially titled musical tracks and upload them to the cloud. The track titles are the words or phrases that you want to test, and the actual content of the tracks is a single, dual, or multitone that enables you to identify the track by its audio content. Generally, "Harvard Sentences"—a collection of phonetically balanced sentences that use specific phonemes at the same frequency they appear in English—are used for the track titles. These are recommended by "IEEE Recommended Practices for Speech Quality Measurements for Voip, Cellular, and other telephone systems" and widely used in telecoms voice recognition. Pre-recorded versions of these can be downloaded online, or your own can be recorded (e.g., if you are testing the device with different accents). These track titles are used to request playbacks, and a limit is used to detect whether the correct, or indeed any, signal was played back.

There are many variations on this test, including testing different voices and accents, testing different speaker locations, testing an active talker versus an interfering talker or background noise levels, testing with different types of background noise, and more. Where background noise is needed, ETSI standard ES 202 396-1 contains binaural recordings made in different noise environments (e.g., cafe noise, traffic noise, subway etc.). While there are many ways of playing these (e.g., through purchasing an expensive dedicated system or by programming your own), SoundCheck offers an optional ETSI standard ES 202 396-1 module. This integrates the ETSI standard library with the test system, playing the background noise at calibrated levels that can be controlled and adjusted as part of a pre-programmed test sequence. This test sequence may include loops to incrementally increase the volume or change the noise and repeat the test, making it simple to create a test that increases the background noise by fixed levels until the voice is no longer accurately recognized. In addition to being a very cost-effective approach, it also significantly simplifies the physical test setup and reduces test development time. Background noise is necessary for a wide range of smart device tests including voice recognition, signal-to-noise ratio (SNR) optimization of mic arrays, beamforming directionality studies, and more.

Indy Acoustic Research

Independent acoustic design and test lab Indy Acoustic Research, LLC (IAR) has developed extensive expertise in testing smart devices such as voice-activated speakers. With a solid background in speakerphone, loudspeaker, and microphone tests, IAR's team has a thorough understanding of how to test the various elements of smart devices. In addition to contracted testing of smart devices, IAR has also invested in developing new test methods for characterizing smart speakers.

Smart speakers are evaluated in an anechoic chamber using SoundCheck and a test setup very similar to that described in the article. The device under test (DUT) is placed on a suspended table, a calibrated Brüel and Kjäer mouth simulator is used to generate the stimulus for the microphone tests, and a speaker monitor is used to generate white, pink, and ETSI noise to enable measurement of performance with background noise and interfering talkers.

Basic performance tests on smart speakers include frequency response, sensitivity (output versus input), total harmonic distortion (THD), and total harmonic distortion plus noise (THD+N). The calculations used to create these graphs are carried out within SoundCheck, so once the test signal is captured and routed into SoundCheck, there is no additional operator intervention. The effect of background noise interference is also quantified by adding interfering noise played via a speaker monitor in the room. Background noise is chosen from a range of options—ETSI noise, pub noise, office environment, or noisy children—and the noise level is selected. Measurements are repeated, and using a post-processing step in SoundCheck, a difference response is created based on the baseline and noise-degraded response.

All data is exported to an Excel file containing the raw numerical data, graphical results and testing comments. Retaining all test data in one place helps with any future re-analysis or comparison. For example, if it is necessary to compare a new, redesigned model to an older one it is easy to run the tests for the new model and compare it to identical saved data from the older model. Storing all data including wav files even makes it possible to listen to the comparative recording quality as well as compare measured parameters.

Photo 1: A smart speaker test is being conducted in an anechoic chamber at Indy Acoustic Research.

Voice Quality

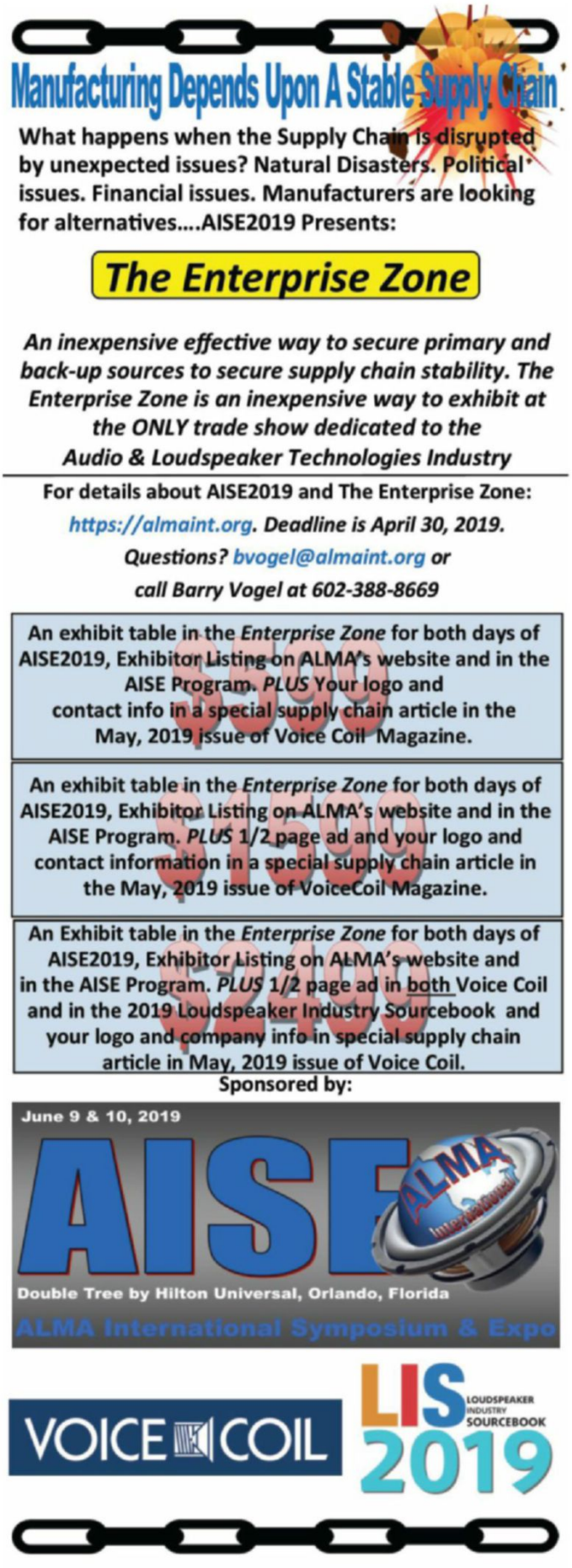
Smart speakers often function as handsfree communication devices. For such use cases, we can lean heavily on telephone test sequences such as the Telecommunications Industry Association (TIA) standards for handsfree devices (TIA920-B). In fact, a smart speaker can be tested as a speakerphone exactly as defined in the TIA standard. Although the test setup is identical to the basic smart speaker test previously outlined, the test sequences for measurement to the standard are highly complex and either require considerable expertise in this area to create them, or they need to be purchased as an off-the-shelf package.

Choosing a Test System

Smart devices represent the convergence of tests from speakers, surround systems, microphones and recording systems, telephones, hearing aids, MP3 players and more. It is, therefore, advantageous to select a system from a manufacturer that truly understands these varying applications and has the various test stimuli, algorithms and post-processing functionality for these in their toolkit, along with the expertise. A checklist of the test system functionality that is important for characterizing smart devices should include:

- Tests both speakers and microphones
- Can create a compound stimulus with frequency and/or log chirp trigger to enable accurate open-loop testing
- Can be set to analyze only parts of the response signal, so that the trigger tone can be eliminated from the analysis
- Ability to use speech and music as test signals, calibrate levels, and equalize a mouth simulator (requires the inclusion of active speech algorithms)
- Background noise generation—although this can be generated externally, testing is much simpler and faster if it can be created by the test system and integrated with the test sequence
- Can accept signal via virtual audio cable in order to route in signals from the cloud through the computer's multimedia interface (e.g., WDM or Core audio)
- Availability of pre-written TIA test sequences if device needs to be tested as speakerphone (may be manually programmed with sufficient expertise)
- Sophisticated sequence writer that allows duplication of sub-sequences and loops - essential when repeating a test at various levels
- Bluetooth interface available, if testing over Bluetooth is desired
- Mentor A2B interface compatibility and control from within test system if testing automotive smart devices (e.g., in-car voice control)

It is a bonus if the test system is also available in a lower-cost, lighter version so that basic tests can be simply transferred to the production line rather than being recreated in a new system.



Industry Focus on Voice Applications

By
J. Martins
(Editor-in-Chief)

A new study from Juniper Research forecasts that there will be 8 billion products supporting digital voice assistants in use by 2023, up from an estimated 2.5 billion at the end of 2018. The research shows growth will be driven by smart home devices, with smart TVs, wearables, and Business to Business (B2B) and Business to Consumer (B2C) applications showing large growth opportunities. No wonder we can see manufacturers from all sides of the industry focusing on voice capture and recognition technologies.

As Juniper details, "After the launch of the Amazon Echo in 2014, the rest of the Big Five technology companies like Apple, Facebook, Google, Microsoft, Samsung, and a host of smaller companies all began releasing their own digital assistants on a variety of platforms. With an increasing number of these assistants integrating third-party apps and functionalities, their scope and capabilities is reaching the point where they can be used to provide and facilitate many services in consumers' lives and the beginnings of Business to Business (B2B) usage can be seen."

In fact, all market research companies highlight that the adoption of voice assistants has been one of the fastest-growing trends in consumer technology, and that voice-only interactions will increase in the future, with availability in personal wearable devices now propelling even more its adoption.

While voice digital assistants still need some time to reach a stage where their use will truly meet consumers' expectations, evolving from simple voice commands, and "skills," to true natural language interactions, the audio industry understands that enabling high-quality voice capture and speech-recognition processing solutions represent one of the largest business opportunities ever.

Voice Platforms

For those platforms to evolve, companies are both perfecting speech recognition tools that convert the spoken word to written text—an area with huge potential in business applications by itself, and Natural Language Processing, which involves much more sophisticated processing and is currently the core focus for artificial intelligence engines. Still, for a machine to be able to understand human speech, it first needs to be able to hear it. And, the usage scenarios still present several challenges, the first of which is to determine when a user's voice should trigger an interaction with a digital assistant.

Until now, the solution adopted by all voice digital assistants was the use of voice triggers or "wake-up words." The industry is currently focusing on local-processing through edge-computing to address the privacy concerns of always-listening and always cloud-connected devices (e.g., Amazon Echo speakers). The next step is to implement voice detection and user identification, in order to allow for multi-user scenarios and accurate separation of voice

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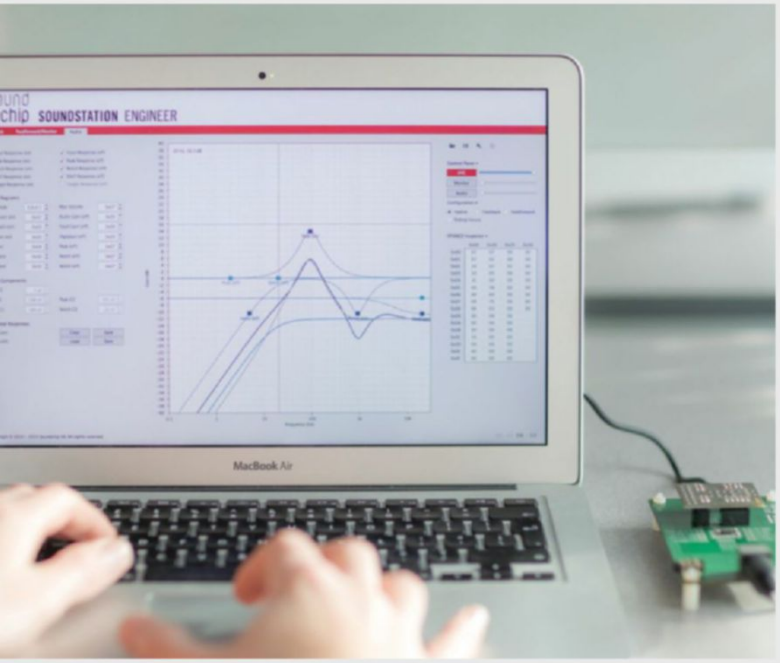

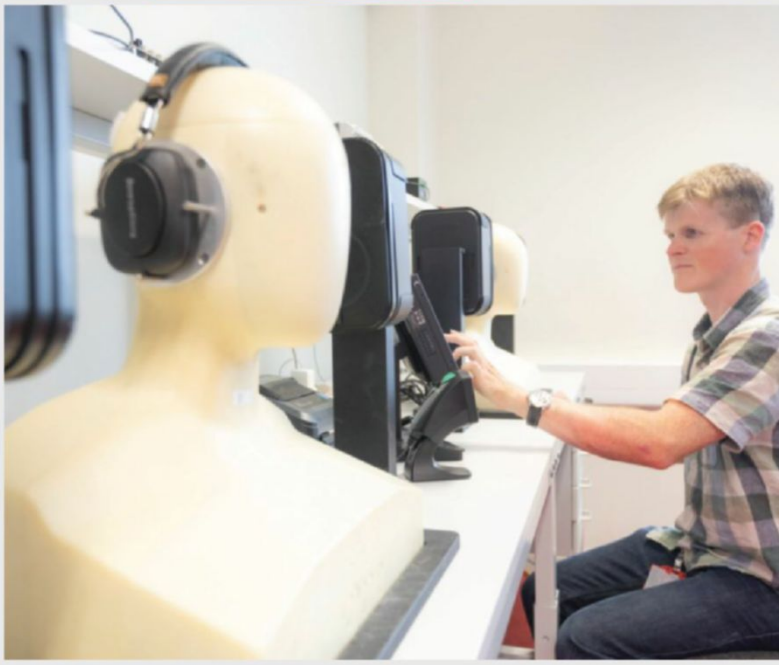

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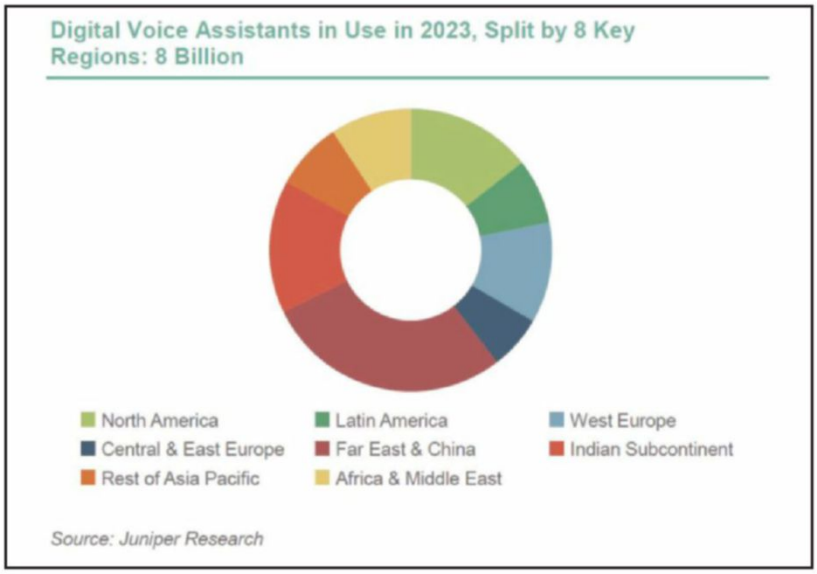

There will be eight billion Digital Voice Assistants in use in 2023, split by eight key regions, according to Juniper Research.

interactions even in noisy situations and—the greatest challenge—with music playing or TV sound in the background.

Of course, with the novelty of voice recognition now in high demand, many consumer electronics companies will opt to simply embrace Amazon's or Google's APIs and ready-to-use development platforms. But there are already many alternatives in speech recognition engines that are able to run on edge-devices, including fast-growing open-source alternatives, while many companies (e.g., IBM with its IBM Watson machine learning and analytics engine) can provide a powerful platform for research. Microsoft is another player in this field, increasingly focusing on services offered through its Azure cloud platform leveraging Azure IoT and AI services, including Microsoft's virtual assistant solution with Azure Bot Service and Azure Cognitive Services.

Sensory is one of the most successful companies providing high-accuracy speech-recognition solutions that can be embedded into mobile and other consumer electronic products. For companies looking for different types of voice platforms, companies such as Voicea are able to offer voice applications for business. Voicea, a company based in Menlo Park, CA, is the maker of the Enterprise Voice Assistant (EVA), which evolved from a voice collaboration platform with transcription and note-taking abilities to add new features, including an Artificial Intelligence (AI) phone dialer, voice-activated calendar reminders, and visual integrations to help increase productivity from workplace conversations. Voicea's EVA takes a dual-system artificial intelligence approach, which uses both AI and natural language processing (NLP) systems, emulating human behavioral systems by maintaining two systems: one focused on rapid response, broad-facing, and is always on, while the second system is deeper, deliberate, highly accurate and engages only when necessary.

A second company that is making significant inroads as a voice assistant platform comes from SoundHound, a voice-enabled AI and conversational intelligence technology company. SoundHound's Houndify Voice AI Platform, already supports highly sophisticated and natural interactions, leveraging its proprietary Speech-to-Meaning and Deep Meaning Understanding technologies, and intelligence in voice recognition and understanding.

SoundHound's Houndify voice and conversational AI platform has distinctive advantages, including the ability to support natural interactions. Houndify understands complicated queries and context, such as the user's location, previous queries, and compound criteria including exclusion. In addition, SoundHound's Collective AI architecture allows partners to have complete control of the user experience and their own innovation, while benefiting from Houndify's growing developer ecosystem.

SoundHound is currently the platform of choice for a number of automotive companies such as Daimler, Hyundai, Peugeot, Citroën, and Opel. It is also the voice platform used by companies such as Orange, NVIDIA, HTC, Onkyo (speakers), and Motorola (phones).

The Houndify platform's Collective AI architecture already provides access to knowledge and data from Yelp, Uber, and Expedia, as well as more than 100 other domains such as weather, stocks, sports, local businesses, flights, hotels, mortgage, and even interactive games.

Audio Processing

Another essential technology to enable accurate speech recognition is naturally digital signal processing (DSP). Advanced DSP algorithms are rapidly evolving to eliminate noise and isolate the speakers' voice for increased understanding, requiring higher processing capabilities and improved energy efficiency.

DSP Concepts, headquartered in Santa Clara, CA, is one of the pioneers in this field and provides embedded audio digital signal processing solutions delivered through its popular Audio Weaver platform. DSP Concepts specializes in microphone processing as well as playback processing, and is a leading supplier to top tier brands in automotive and consumer products.

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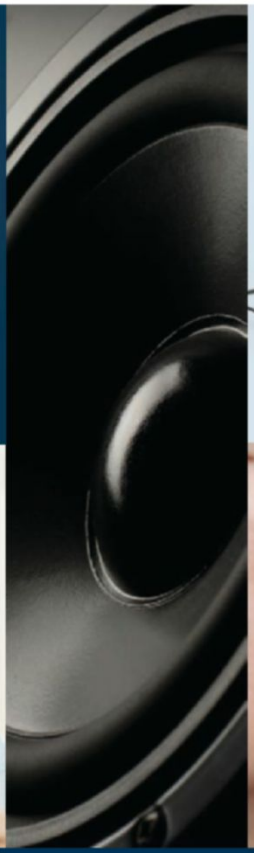

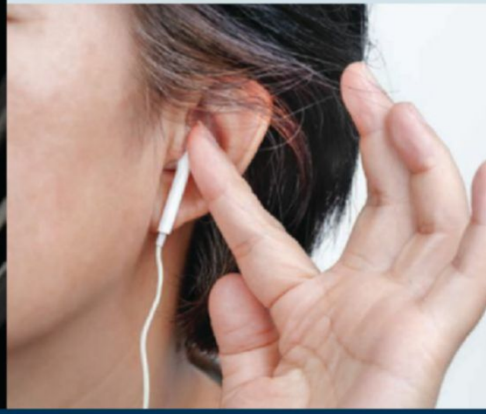

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Voice applications are showing up in the most unexpected categories. Japanese music instruments maker Roland introduced an Alexa-controlled digital piano this year.

recognition of voice commands by analyzing and processing the signal, separating voice relative to the level of surrounding noise and the level of the program material coming from the device itself, like when a smart speaker is playing music. In order to solve those challenges, DSP Concepts developed advanced processing technologies such as acoustic echo cancellation (AEC), noise reduction algorithms, and more recently its Adaptive Interference Canceller (AIC). AIC uses machine-learning techniques to work differently—and better—than other noise cancelling or noise reduction technologies on the market. Because AIC re-evaluates environmental noise continuously and can work with multiple channels, it adapts almost instantly to any new environment. This makes it ideal for portable applications, as well as for automotive applications. AIC is included as a module in DSP Concepts voice UI and automotive algorithms, and in the Audio Weaver modular DSP programming environment, making it easy to evaluate and adapt to any voice UI product.

DSP Concepts partnered with many of the industry's voice-focused companies to deliver development solutions targeting all sorts of voice capture applications. From supporting STMicroelectronics and Ambiq Micro microcontrollers, to Sensory's TrulyHandsfree voice engine, to MEMS microphone array solutions developed with Knowles and Vesper.

Another key player in the voice processing area is Cadence Design Systems, which recently announced its latest Cadence Tensilica HiFi 5 DSP solution for audio and voice. This is the company's first IP core optimized for high-performance far-field processing and artificial intelligence (AI)-based speech recognition processing. This fifth-generation HiFi DSP offers 2× audio processing and 4× neural network (NN) processing improvements versus the previous HiFi 4 DSP platform, making it ideal for voice-controlled user interfaces in digital home assistants and automotive infotainment. NN-based speech recognition algorithms are able to perform more tasks locally, rather than in the cloud, providing an answer to concerns of latency, privacy and network availability.

Ambiq Micro was the first silicon licensee of Cadence's HiFi 5 DSP, porting the solution to its Subthreshold Power Optimized Technology (SPOT) platform, available to product designers, ODMs, and OEMs.

Synaptics is another leading provider of digital audio processing solutions and far-field voice technology. The company's low-power far-field AudioSmart product portfolio was developed to support electronics manufacturers' efforts designing voice-enabled consumer and enterprise products. Synaptics' portfolio of solutions includes complete far-field voice DSP platforms for Amazon Alexa and Microsoft Cortana.

At CES 2019, Synaptics announced its new AS3*xx* series, a comprehensive family of Smart Edge AudioSmart solutions including the world's first fully integrated SoCs comprising neural network acceleration, a proprietary wake word engine with support for custom wake words, and highly-advanced far-field voice processing. Designed using a 22 nm process node—and targeting a broad variety of voice-enabled Smart Home devices including hubs, Wi-Fi repeaters, speakers, and appliances—the AudioSmart AS371 incorporates a new powerful machine learning

Resources

Ambiq Micro, https://ambiqmicro.com

Cadence Design Systems, www.cadence.com

"Digital Voice Assistants: Platforms, Revenues & Opportunities 2019-2023," Juniper Research, February 2019, www.juniperresearch.com

DSP Concepts, www.dspconcepts.com

Frontier Smart Technologies, www.frontiersmart.com

Sensory, www.sensory.com

SoundHound, Inc., www.soundhound.com

StreamUnlimited, www.streamunlimited.com

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Far-field voice capture applications in the home (e.g., smart TVs) are still a challenge for the industry. Cadence Design Systems recently announced its Cadence Tensilica HiFi 5 DSP for audio and voice, the company's first IP core optimized for high-performance far-field processing and artificial intelligence (AI)-based speech recognition processing.

SoundHound's Houndify voice and conversational AI platform has distinctive advantages, including the ability to support natural interactions.

engine using innovative Synaptics Neural Network Acceleration and Processing (SyNAP) technology.

The solution is able to support the industry's efforts for performing Automated Speech Recognition (ASR) and Natural Language Understanding (NLU) locally on-device. SyNAP enables advanced features such as user identification and behavioral prediction which allows voice assistants to perform ambient computing and a more intuitive interaction with users.

Full-Service Providers

For manufacturers looking to get to market quickly with advanced voice-enabled products, companies such as DSP Group, Qualcomm, and XMOS are able to support development efforts with some of the most advanced platforms currently available, as illustrated in *audioXpress'* extensive Voice Market Update published in our December 2018 edition.

The other option is to approach full-service providers such as StreamUnlimited, a company that recently expanded its range of solutions from wireless audio, multiroom and multi-channel platforms also to voice implementations. The Austrian audio development specialists now offer the modular StreamSDK software solution with Google Assistant integration and Chromecast built-in, as illustrated in its recent collaboration with Harman for the Harman Kardon Citation range of multiroom and multi-channel speakers.

Founded in 2005, StreamUnlimited originated as a team of 25 engineers, and in 13 years has built its own IP while more than tripling its size. StreamUnlimited is now a supplier of software solutions and modules for smart audio and voice assistant products, with office locations in Vienna, Bratislava, Shenzhen, and Mountain View, CA. Partnering with all major semiconductor companies and technology providers in the consumer electronics channel, StreamUnlimited also delivers solutions for Apple AirPlay 2, Amazon Voice Services, Spotify Connect, and a vast range of platform support.

Another pioneer in development platforms for wireless audio, digital radio, and voice recognition solutions is UK-based Frontier Smart Technologies. The company recently expanded its Minuet Smart Audio platform, to allow smart speakers, soundbars, and AVRs to be simultaneously controlled by voice-enabled devices from Google, Amazon, and Apple, helping manufacturers to differentiate, and not have to compete head on with first-party devices.

Frontier's new "WorksWith" reference design is powered by the company's well-established Minuet Smart Audio module and SmartSDK. This upgrade enables Smart Audio products, without built-in microphones, to be controlled by any voice-enabled device incorporating Google's Assistant, Amazon's Alexa Voice Services, or Apple's Siri—giving consumers the freedom to choose their preferred voice ecosystem.

With this approach, audio manufacturers and brands are able to position their devices as being controllable via any of the major voice assistant ecosystems, launching new products that are able to support "Works with Google Assistant," "Works with Alexa," and can be voice-controlled via an Apple HomePod. In addition, the platform also supports multiroom applications with Google Chromecast built-in and Apple AirPlay 2.

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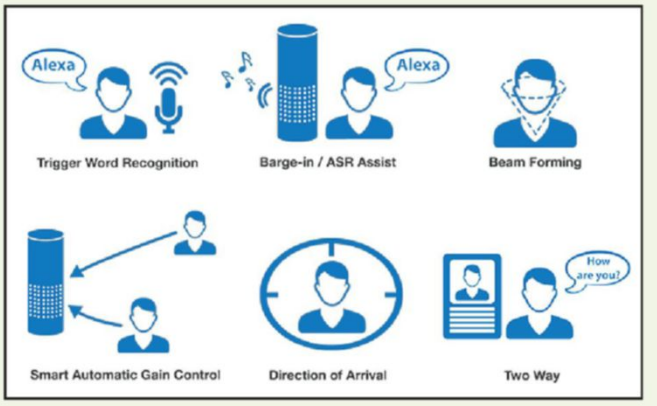
Development Kit for Amazon AVS Supports Far-Field Voice Interaction with Multiple Microphone Array Options

Microchip Technology, via its Microsemi Corp. subsidiary, announced that its AcuEdge ZLK38AVS Development Kit for Amazon Alexa Voice Service (AVS) now introduces far-field voice pick up and the flexibility to build devices with multiple mic array configurations. The audio front-end solution with one, two, and three mic configurations reduces bill of materials costs while addressing the requirements of small-form-factor designs.

As a leading audio front-end development kit with one-mic hands-free, and two- or three-mic far-field configurations qualified by Amazon, the solution features the ZL38063 audio processor, which includes a 300 MHz DSP and dedicated hardware accelerators for voice processing. It also features Stereo Acoustic Echo Cancellation (AEC), which enables barge-in recognition to improve the consumer experience in extreme audio playback environments.

Designed for world-class voice front-end audio applications, Microsemi's ZL38063 audio processor features the company's AcuEdge technology. This innovative technology is a set of highly-complex and integrated algorithms that significantly improves automatic speech recognition (ASR) for both embedded and cloud-based ASR solutions. The ZL38063 delivers audio enhancements that perform noise reduction and smart automatic gain control (AGC), enabling speech recognition at distance in noisy, real-world conditions.

With the flexibility to work with a variety of microphone options and configurations, ODMs and third-party developers can use the development kit to cost effectively achieve their design criteria across a range of applications and form-factor constraints. Available in multiple microphone configurations, developers can explore one-mic configurations in hands-free voice coverage for products where form-factor or cost constraints preclude the use of a multi-microphone array; two- and three-mic linear (180°) configurations, especially designed for soundbars, smart plugs and switches, set-top-boxes, or other applications typically positioned against a wall that benefit from 180° far-field coverage; and three-mic triangular (360°) configurations. Also with far-field coverage qualified by Amazon, the three-mic configuration is well suited for applications that may require user interaction



Here are some of the algorithm differentiators available with the ZLK38AVS Development Kit for Amazon AVS.

The ZLK38AVS Development Kit for Amazon AVS is available for $299.

One of the most convincing Alexa demonstrations at CES 2019 was available in the Microchip demonstration room, with the example of an already-available product—the latest model from RIVA Audio the "voice-enabled" Riva Stadium.

from any direction (e.g., portable speakers).

The development kit enables far-field speech recognition in the presence of interfering noise sources, including device playback and external noise sources. The field-programmable, field-upgradable solution features signal processing algorithms proven to improve both local trigger detection performance and cloud speech recognition accuracy, while the multi-microphone configurations include Direction of Arrival (DOA) estimation to indicate where the primary voice sound source is located.

The development kit includes a ZL38063 audio processor that connects directly to a Raspberry Pi 3B with plastics and mounting hardware to simulate a typical, recommended end-application mic-speaker arrangement. It supports all microphone options and includes an LED ring to report the detected sound location, as well as a high-quality speaker that is representative of a typical smart speaker application.

Microchip has audio labs and tuning facilities located worldwide to support customers from design through Amazon certification.

www.microsemi.com/product-directory/connected-home/4628-zlk38avs

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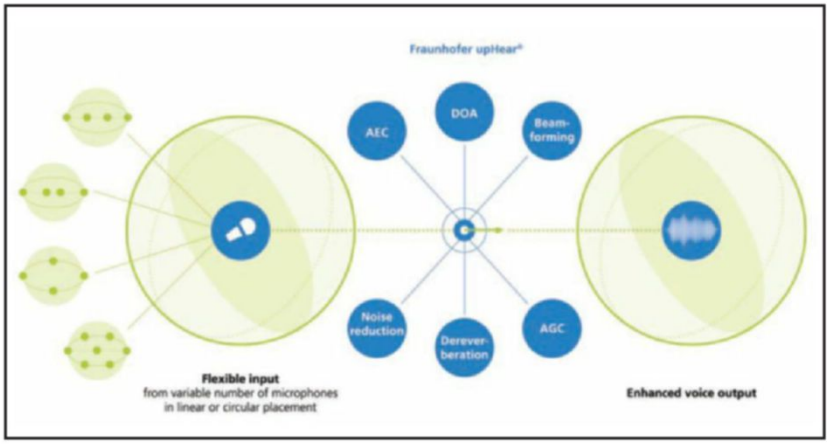
Yandex and Fraunhofer IIS Team to Bring Russian-Speaking Smart Speaker to Market

Yandex, a Russian technology company that builds intelligent products and services powered by machine learning, has incorporated Fraunhofer's upHear Voice Quality Enhancement technology into its first smart speaker, enabling the device to accurately hear voice commands issued from anywhere in a room. This is the first smart speaker powered by Alice, the first virtual assistant designed to serve the needs of Russian users, which was initially offered in Yandex's iOS and Android apps.

Fraunhofer's upHear Voice Quality Enhancement reduces background noise and acoustical echoes, ensuring that Yandex's new smart speaker, Yandex.Station and the embedded intelligent assistant Alice, can hear intelligible verbal commands, even when the speaker is streaming music at the same time.



Fraunhofer's upHear voice quality enhancement

Fraunhofer provided a software solution that applies advanced digital signal processing techniques to enhance the signals collected by the microphone array technology for far-field voice capturing, Yandex.IO.

Fraunhofer's upHear supports full-bandwidth audio and is a suitable voice enhancement step for a wide range of applications, including natural language understanding in mobile and smart home devices. It is a fully integrated and flexible solution which enables full-duplex conversations and barge-in support with outstanding audio quality. This is achieved by combining advanced multichannel source localization and beamforming techniques with echo- and noise-reduction algorithms.

Yandex's AI assistant Alice listens for and understands natural Russian-language requests and provides contextually relevant answers using Yandex's search engine and services that include weather, news, maps, transport and music streaming services. In developing Alice, Yandex built on its knowledge of the more than 50 million monthly users who interact with its services, plus its machine-learning capabilities and 20 years of experience with Russian users and the Russian language.

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Wave Behavior of Sound in Rooms

In this article, our columnist looks at wave effects and how they can affect sound in different rooms. Most computerized acoustical modeling programs such as EASE, CATT, ODEON, ULYSSES, and the several manufacturer-specific speaker-coverage modeling programs do not reveal wave effects, but it is important for a designer to account for wave effects.

By
Richard Honeycutt
(United States)

As you most likely know, sound consists of regions of alternating compressed and rarefied (decompressed) medium such as air. The individual air molecules making up the packets move back and forth, but the disturbance (the sound wave) progresses in a straight line—assuming that the medium is isobaric (all at the same pressure) and isothermal (all at the same temperature) —at a particular speed that depends upon the material in which the sound wave moves, as well as its temperature and humidity (if it is a gas such as air). Since the reciprocating particle motion is aligned with the wave motion, sound waves are classed as longitudinal waves, unlike waves in a pond or a stretched string, which are transverse waves.

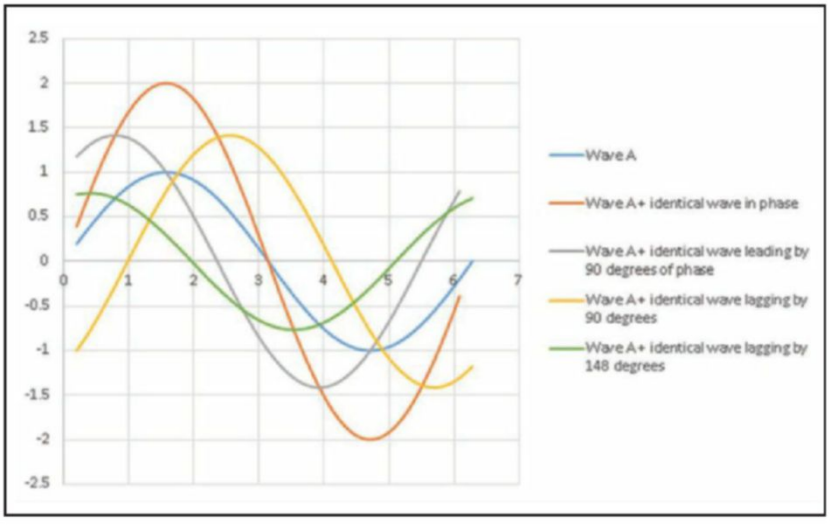

Figure 1: Interference results from the linear addition of waves.

Interference

When two sound waves affect the same point in the atmosphere, their effects add linearly; this is called interference. When waves are more or less in phase (both having compressions at pretty much the same place at the same time), the resulting wave has higher amplitude or sound pressure level (SPL). If they are more or less out of phase (one has a compression at the observation point when the other has a rarefaction), the resulting wave has lower amplitude. We call the first case constructive interference; and the second, destructive interference.

Figure 1 shows graphs illustrating the instantaneous pressure (vertical axis) plotted vs. time (horizontal axis) for "sound wave A" and several cases of interference. Wave A is represented by the light blue line. The grey and yellow lines show the slightly larger amplitude resultants caused by constructive interference between Wave A and waves leading and lagging (respectively) by 90°. The orange wave is the resultant of Wave A and another identical wave interfering in phase. Notice that it has a higher amplitude than the resultants or waves 90° out of phase. The green line represents the resultant of Wave A interfering with another wave 148° out of phase and, of course, it has a lower amplitude than Wave A by itself. Two identical waves exactly 180° out of phase will cancel completely, leaving no resultant.

Two (or more) sound waves impinging on the same point do not have to come from separate sources. Wave A could come from a loudspeaker, and Wave B could be the reflection of the sound

from that speaker, bouncing off a wall. This happens all the time in real listening spaces, and is one reason that it's difficult to make meaningful frequency response measurements of loudspeakers in typical rooms.

Modes

Sometimes, a sound wave travels from a source to one wall, reflects to the opposite wall, constructively interferes with the original wave, and keeps on reflecting back and forth, building up the amplitude with each new constructive interference. Such a wave is called a standing wave. Development of a standing wave requires a special relationship between the wavelength and the dimensions of the room. For example, if the length of the room is exactly one wavelength of the sound at frequency f1, sound pressure will build up at that frequency. We say that the room has an axial resonant mode at that frequency; in this case, a longitudinal mode, since it involves the long dimension of the room. Sidewall-to-sidewall and ceiling-to-floor modes are also axial modes, but side-to-side modes are also called transverse modes. Floor-to-ceiling modes are called vertical modes (see **Figure 2**). Modes can also involve reflections bouncing at angles off all four walls: such modes are called tangential modes (see **Figure 3**), or all six boundaries, including the ceiling and floor (see **Figure 4**). These are called oblique modes. The effect of modes is to make sounds of certain frequencies vary widely in SPL from place to place in the room.

Modal Frequencies

There are equations for calculating the modal frequencies in rectangular rooms. The frequency of each mode can be calculated from the equation:

$$f = \frac{c}{2}\sqrt{\frac{p^2}{L^2}+\frac{q^2}{W^2}+\frac{r^2}{H^2}}$$

where c = the speed of sound in air,
and p, q, and r are integer indices: 0, 1, 2, 3, etc., and L, W, and H are the room's length, width, and height, respectively. The speed of sound and the room dimensions must be in the same system of units: SI or Imperial.

For example, the lowest mode in a typical rectangular room is a longitudinal mode in which the room length equals a half wavelength. Since the side walls, ceiling, and floor are not involved, p = 1, q = 0, and r = 0. Thus, for this "1, 0, 0" mode, the modal frequency equation reduces to:

$$f = \frac{c}{2}\sqrt{\frac{1}{L^2}} = \frac{c}{2L}$$

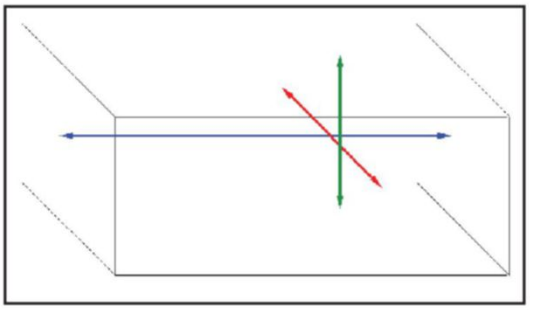

Figure 2: Axial standing waves can be longitudinal (blue line), transverse (red line) or vertical (green line).

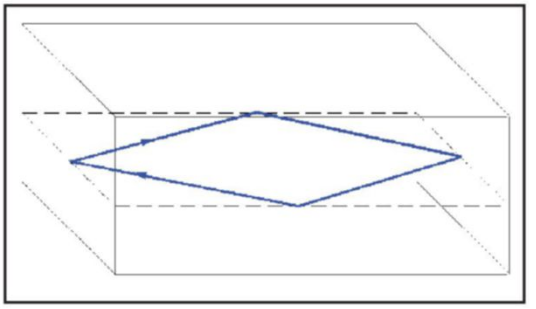

Figure 3: A tangential standing wave involves reflections from all four walls.

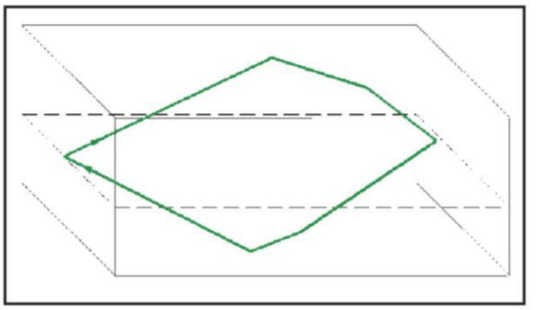

Figure 4: An oblique standing wave is formed via reflections from the walls, floor, and ceiling.

Using Imperial units, we can rewrite this equation as:

$$f = \frac{565}{L}$$

For a room whose length is 20′, the frequency of the lowest mode becomes 565/20 = 28.25 Hz. The longitudinal modes have q = 0 and r = 0. Their frequencies are, therefore, integer multiples of the frequency of the 1, 0, 0 mode. Transverse modes have p = 0 and r = 0. For vertical modes, p = 0 and q = 0. For tangential modes, r = 0, and for oblique modes, no index equals zero.

Figure 5 illustrates the modes of a 20′ × 13′ × 8′ room, as calculated by the amroc online room-mode calculator (https://amcoustics.com/tools/amroc?l=20&w=13&h=8&ft=true&r60=0.6).

You have probably noticed that there are only a few modes at the lowest frequencies, but more and more as we look higher in frequency. The number of modes per one-third-octave band is called the modal density. **Figure 6** shows the variation of modal density with frequency for the room whose modes are shown in Figure 5. Above a certain frequency, the modal density is high enough that the modal effects overlap and individual modal behavior is no longer a problem. This frequency is called the Schroeder frequency,

which is given by:

$$f_s = 2000\sqrt{\frac{RT_{60}}{V}}$$

where RT_{60} = the reverberation time of the room, and V = the volume of the room in cubic meters. For the room we have been examining, this frequency is 202 Hz, which is about the border between the two right-hand-most third-octave bands shown in Figure 6.

The Schroeder frequency of most residential listening rooms, and rooms of similar size, is between 100 and 200 Hz. In contrast, a 100′ × 80′ × 26′ auditorium would have a Schroeder frequency of 20.2 Hz, so audio-frequency modes would not be a problem.

We usually graph sound waves as pressure vs. time. But the velocity of the air molecules (particle velocity) in a sound wave also varies with time. However, particle velocity is much smaller than wave velocity c (the speed of sound), just as when you line up a row of billiard balls and strike the ball at one end, the disturbance moves rapidly through the row of balls, and it looks as if the ball at the other end responds immediately. The ball that was struck moves only a small distance and has a low velocity, whereas the disturbance (mechanical wave) moves very rapidly. In a sinusoidal sound wave, the particle velocity graphed vs. time is also a sine wave, but close to the sound source, it is displaced in phase from the pressure wave by 90°. Far from the source, pressure and particle velocity are in phase.

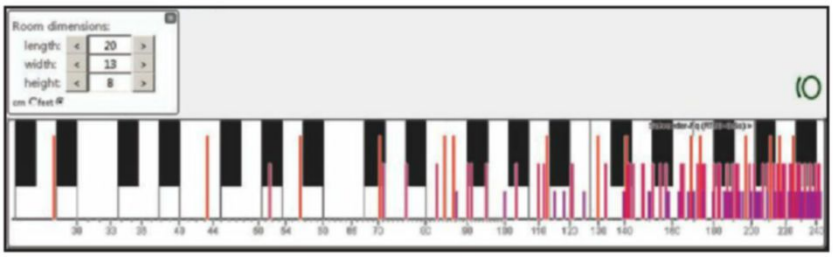

Figure 5 : Room modes in a 20′ × 13′ × 8′ room, from the amroc calculator (Image courtesy of amcoustics.com)

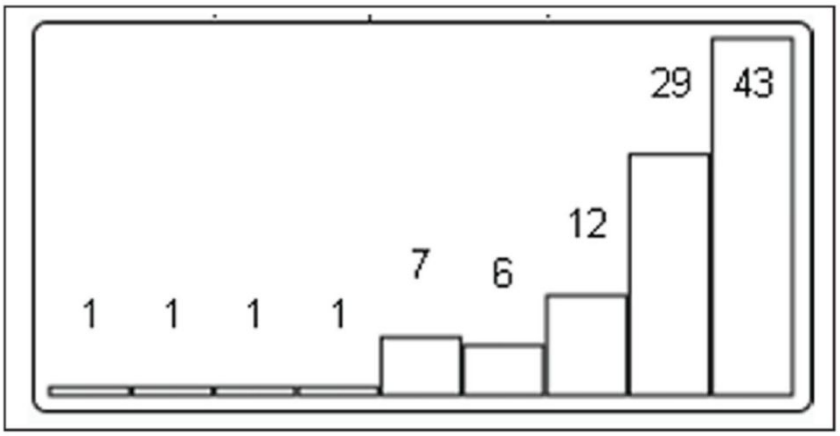

Figure 6: Modes per one-third octave band

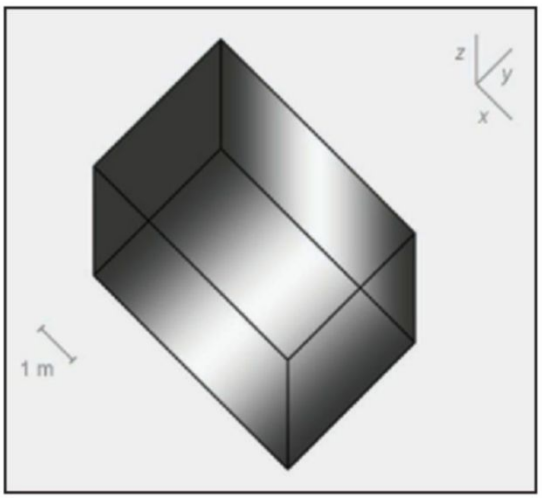

Figure 7: High-pressure zones at a modal frequency (Image courtesy of Dr. J. Hunecke, www.hunecke.de/en/calculators/room-eigenmodes.html)

Figure 7 shows the high-pressure zones caused by the standing wave at the lowest modal frequency (mode 1, 0, 0) for the 20′ × 13′ × 8′ room we have been examining. The dark regions correspond to high SPL. The location of the high-particle-velocity zones in comparison with the high-pressure zones will depend upon the source location. This can affect the optimum placement of bass traps used to mitigate modal behavior, since the absorption efficacy of a specific material will depend more on either pressure or particle velocity.

Reducing Modal Effects

Rectangular rooms are easy to draw, analyze, and understand. This has led some people to conclude that non-parallel walls will prevent the development of modes. While it is true that non-parallel walls will prevent flutter echo and the comb filtering that can result from it (causing a "boxy" sound in a small room), angling the room boundaries does not prevent modal behavior. **Figure 8** illustrates the effect of room shape (in plan view) on modes at two frequencies: 43.28 Hz (upper plots) and 71 Hz (lower plots), for a 20′ × 13′ × 8′ room. All the rooms have substantially the same volume and roughly similar overall outlines, but some have nonparallel walls, an "L" shape, or other features. All the rooms have hard ceilings and walls and thinly carpeted floors. No furniture or occupants are included. The modes illustrated correspond to the ones at 43.28 and 71 Hz—the third (0,1,0) and sixth (2,1,0)—modes shown in Figure 5. Note that room shape has more effect on the higher-frequency mode than on the

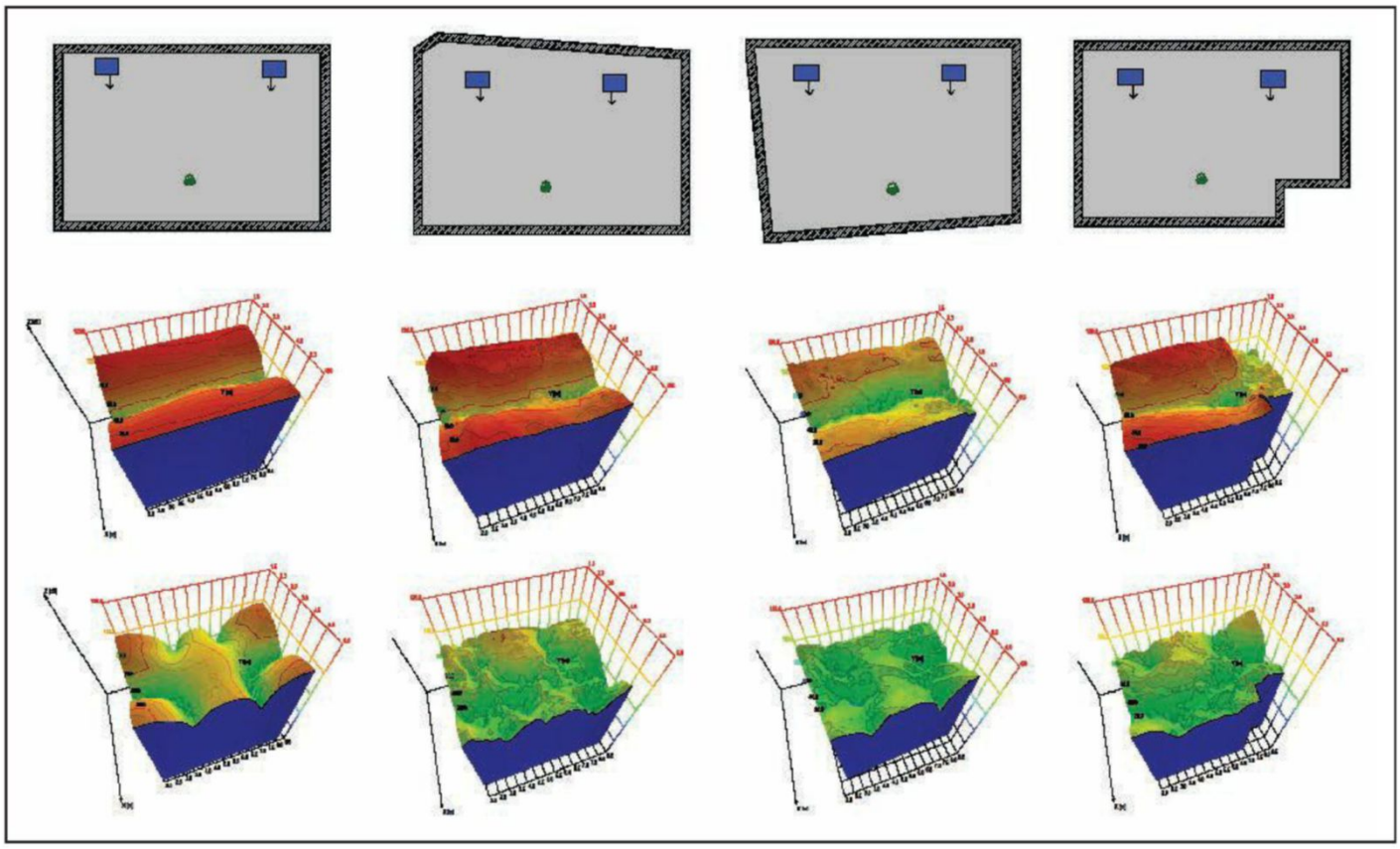

Figure 8: Effect of room shape on modal behavior

lower-frequency one but the shape variations do not eliminate either mode.

Square rooms or rooms with two dimensions the same (width = height, etc.) are subject to "modal pileup," in which multiple modes can be superimposed in the same location, worsening their effect.

The usual treatment to reduce modal effects is to place bass traps in the room. It may seem that using acoustical diffusion on the walls would help, but since the features of a diffusor must be an appreciable fraction of a wavelength (λ = 28.75′ at 40 Hz), an effective diffusor at modal frequencies will not fit in the room. A room large enough to accommodate a diffusing surface effective at 40 Hz would not have any significant modes that low, since the Schroeder frequency would be well below 40 Hz.

One other wave effect in room acoustics in auditoria is due to "seat dip effect." People seated in the middle of the audience area can experience irregular frequency response due to "seat dip effect," which is caused by diffraction and interference resulting from the sound passing over the quasi-periodic "hill-and-valley" topology of the audience members in front of them. Seat dip effect is more noticeable in rooms having a short early reverberation.

Most computerized acoustical modeling programs such as EASE, CATT, ODEON, ULYSSES, and the several manufacturer-specific speaker-coverage modeling programs do not reveal wave effects, such as modes in small rooms or seat dip effect in auditoria. A designer must remember to account for wave effects in some other way.

You Can DIY!

The Geek Columns

Learn how to build these unique "Geek Columns" and enjoy a full-range omni-directional sound. Because they are easy to build, they can be ready to enjoy in your outdoor space once the warmer weather arrives.

By
Ken Bird
(United States)

Photo 1: One of the completed omni-directional Geek column speakers is being used outdoors.

Outdoor speakers in gardens, patios, pool side, or porches have been popular for years and a number of commercial models are available for use. Most are only suitable for outdoor use and are intended primarily for background music use. This project features a stereo pair of columns that can be used outdoors in the summer and then moved indoors for use in a den or other room. The speakers produce full-range omni-directional sound from inside these unique enclosures that are easy to build. You can elect to build only one column for mono outdoor sound or two columns for stereo.

The enclosures are 44" tall, 10" in diameter Greek-style plastic columns that can also be used for trade show props, weddings, or other decorative applications. The 1/4" walls are very rigid and when properly weighted and stuffed with polyester fill, they make a fine infinite baffle for an 8" speaker. I named them the "Geek Columns," in salute to their unusual use as speaker enclosures (see **Photo 1**).

Photo 2: I chose the Goldwood GW8004 coaxial speakers for the columns.

The Driver

The driver I selected is the Goldwood GW8004 full-range 8" model (see **Photo 2**). The polyester cone and rubber surround make the speaker useful for harsh environments. The GW8004 has a smooth response from 35 Hz to 20 kHz with a coaxial neodymium tweeter. The driver's high 0.9 Qts makes it ideal for the infinite baffle formed by the plastic columns.

Preparing the Columns

This particular pair of columns was molded with a closed top where another mating column could be screwed to it make an 88" column (see **Photo 3**). They were manufactured by Events Wholesale, Inc. (www.eventswholdsale.com) and come in a variety of heights and types. I found the pair that I used for this project at a garage sale and I learned that they had once been used as part of a trade show booth.

I measured the screw portion at exactly 8" and cut out the top using a Jasper router jig, to 8 1/4". This revealed the hollow interior and square base. The base had a quarter-sized hole in the bottom that I sealed up with a piece of the plastic from the top cut outs using epoxy cement. Next, I wired the rectangular banana jacks and mounted them on the column base (see **Photo 4**). I secured the wire at the

top and added 15 lbs of clean sand to the base to provide better stability. Then, I stuffed the columns with 16 oz of polyester pillow stuffing. I made sure to tease the polyfill to reduce the density and I filled the columns to within 6" of the top.

Building the Sound Diffuser Housing

I constructed the sound diffuser from 3/4" birch plywood and plastic wedding cake support columns. The diffuser spreads the mids and highs, while providing a measure of protection for the speaker. **Photo 5** shows all the parts except the 4-20 steel threaded rods.

I used the Jasper router jig to cut the parts. I began construction by cutting out a 12" × 12" piece for the base. I cut an 8 3/4" hole in the center of this piece and rounded the outside edges using a router bit. I also saved the piece of 8 3/4" cut out material and rounded the edges. This will serve as the final top piece.

Then, I cut an 11.5" circle with an 8.75" hole in the center. I did not round the edges of this piece. Next, I cut out an 11.5" circle for the top piece. I clamped the base, the center circle, and the top together. I used a 3/8" drill bit to drill four holes equal-distant in the base, up through the circle piece and just 1/8" into the top. Use of a drill press will make the accuracy of these holes better.

With the holes just drilled as a guide, I used a 3/4" Forstner bit to drill into each hole in the base to 1/2" depth and to each hole in the underside of the top, 1/2" deep. I cut an 8" Styrofoam ball in half and painted it with acrylic white paint to form the diffuser. Once dry, I centered the ball to the bottom of the top and cemented it in place using a no-oil based glue (e.g., Tacky Glue).

The holes in the top will receive the 4-20 "T" nuts. However to assure a good grip in the plywood, it was necessary to first apply some expanding Gorilla Glue into the holes then hammer in the "T" nuts.

I then cut four pieces of 4-20 threaded rod 7 1/8" long. I inserted and screwed the rods into the "T" nuts in the top. Then, I slipped the plastic wedding cake columns over each rod. I lined up the plywood ring and slipped it over the rods. I did the same with the base and then added the washers and nuts on the rods. After that was completed, I lined up the top of the plastic columns so the flat top side with decoration could be seen. Finally, I carefully tightened the rods to complete the assembly (see **Photo 6** and **Photo 7**).

Painting the Columns

First, I sprayed the top section housing—the sound diffuser—with auto primer and then I resanded and recoated any rough spots. Next, I applied two coats of exterior semi-gloss house paint. I elected not to paint the columns as I liked the appearance of the white plastic and they didn't need any protection from the elements.

Photo 3: The plastic Grecian columns used for this project are typically used as props in trade show booths or wedding celebrations.

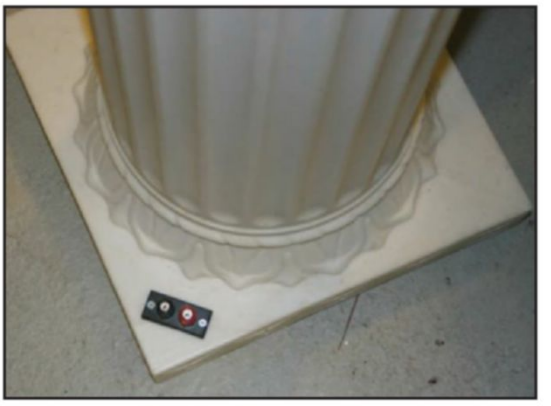

Photo 4: Here is the installation of the speaker terminals in the hollow column base.

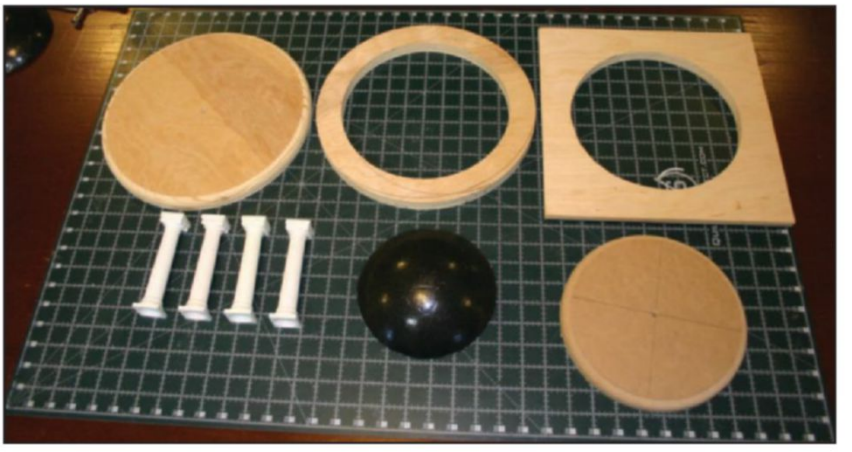

Photo 5: I used these parts in the construction of the sound diffuser housing (aka the "wedding cake top").

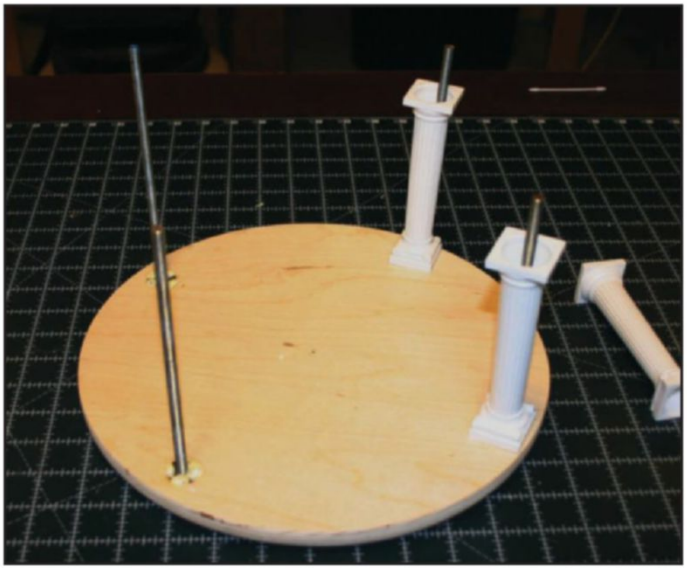

Photo 6: To hold the top together, insert the 4-20 threaded rods into the "T" nut in the top and installing the mini-columns.

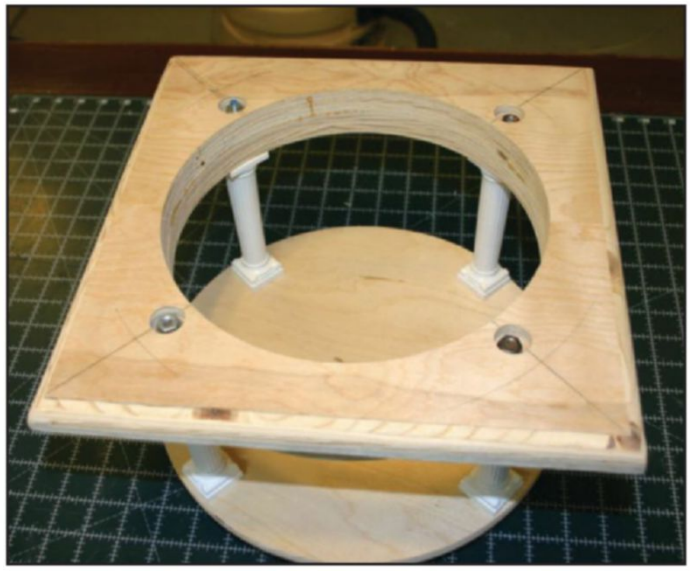

Photo 7: Here is a bottom view of the diffuser housing showing the nuts holding the assembly together.

Photo 8: One of the final steps is to mount the sound diffuser on the column.

About the Author

Ken Bird has had a 40-year engineering and product management career focusing on consumer, industrial, and telecommunications electronics. Ken retired to his native Missouri and now pursues his interest in building speakers and tube amplifiers, along with model railroading, amateur radio (WOKLB), and digital photography.

Parts List for One Speaker (Double for a Pair)

1 each plastic column 48" high, 10" in diameter, (www.eventswholesale.com)
4 each Wilton Wedding Cake Grecian Pillars (www.wilton.com/shop-cake-accessories)
1 speaker, Goldwood Model GW 8004 (Parts Express #290-378, www.parts-express.com)
1 speaker terminal (Parts Express #260-301, www.parts-express.com)
1 each 2' × 4' 3/4" plywood for diffuser housing parts
1 8" Styrofoam ball cut in half (local craft store)
1 each 4-20 threaded rod 36" long
4 each 4-20 nuts and washers to fit rods
72" two conductor 18-gauge speaker wire
15 lbs dry sand
1 roll 1/4" foam insulation

Wiring and Speaker Mounting

For the next step, I soldered the speaker wire, noting the proper polarity. After applying a length of 1/4" foam insulation to the rear of the speaker mounting lip, I mounted the speaker directly to the lips of the columns with a 1 1/4" dry wall screw. A 1/6" pilot was first drilled to guide the screws.

Mounting the Diffuser Housing

The diffuser housing is not attached to the column, I found that its weight allowed it to be merely placed on the column top. This makes it easier to maintain the speaker. The housing has withstood a number of summer thunder storms remaining upright with the housing intact. For those who want additional protection from the elements for the speaker, a plastic covering can be temporarily placed over the top if rain is in the forecast. This can also discourage birds from trying to build a nest in it in the spring (see **Photo 8**).

Sound Quality

For the best sound quality, I would recommend that you place the speakers in your garden or on your deck about 12' apart. Outdoor speakers are usually used for background listening and not typically played very loud, but the Geeks can handle volume levels that could aggravate your neighbors. Use common sense when listening to them outdoors. The Geek Columns can also be brought indoors for use in your sun porch, family room, or basement man cave for the winter. They produce a fine omni-directional sound perfect for any type of music you enjoy.

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Comply Foam
Computer Audio Design
Connection
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Luxman
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Pure Audio Project LTD (Israel)
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Western Electric
Wolf Audio Systems
Wynn Audio
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ZMF Headphones

Exhibit list as of 2/6/19. Visit www.axpona.com for an updated list

Industry Calendar

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

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

"Musikmesse and Prolight+Sound 2019: Stronger together!" is the slogan used by Frankfurt Messe to promote the 2019 edition of these two prestigious shows, which could risk being the last ones if the new model doesn't work. After many years of forced separation, the Frankfurt show complex will finally open the new Hall 12 (two levels), enabling Prolight+Sound to grow into a modern space, while Musikmesse returns to Hall 3. More important, both shows will take place again concurrently! The Live Sound Arena—a key area for Prolight+Sound—is being relocated to outdoor area F10 in the immediate vicinity of Audio Hall 8.0. Additionally, there will be several special areas where companies can present their products to specific special-interest groups of visitors. With a new sequence of days—from Tuesday to Friday—the shows can strengthen their profile as a platform for the professional exchange of ideas and information between music, audio, and entertainment-industry professionals.

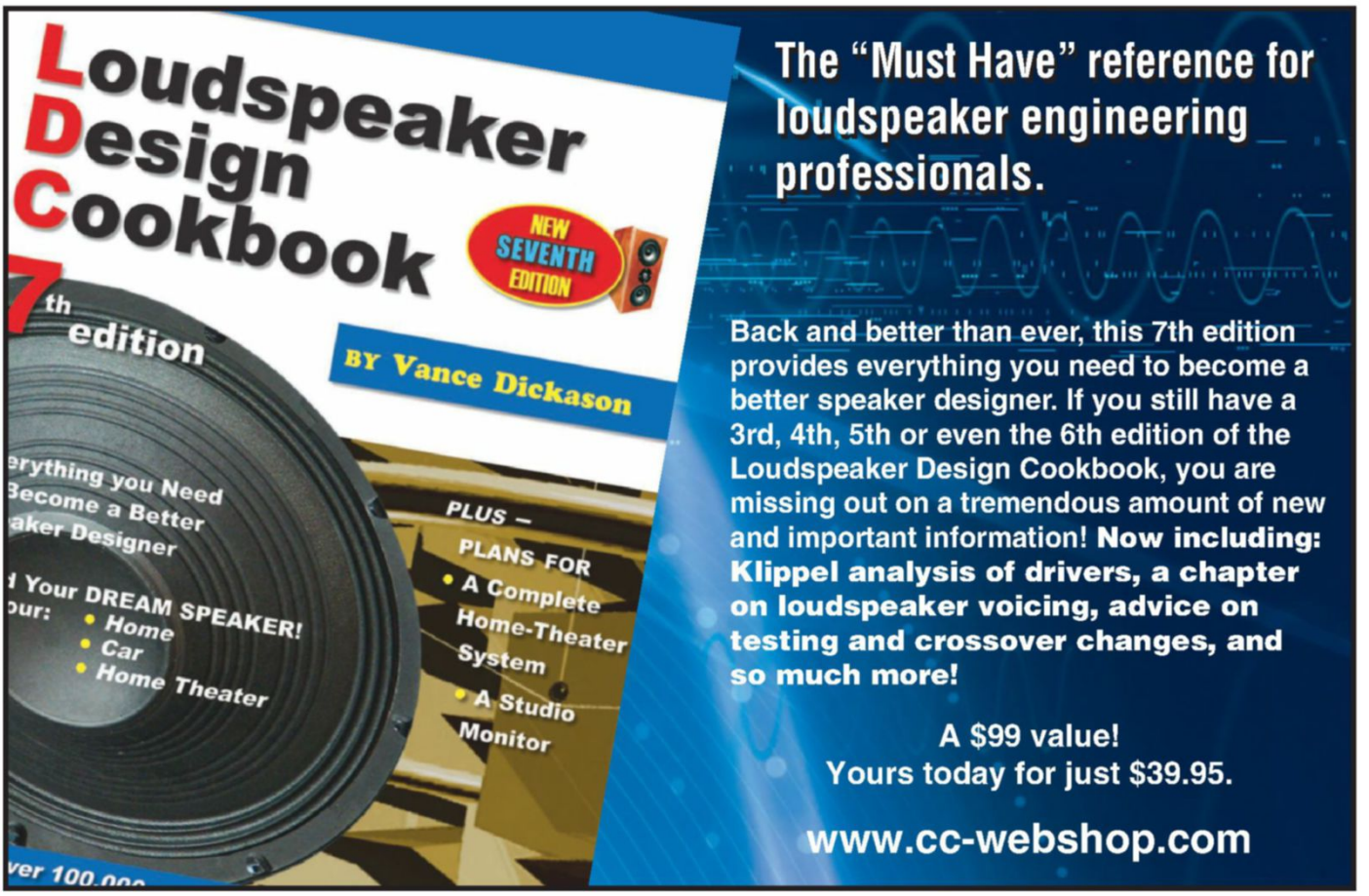
April 8–11, 2019
NAB Show
Convention: April 6-11
Las Vegas Convention Center (LVCC), Las Vegas, NV
www.nabshow.com

From media entertainment creation to distribution to consumption and technology, across multiple platforms and countless nationalities, NAB Show is where global visionaries convene to bring content to life in new and exciting ways. The annual Las Vegas gathering is represented by professionals in advertising, app development, artificial intelligence, audio, augmented reality, broadcast, cable, cloud solutions, cybersecurity, digital video, digital signage, eLearning, Esports, film, game development, government and military, houses of worship, in-vehicle entertainment, IOT, IT, live events, mixed reality, mobile, online video, podcasting, post-production, radio, retail, social media, sports, streaming, system integration, television, virtual reality, 5G, and more. The Show Floor spans more than one mile and contains hardware, software, and all of the latest technologies from global brands and cutting-edge start-ups.

April 12-14, 2019
Audio Expo North America (AXPONA)
Renaissance Schaumburg Hotel & Convention Center, Schaumburg, IL
www.axpona.com

Audiophiles, music lovers, vinyl collectors, audio engineers and musicians, will join 500 manufacturers in 185+ listening rooms, and Live Music events, for the 2019 edition of Audio Expo North America (AXPONA). Three jam-packed days of high-quality sounds, technology, presentations, design, and the very latest products in the industry.

Now in its 10th year, AXPONA is competing to be the largest high-end audio show in North America, attracting an increasing number of dealers, distributors and manufacturers, as well as music-loving consumers. AXPONA's venue will feature listening rooms, an Expo Hall featuring the Record Fair and Master Class Theater, a new hall dedicated to the headphone-oriented Ear Gear Expo, a robust seminar schedule, and Live Music nightly. Attendees are encouraged to experience the listening rooms and compare the high-performance systems.



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