

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Market Update

True Wireless Earbuds Evolving Toward Hearables

By J. Martins

R&D Stories

The Evolution of True Wireless Stereo

By Jean-Marc Luneau
(7 Sensing Software)

Market Update

Voice and Communication

R&D Stories

Speaker ID Technology

By Vikrant Singh Tomar
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You Can DIY!

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Audio Scope

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By Michael Steffes

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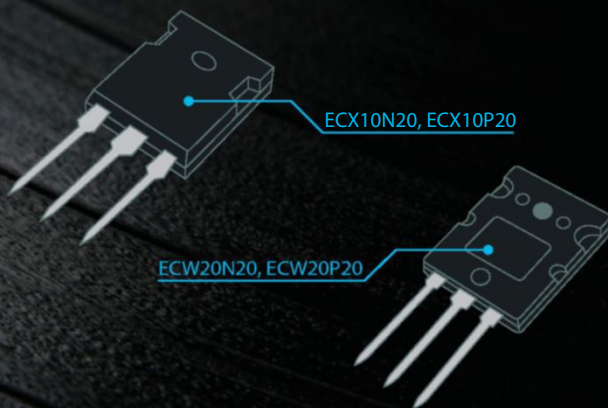
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Enhanced Audio Quality Frontiers

This edition of *audioXpress* offers a glimpse into an extremely important transition that is currently pushing the entire audio industry forward. In two complementary Market Update articles, we've tried to cover the technology and development platforms for true wireless stereo (TWS) earbuds and hearables, as well as exploring examples in voice recognition and transcription (voice-to-text) and voice communication. There are multiple product and market segments for these emerging technologies and that is why we devoted two Market Update features, even if the underlying technologies are converging.

We are extremely honored to have two exceptional contributed articles from two relatively new technology companies in the voice and DSP space, addressing these emerging technologies and diversity of applications. First Jean-Marc Luneau, an audio DSP engineer and co-founder of 7 Sensing Software, discusses expanding voice communications and the frontiers of augmented audio in TWS earbuds. And Vikrant Singh Tomar, the founder and CTO of Fluent.ai, explains how his company is taking voice recognition even further by focusing on user recognition, personalized responses, and even addressing multilingual challenges. Both articles perfectly illustrate the exciting perspectives for their respective fields of research.

It was tempting to move full speed ahead and address the cutting-edge areas of voice engines and conversational AI, machine learning and artificial intelligence, and all that is propelling these disciplines in audio. We also had to resist the temptation of digging deeper into fascinating areas, such as advanced audio processing for active noise cancelling, exploring all the perspectives in spatial audio applications, and the potential implications in augmented reality and automotive applications, but we will address these topics in future editions.

All these new technology advances are moving the audio industry into a remarkable expansion, now employing more people and attracting more professionals to audio than ever. In 2019 that expansion was already clear, but it was remarkable to see how it has accelerated throughout the pandemic, with these new product categories now leading in consumer electronics. TWS earbuds and voice-enabled products (including smart speakers) are selling more units per month, quarter, and year, than practically all remaining audio product categories added together. The global smart speaker install base is forecasted to reach 640 million by 2024 (Canalys), while the number of TWS earbuds and hearables will grow to more than 970 million also by 2024 (Juniper Research).

As the mass adoption of wireless earbuds and smart speakers reaches new heights—precisely at a time when people's mobility is severely restricted—those concepts are just going to evolve and expand when we are able to return to traveling and freely visit public spaces again.

While being forced to stay at home longer, we all had to adapt to using new communication tools and create new ways of socializing and networking. And those new habits will carry on with us in the next few years, as we transition to new hybrid working environments and work from anywhere.

Certainly, no one wants to go back to offices as they were before, or to spend more time at home as mobility opens up. Which means taking our new technologies—that helped us all get through the pandemic—to new environments as we embrace more remote work.

In any given circumstances, I am confident that most people will now be more discerning regarding audio quality, and how technology delivers the impression of being closer together, no matter where we are. We will all be less willing to accept restricted quality in return for convenience or reliability.

The next few years will be about taking audio technology to new levels of quality, mobile, wireless, everywhere. Maybe just not defined according to former standards.

To design those audio products, countless more professionals will be involved. A new generation that believes as strongly in great audio quality but are now focused on working on completely new product categories and form factors. And while we have extremely talented people involved in advancing these emerging technologies, we cannot lose track of the essential audio knowledge. Now more than ever, the audio industry needs to engage the experience of audio industry veterans to help this growing number of professionals push their efforts forward, without the need to repeat past mistakes. We need to familiarize a new generation of professionals with the accumulated audio knowledge, and it is particularly important to involve people with a living memory of past research efforts. And that makes our work at *audioXpress* feel more important than ever.

J. Martins
Editor-in-Chief



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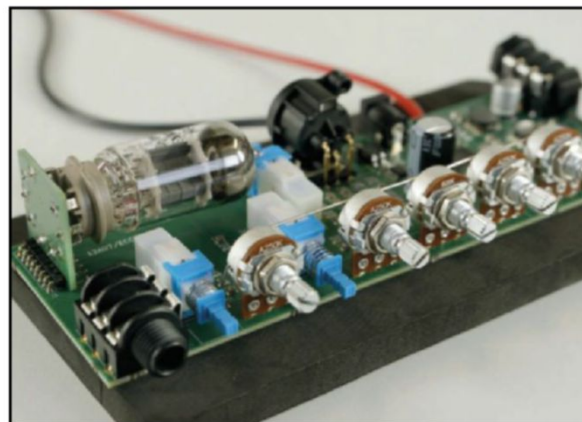


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The Research

By George Ntanavaras

Several years ago, George Ntanavaras designed a system consisting of the ACH-01 accelerometer and an amplifier with an integrated analog signal processor. Over the years, he has been looking for other possible solutions concerning the vibration measurements. In this article, he describes his research for the design, the construction of such a system, and presents the results and the measurements.

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Considering the IC Options

By Mike Meechan

In the previous two articles, Mike Meechan discussed the fundamental physics that play a part in

microphone preamplifier design, and briefly explored its inhabitance in the practical world and how a good design can retain the fidelity of the mic signal along its path from transducer to output. At this stage, he considers constructing some of the theoretically-perfect building blocks of an active mic preamp.

62 The Best of Two Worlds An Old-Fashioned Tube Amp and Multilevel Class-D Hybrid DIY Bass Guitar Amplifier Project

By Jens Tybo Jensen

This article describes the practical and basic electrical design of a DC or battery-powered, 60W, professionally featured and ultra-efficient bass instrument amplifier. It is modelled after a classic vacuum-tube bass amplifier topology, while utilizing the exceptional audio quality and best in class efficiency of the Infineon MERUS amplifier technology to amplify every nuance of a genuine vacuum-tube preamplifier.

COVER ART

Photo Credit:
Melomania Touch true
wireless stereo earbuds

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● Columns

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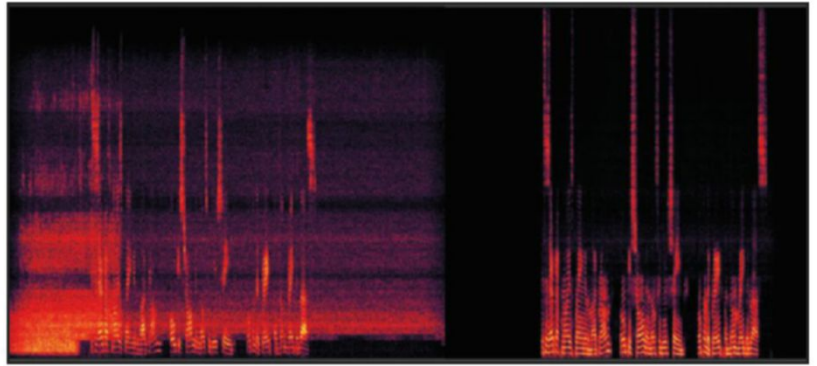
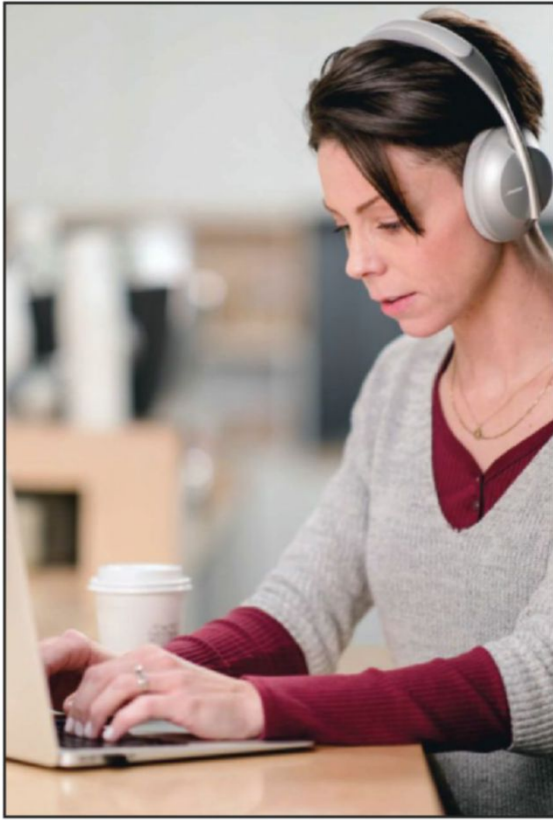
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This market update article explores the exciting convergence of voice and speech-to-text technologies, now empowered by artificial intelligence and machine learning to enhance audio communications: A crossroads perspective of cutting-edge innovations and opportunities.

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By Vikrant Singh Tomar

Increasing consumer demand for personalized experiences has sparked the creation of speaker ID technology. This article explores the challenges that this type of technology still faces.

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● Market Update—True Wireless Earbuds

22 True Wireless Earbuds Evolving Toward Hearables

By J. Martins

There's a lot happening in the true wireless stereo (TWS) category and this market update article intends to provide a quick overview of the state-of-the-art technologies and platforms in the space, as well as some reference approaches in product design.

30 The Evolution of True Wireless Stereo

By Jean-Marc Luneau

In this article, we will look at some meaningful new features and improvements that can help True Wireless Stereo (TWS) earbud brands differentiate their products and make them stand out in today's crowded market.



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Unifying Communications with Amplified Intelligence



By
J. Martins
(Editor-in-Chief)

This market update article explores the exciting convergence of voice and speech-to-text technologies, now empowered by artificial intelligence and machine learning to enhance audio communications: A crossroads perspective of cutting-edge innovations and opportunities.

In just two years, extraordinary developments in the field of voice capture and recognition have completely changed the technology landscape. *audioXpress* has been covering that ground research in specific disciplines, from microphones and smart speakers to voice interfaces and audio communications.

For this market update report we have looked at recent developments from a different angle: How the evolution in the field of voice capture for voice recognition products is vitalizing communications and specifically Unified Communications & Collaboration (UC&C), a market segment that was abruptly thrown into disarray at the start of the pandemic. Sudden changes in people's behavior and the forced closure of corporate spaces have affected a thriving market segment, but there might be a positive consequence. The pause is actually enabling time for some convergence to catch up with the intense research that was taking place in microphone technology, microphone arrays, beamforming techniques, audio signal processing, and algorithms—all fostered by “smart” consumer products promising the ability to translate human voice commands.

The Pandemic Effect

The UC&C market was already promoting the potential evolution with AI and machine learning technologies, but was mainly concerned with the need to converge with IP and networking standards and too focused on corporate-centric solutions and trying to distance itself from consumer trends.

As I already noted in a previous article, it was increasingly impossible to ignore the bring-your-own-devices (BYOD) office trends, and not to notice the divergence from the focus on large investments for centralized management and control solutions,

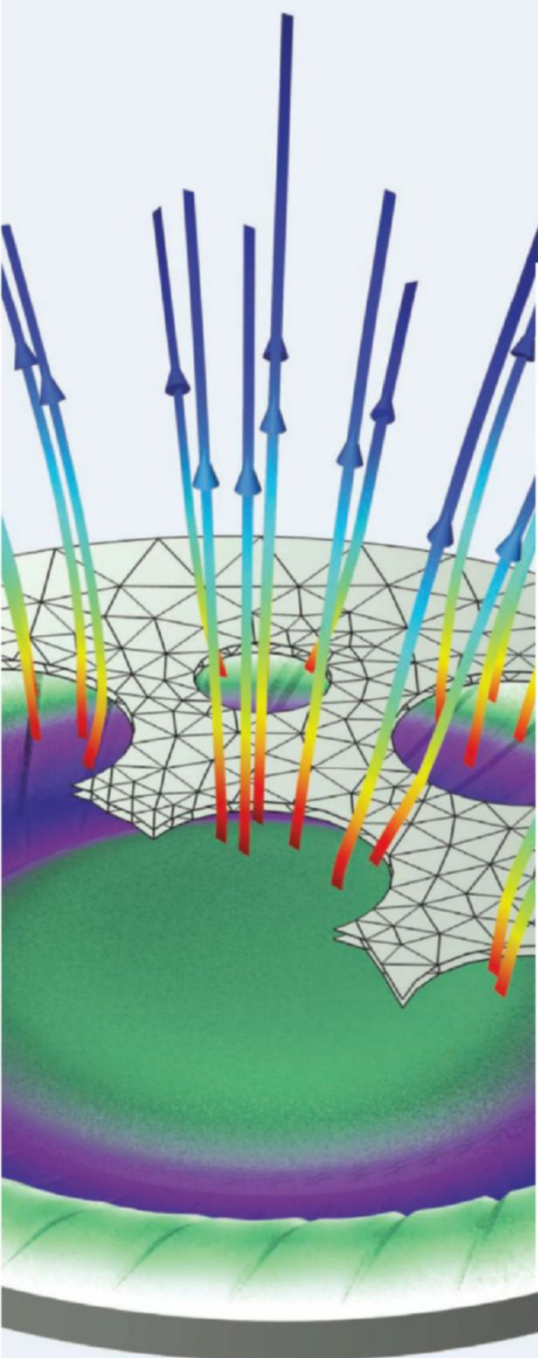
over integration of web tools, mobile devices, and the fast-evolving Bluetooth and USB interfaces.

The pandemic changed all those priorities and brought about the urgency to connect users working from home, using web and mobile tools, with professional-level communication solutions. And nothing better shows how powerful that change was than the sudden emergence of Zoom Video Communications as a leading web conferencing player. Anyone following the UC&C space was familiar with Zoom, as an actor playing side by side with companies such as BlueJeans and competing with Microsoft's own efforts in the remote collaboration space, evolving Skype for Business into Teams. Suddenly, Zoom became the leading player in the global communications market. Now, Zoom is already refocusing its strategy to help organizations safely re-enter the office and sustain an “everywhere workforce.” There's an understanding that traditional workplaces have evolved and adapted with employees and organizations embracing more places as workspaces—whether in an office, co-located space, remote location, or at home.

And leading the change are educational institutions, healthcare facilities, and governments, which have traditionally trailed behind private corporations. Critical to those changes is the ability to pair mobile devices, connect health monitors and environmental sensors, and support contactless solutions for access and control by adopting voice interfaces.

All hardware solution manufacturers in the UC&C space are quickly adapting their existing systems, while they are working on introducing a new generation of “hybrid” solutions that support this new (more demanding) environment.

And it's not only about communications and collaboration. The consequences of this technology evolution, in hardware and software,



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The Meeting Owl Pro from Owl Labs, an all-in-one 360-degree camera and speakerphone

is also fostering new opportunities for other traditional voice-based applications, such as speech-to-text processing, transcription, and translation, which platforms such as Zoom, Slack, and many others are starting to integrate in their basic services.

The next stage promises to accelerate the evolution in voice processing for all these applications even more, with machine learning (ML) and Artificial Intelligence (AI) now transferring directly to endpoint devices (or edge devices, if you will)—taking sophisticated processing even closer to the source.

Another factor in the way things are progressing faster is related to the technology adoption in new categories of “smart” products. Speaker drivers optimized for compact, low power solutions, are now generally available, with new speaker design topologies promising exciting possibilities; microphones are also getting much better due to the fact that they are directly integrating with smart amplifiers and processing; and software and signal processing is increasingly more powerful and sophisticated.

It also helps that AI on the edge enables users to separate speech from other sounds and background noise, allowing systems to process signals exactly according to the application requirements—be it for security, entertainment, or communications.

Front-End Perspectives

UC&C solutions usually include the integration of a variety of communications and collaboration tools including calendars and scheduling, email and voicemail, messaging, and web/audio/video conferencing.

Currently, voice recognition, voice digital assistants, voice-to-text, and text-to-voice technologies are disrupting all those collaboration tools. In fact, Amazon promotes its Alexa-powered devices precisely to do all the above. Using Alexa, users can dictate emails and instant messages, while incoming emails and text messages can be read aloud. And Apple has long promoted those same capabilities in iOS/iPadOS, now also possible with Siri’s advanced integrations in macOS. Even for video-based communications, devices such as the Amazon Echo-Show, Facebook Portal Plus, or the Meeting Owl Pro, all have cameras that combine image processing while tracking the voice direction to automatically pan and zoom, keeping users

in frame while they move and talk freely.

The Meeting Owl Pro from Owl Labs, is an example of a new all-in-one device, combining a 360° camera with microphones and speaker, originally designed for meeting rooms. Now, it is being adopted for online classes and tutoring, and generally for remote meetings, offering more freedom of movement to users, while delivering an easy-to-use solution that integrates with all conferencing platforms. The unsolved hurdle in Owl remains the fact that the microphone is only able to work well in ideal acoustics and at close proximity to the user. Also, the speaker is optimized for voice and doesn’t compare favorably with any of the smart speaker products from Amazon, Google or Apple—all those products are now also used for music listening, not just calls.

As an example of future product concepts, we can imagine an Owl camera, with its clever 360° capabilities and image tracking intelligence, combined with something like the Noveto Systems’ recently introduced Noveto SoundBeamer 1.0.

Noveto is an Israeli company with an impressive acoustic beamforming technology that is able to create a personal soundfield. At CES 2018, Noveto demonstrated doing a Skype call on a computer right in the middle of a busy show floor, and we could hear perfectly the other person on the call without using headphones, while a person 2’ away couldn’t hear anything.

Noveto’s latest AI chipset solution is now ready and the company has announced a proof-of-concept speaker product, the SoundBeamer 1.0. The SoundBeamer 1.0 all-in-one compact desktop device uses Bluetooth to pair with any device—computer, tablet, smartphone or TV—in order to create a personal audio zone, ideal for web conferencing, gaming, or watching any type of content without disturbing others in the vicinity. Noveto’s proprietary multicellular piezo-electric transducer array converts audio signals into non-audible acoustic waves, which beam audio to just outside the ears, creating the personal audio experience. For this, the SoundBeamer 1.0 also features face detection, tracking capabilities and state-of-the-art DSP engines. The direction of the audio beams is optimized by a 3D sensing module, which locates and tracks the position of the user’s ears. This, as *audioXpress* could already attest in 2018, effectively generates a good-quality soundfield, with a very strong attenuation nearby (–20dB).

This means that, for home communications, entertainment, and inside the car, personal audio zones can be created without the need for headphones. And this concept can certainly benefit any type of voice recognition applications, including detecting multiple talkers in a room.

Speakers and Microphones

In the UC&C space, companies have long recognized the need to provide far-field voice capture with audio processing for basic echo cancellation and noise reduction (at least), integrated within a good quality speaker that is able to provide an integrated solution for any conference space.

As mentioned, this is an area where all the research that is happening to smart speakers or any type of device that supports voice assistant applications is benefiting from any design that needs to optimize voice capture.

A great example is Nureva, the fast-expanding Canadian audio conferencing solutions company, which is gaining wide industry recognition with its true full-room coverage in virtually any space configuration for standard meeting rooms. The company's approach is based on its Microphone Mist technology, a patented concept of generating thousands of virtual microphones in a 3D space, which was already detailed in several articles published in *audioXpress**.

Nureva's HDL200 supports up to 4,096 virtual microphones providing full-room coverage in any space up to 18' x 18' (5.5m x 5.5m) and it installs easily almost anywhere—on a wall, a floor stand, or a display. Powered by Microphone Mist, the system calibrates automatically and continuously and in-room participants can be dispersed for safety protocols or move around and still be clearly heard by remote participants. The technology also provides intelligent sound targeting to eliminate unwanted sounds, making it an ideal match for today's hybrid meeting and learning spaces. The product line includes the HDL200 for small rooms, the HDL300 for mid-size rooms, and the Dual HDL300 for large rooms.

Another example of a company leading in the field of voice capture for communications is Yamaha. Yamaha has been working on communications and related technologies for several years. Adding to Yamaha's own research efforts in DSP and digital variable acoustics, in 2014 the company acquired Revolabs, the pioneer in wireless microphones and presentation technologies. Shortly after, the first member of the current YVC Speakerphone family, the YVC-1000, was introduced.

Following the introduction of the Yamaha YVC-200 Portable USB + Bluetooth Speakerphone in 2018, Yamaha's Unified Communications division gradually increased its range of professional-quality portable conference solutions, including new YVC Speakerphone models, the CS-700 Video Soundbar and further conference room speaker and microphone solutions. Since 2020, Yamaha has been refocusing its powerful range of communication products to respond to the more diverse needs of work from home and remote collaboration.

The Yamaha YVC-200 portable speakerphone, originally designed for a very distinct envisaged scenario of hotel rooms, co-working spaces, and huddle rooms, is now proving its worth as individual users started buying them to work from home. The speakerphone features Yamaha's professional quality DSPs optimized for voice, with a 360° and maximum radius of 6.6' (2m) microphone pick-up range, and a rechargeable battery for up to 10-hours of use. Most people simply use it connected and powered by the computer USB port, but it also quickly pairs and connects to phones and tablets via NFC and Bluetooth. The YVC-200 speakerphone also serves to demonstrate the power of sound processing technologies, including adaptive echo cancellation and Yamaha's own Human Voice Activity Detection, which focuses on speech rather than background noise. The unit intelligent microphone processing delivers full-duplex conversation without sound dropouts, even when multiple people are speaking at once.

The YVC-200's only limitation is the small loudspeaker, limited in volume, but still effective for voice communications. And users love the mute button, which is directly supported by popular conference apps such as Skype, Slack, and Zoom.

More recently, Yamaha reinforced its efforts in UC audio with the latest YVC-330 Portable USB & Bluetooth Conference Phone



Noveto's proprietary transducer array converts audio signals into non-audible acoustic waves, which beam to just outside the ears, where the sound can be heard again as normal acoustic waves, creating a personal audio experience.

with SoundCap technology, and the CS-700 Video Soundbar family. Both are better at dealing with background noise leveraging far-field noise reduction, while SoundCap zeros in and captures only the voices of speakers sitting around a meeting table. The larger speaker also offers better voice reproduction with auto-speaker volume adjustment and a superior microphone array for more users.

This is precisely the level of integration that many users have learned to recognize in Apple's MacBook laptops. Apple always combined the microphone and speaker integration in its MacBooks with powerful processing algorithms that intelligently manage communications in the background, without requiring any user intervention. Of course, neither the MacBook speakers nor the built-in microphones could compete with Yamaha's dedicated designs and embedded DSPs, and the benefits of using the YVC speakerphones are evident.

Probably the largest problem that Yamaha faced in the market was unfair comparisons with cheap USB and Bluetooth speakerphones that many computer accessories manufacturers sell on Amazon, which degraded the potential for the product category with earlier



The Yamaha YVC-200 speakerphone is proving a valuable solution at home, leveraging integrated sound processing technologies, including adaptive echo cancellation and Yamaha's own Human Voice Activity Detection, which focuses on speech rather than background noise.



Bose Professional launched its Noise Cancelling Headphones 700 UC, a variation of the existing Bose 700 consumer headphones, certified for Microsoft Teams and compatible with both Zoom and Google Meet, for home office and working remotely.

cheap designs that didn't even match what a professional laptop could do. As many users working from home are now discovering, speakerphones from Jabra, Poly, beyerdynamic, Shure, Sennheiser, and other brands that launched updated designs in the last two years, are a completely new breed, and highly effective. And they will improve even more dramatically in the next two years, with USB-C, Bluetooth 5, low-power processors, and much better microphone arrays and software algorithms, now starting to use AI on-device.

Headsets Getting Smarter

Meanwhile, the product segment with faster progress benefiting from the convergence of voice technologies are headphones/in-ears. In this same edition, we are reporting on the remarkable progress happening in the true wireless space, which also includes optimizations for using voice personal assistants. Not surprisingly, true wireless stereo (TWS) earbuds are currently the fastest growing consumer product segment, beating records in volumes shipped and quickly becoming the dominant form-factor in personal audio.

But things are looking good as well—even if on a different scale—for all remaining headphone categories. As Steve Nason, Research Director at Parks Associates detected in its research, "Everyone in the household now needs their own headphones and earbuds for privacy during this time of shelter-in-place orders and work-at-home mandates."

According to Frost & Sullivan, even professional headset shipments are growing during the pandemic with the surge in cloud meeting services. The global professional headset market is expected to reach \$2.94 billion by 2026, from \$1.41 billion in 2019. Frost & Sullivan also reported that the demand for professional PC USB and UCC headsets has surged to unprecedented levels to enhance software communications experiences in 2020, moving more than 25 million units.

Similar growth is reported by market firm Futuresource Consulting. "The rise of remote working, remote schooling and the

growth of gaming have all made their positive contributions," writes Luke Pearce in Futuresource's latest report for the segment. And as the gaming accessories market also grew more than 20% in 2020, according to Futuresource, with more than 40% of the worldwide gaming accessories retail market value, gaming headsets account for the lion's share of the market, with Futuresource forecasting 26% year-on-year volume growth in 2020. Even when target gamers, leading top brands in the gaming headset segment, Turtle Beach, Logitech G, Hyper X, and Razer, are well aware that many of its products are selling also for communication. In 2020, headsets from those brands were proudly displaying the "Certified for Discord" label, instead of Zoom, Skype, or Teams.

And that's precisely the reason why even Bose tuned its marketing message for the Bose Noise Cancelling Headphones 700 UC—a special edition of the consumer model introduced in 2019, and which finally became available worldwide at the end of 2020. Certified for Microsoft Teams and compatible with both Zoom and Google Meet, the Bose NCH 700 UC was optimized for collaboration and the workplace environment, and is now mainly promoted for remote work or work from anywhere. It features an adaptive eight-microphone proprietary system—with four of the mics dedicated to isolating the user's voice—while the array actively hunts down and rejects noise. This, coupled with 11 levels of noise cancellation, let users personalize what is heard from the surrounding environment—from virtual silence to open and ambient. The specific UC model ships with the Bose USB Link Bluetooth module for Windows PCs and devices without Bluetooth.

Another example of a company making headway to the thriving communications headset market, is a relatively new brand but with a vast experience. EPOS was established from the original 2003 joint-venture between Sennheiser Electronic GmbH & Co. KG and Demant A/S, a leading audio and hearing technology group from Denmark. The two companies decided to end the joint venture in 2018, and Demant launched a new owner-branded portfolio under the name EPOS, while continuing to sell the current Sennheiser Communications portfolio of products, co-branded as EPOS | Sennheiser. The 50/50 joint venture in Sennheiser Communications has successfully developed and promoted Sennheiser branded portfolios of Enterprise Solutions, Gaming, and Mobile Music headsets. In 2020, Sennheiser decided to evolve the initiative in new directions and formally end the joint-venture, moving the Mobile Music headsets segment to the Sennheiser Consumer business, while Enterprise Solutions and Gaming headsets evolved as a new independent company, EPOS, under the direct management of Demant.

With headquarters in Copenhagen, Denmark, EPOS now operates in the global market to deliver high-end audio solutions designed for enterprise and gaming. The first major EPOS-only branded gaming product was launched in 2020, while the EPOS enterprise headsets have now strongly refocused toward remote work and four new EPOS ADAPT headset models, all Certified for Microsoft Teams, were launched in the same year. The portable, wireless ADAPT headsets all feature an advanced state-of-the-art active noise cancellation (ANC) system that eliminates unwanted background noise, all while delivering the best possible audio experience for both speaker and listener. The headsets upgrade to further audio performance thanks

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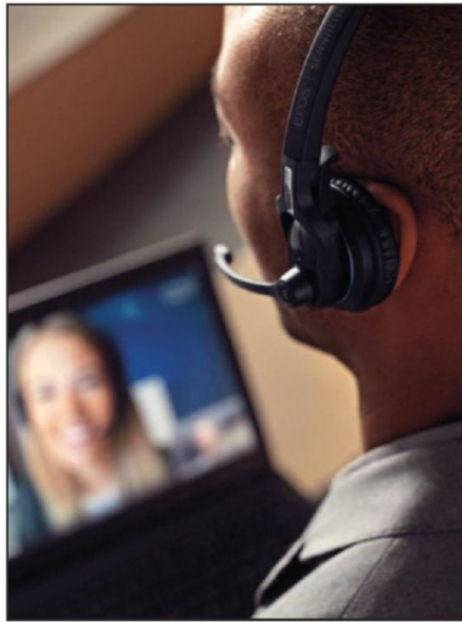


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EPOS' current IMPACT range of communication headsets includes four different series and a total of 11 models, some of which feature ANC, and all previously certified for Skype for Business, are now offering a firmware upgrade to be certified for Microsoft Teams.



to UC optimization, as well as multipoint connectivity that enables users to easily switch between devices by pairing two Bluetooth devices simultaneously, thus enhancing both flexibility and versatility. The four new EPOS ADAPT headsets all feature a dedicated Microsoft Teams button on the headset, so users can launch the app instantly to ensure an intuitive and streamlined experience.

The ADAPT 660 headset, featuring an advanced four adaptive ANC microphone system, is the first product within the EPOS catalog to introduce EPOS AI technology, offering intelligent adaptive audio through unique algorithms that optimize voice pick-up regardless of the user's environment. And the headset also supports Alexa Voice Assistant, and offers full UC optimization with multi-point connectivity. When not in use, the dual-foldable design neatly fits into a handy carry case.

Voice Applications

Market applications for voice are expanding well beyond communications. A growing number of companies are in fact pursuing new opportunities to adapt voice interfaces and voice personal assistants for many new forms of conversational interactions, from technical support in call centers, to health applications. "Intelligent assistants have moved into a new phase of adoption as thousands of firms employ chatbots, voicebots, or virtual agents to improve customer experience and employee productivity," says Dan Miller, lead analyst at Opus Research. "Their challenge is to make intelligent assistants proficient, scalable and omni channel."

In fact even industry pioneers in voice recognition are pursuing those opportunities. Nuance Communications, the pioneer in text recognition, voice-to-text transcription, and natural language interfaces—the company that was once Dragon Systems and ScanSoft—has been building deep domain expertise across verticals such as healthcare, financial services, telecommunications, or retail. Now focusing on conversational AI and "building AI-powered solutions that amplify human intelligence," Nuance is pursuing a

consistent policy of company acquisitions in voice and AI, while gradually divesting from its former Dragon transcription application businesses.

Even before the pandemic, many companies were already strategically expanding their scopes to promising or thriving voice-based applications. The growth of talk-radio, audiobooks, podcasting and other conversational content, was already an indicator that consumers were receptive to enjoying audio-only experiences and that emerging voice technologies could be applied in innovative ways.

In May 2019, Soundtrap, a company headquartered in Stockholm, Sweden, officially launched Soundtrap for Storytellers, a cloud-based solution for collaborative podcast creations that offers a user-friendly recording and production process. The comprehensive podcast creation solution offers all-in-one recording, remote multi-track interviewing, audio-to-text transcribing, full audio production and publishing to Spotify. Soundtrap started as a cloud-based audio recording platform that worked across all operating systems and platforms, enabling users to co-create music and podcasts anywhere in the world. The concept caught Spotify's attention, which acquired the business on December 2017. Since 2019, Spotify has been spending hundreds of millions of dollars on podcast-related content and technology acquisitions. Having text transcription associated with podcasts is key for users to being able to search and identify content.

There are also companies such as Audioburst, which specialize in analyzing and indexing podcasts and live broadcasts to generate short clips on specific topics, also allowing content providers and aggregators to personalize audio content for consumers.

And much in the same way as Soundtrap supports audio-to-text transcription, many other cloud-based recording companies have been building those functionalities to leverage the sudden spike in web conferencing and remote collaboration or even online commerce.

Earlier in 2020, Otter.ai, a well-funded startup active in the audio transcription market, introduced the availability of Live Video Meeting Notes, which enables English-speaking participants to open a secure, live interactive transcript directly from a video conference, or after a meeting. Otter.ai's different transcription solutions quickly evolved and now offer the same real-time features for any video conferencing platform.

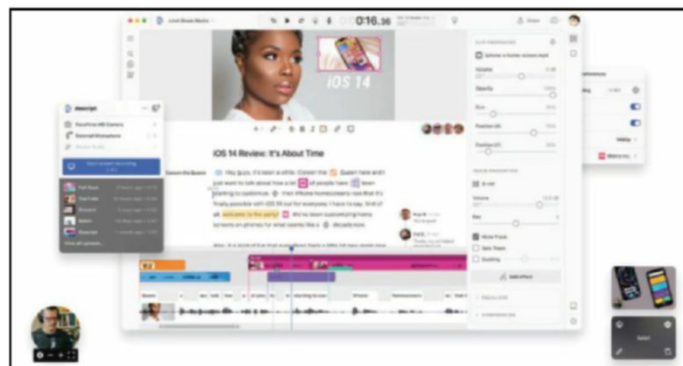
The Otter.ai audio transcription engine was generated with core technology developed by the company's funding team, which previously led development of technologies in mobile, search, speech, and data analytics at companies like Google, Facebook, Uber, Cisco, and Nuance. The Otter.ai transcription engine, which is cloud-based, was optimized for conversations and is also able to detect different speakers. Live text transcript is also a powerful service being offered by web conference platforms to sell their premium subscription options.

Another possibility with having audio transcription is to use the resulting text as useful metadata for editing, an approach that is inspiring a new generation of companies. Simon Says is a company from San Francisco, CA, specializing in timecode-based AI transcription, and is expanding a software suite around its Simon Says platform that makes video editing as simple as copying and pasting text. The application is directly supported by popular

nonlinear editing (NLE) and media asset management (MAM) applications and brings the story forward with narrative-driven editing capabilities that also enable collaboration and speed up video production workflows. Another company is Descript, a San Francisco, CA-based startup developing impressive editing tools with audio-to-text and even text-to-audio tools. Descript offers advanced AI and voice synthesis capabilities, able to analyze a conversation, and eliminate filler words like “Uh...,” “You know...,” or “I mean...,” and create a clean fluid version of a speech.

Those same types of engines are also being increasingly adopted for cloud-based CRM, call-center, contact center, and customer engagement service platforms. This is leading to frequent acquisitions of speech recognition and voice transcription technology companies (mostly startups) by powerful CRM and contact center solution companies, looking to unify all conversations and interactions. And adding voice recognition and voice-to-text transcription is also a “must have” for meetings and collaboration solutions company. As an example, in March 2020, Lifesize, a UC&C company, and Serenova, a leading contact center-as-a-service (CCaaS) provider, announced a merger, creating a new combined business that addresses the convergence of contact centers and unified communications and collaboration with unified software technologies that include advanced voice AI tools.

Even Amazon is recognizing the potential of contact center technology and Conversational AI and is combining resources from its Amazon Alexa program and Amazon Web Services (AWS) to establish new partnerships and offer new solutions in that space. This is an area that Google also explored, at least from the technology point of view, with a strong effort to perfect Conversational AI solutions. No one will ever forget the 2018 controversial demonstration of using an AI-powered assistant to call a real business on the phone to make an appointment (see it here: <https://youtu.be/D5VN56jQMWM>). If anything, that demonstration, which later Google assumed was staged,



Using Descript’s AI-based tools, recorded video is transcribed in real-time, allowing editing in the same way as editing a document. Users just simply cut, copy, paste, and delete text, and the underlying audio and video will follow. Filler words or rambling outtakes can be removed with a single click, and if users want to change a word or phrase in an audio recording, Overdub allows users to create an ultra-realistic clone of their own voice and create speech by typing text in the Descript app.

served to show the potential of a digital assistant interacting as a real human—something that many believe crosses an ethical frontier.

With refinements to Apple, Amazon, and Google’s Natural Language Processing (NLP) and Machine Learning (ML) engines (as well as efforts from Facebook, IBM, and Microsoft), it becomes clear that conversational user interfaces will play a big role in many industries, which in turn will require a corresponding effort from many other audio industry components.

Very soon, many other smaller companies will make inroads to offer specialized voice assistants with more affordable models than the cost of adoption of any of Amazon and Google’s solutions (not compromising on privacy), while a convergence of hardware and computing platforms is making it possible to run implementations of these sophisticated engines directly on endpoint devices—on the edge. And that is where the industry focus, including from leading voice pioneers like SoundHound or Sensory, is now converging. 

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Tectonic BMR Drivers in Unified Communication Products

Balanced Mode Radiator transducer technology, originated from New Transducers Limited also known as NXT. This was some time ago, so these transducers are not so new anymore and the IP now rests just outside Seattle with Tectonic Audio Labs. Going forward, Tectonic is the sole resource for design and development of Balance Mode Radiators or BMRs.

The technology was originally patented in 2006 (US Patent 7916878 B2) and with extension the IP is protected well in to the next decade. One may ask why it has taken so long from BMRs to become prevalent in the market. The problem has everything to do with the commoditization of transducers. BMR's are bulked into a commodity category and with commodity products, the problem is always the lack of differentiation. Without an application that can separate a product, the buyer does not care and they go with the lowest price or offerings that are most convenient. For a mass deployment, one needs the proverbial "killer application." The killer application for BMRs are in space constrained speakers that require full range sound.

The advent and now acceleration of conference speakers and unified communications (UC) products due to COVID-19 has revealed the benefits of the BMR, and Tectonic is placing a significant effort to deploy drivers into UC products. BMRs have a flat, composite diaphragm in the front, but all the other components are the same as those used in traditional drive units. Further, they are a drop-in replacement for standard speakers.

Tectonic is seeing some brands with vision taking the hint. The Zoom or Microsoft Teams' warriors are forcing this issue as they are buying speakers with microphones in them for conferencing at an alarming rate.

The reason why BMRs work so well in such applications is the combination of a mid-range and a tweeter into a single drive unit. Human hearing is most sensitive between 1kHz to 3 kHz, which is the frequency location where two-way systems normally cross-over. With microphones in UC products trying

to pick up your voice and spikes or dips emanating from the speakers at the crossover point, it becomes more difficult for the microphones or mic arrays to operate. These issues tend to be solved with signal processing but the cost savings in engineering from reduced crossover and acoustic echo cancellation work can be quite dramatic. And BMR drivers cover upper bass, midrange, and treble in a single drive unit, allowing improved frequency response and reduced system BOM costs.

Another UC product benefit is that one of the key characteristics of BMRs is the sustained off-axis output and smooth sound power level (SWL) response. This is different from typical on-axis sound pressure level (SPL), which is a measurement from a single point in space, whereas SWL includes all the SPL in a hemisphere around a speaker. BMR drive units intrinsically have a smooth and extended SWL response. This ensures that the total acoustic energy radiated into a given space is extremely low coloration and natural sounding, providing high intelligibility, natural sound throughout the room.

SWL translates clearly to uniform sound level and audibility at all angles from the speaker driver. For the user, this means that with a BMR the sweet spot is tremendously expanded. With UC products, this means everyone around a conference speaker or someone pacing in their office or a parent working from home juggling with kids, coffee and work can all hear the speaker clearly, no matter where they are in the room. Wide directivity is achieved over the full frequency range in a single driver, regardless of size, so clear intelligibility is maintained throughout the space.

Intelligibility requirement also acts in combination with the uniformity of the sound level. Everyone understanding each other on calls using UC products is critical. BMRs with a wide directivity and smooth frequency response from a single driver, together with the lack of distortion in BMRs are proven to increase intelligibility. The single unit approach

A BMR driver with the composite panel (left) vs. a traditional driver (right)



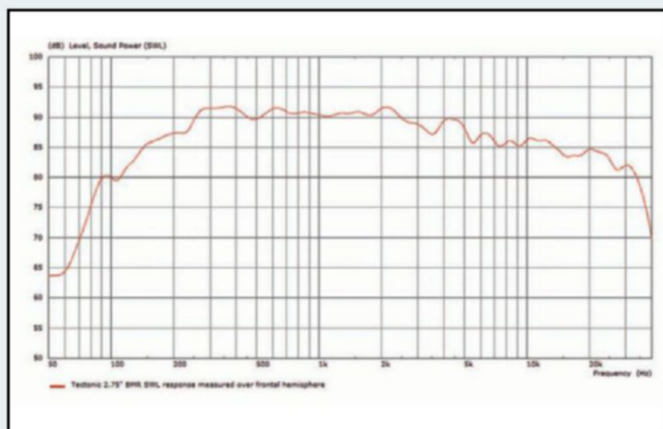
eliminates interference and/or phase cancellation patterns that are typically found in two-way or three-way speaker systems.

The cost analysis of using a BMR driver is based on the system approach, not the individual driver. Further to this, the BMR replaces (at least) two speakers with one and eliminates the associated crossover network, while amplifier channels

and digital signal processing requirements are reduced. Also, it permits faster engineering time to market due to the reduction of the design overhead in two-way or three-way speaker designs. This is especially pronounced for UC products. Factories building with BMRs can use all the same basic equipment and processes used in traditional speaker builds, with a few minor exceptions related to the panel attachment to the voice coil.

Tectonic got the memo from the market regarding UC applications and offers several BMRs designed specifically for such systems. The 1 3/4" TEBM35C10-4, 1 1/2" TEBM28C10-4B, and the new 2 3/4" TEBM54C30-8 are all designed by Tectonic with this application in mind. There are several more BMRs in the pipeline also designed with consideration for UC and voice applications. The drivers are already deployed into Owl Labs products (owllabs.com) and several other major brands have products on the way. The supply chain in China is firmly established so incorporating BMRs into unified communications products is quite simple.

www.tectonicaudiolabs.com/audio-components/bmr-speakers
<https://db-ent.com>



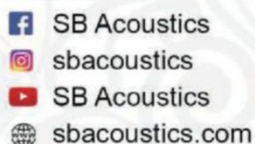
Measured Sound Power Level (SWL) response for the new 2.75" TEBM54C30-8 BMR driver from Tectonic Audio Labs

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Speaker ID Technology

Increasing consumer demand for personalized experiences has sparked the creation of speaker ID technology. This article explores the challenges that this type of technology still faces.

By
Vikrant Singh Tomar
(Fluent.ai)

In recent years, voice technology has proliferated the market, with more devices becoming enabled with speech recognition for added convenience to the end user. Most popularly, smart speakers have dominated the voice recognition market, with more than 157 million smart speakers in homes across the US.[1] Since this success, speech technology is beginning to pop up in other areas, including home appliances, vehicles and more. However, key areas of the market remain untapped, as applications of speech technology are far wider than just smart home devices.

With this in mind, Fluent.ai, an embedded voice recognition solutions company, has developed a technology that operates fully offline. The company was founded more than 5 years ago based on the research done for my PhD thesis on deep learning based speech recognition.

Today, Fluent.ai's solution offers a great deal of features that address common problem areas with speech technology, including being multilingual, able to understand any language and accent, operating fully offline, offering another layer of privacy, and more. The technology utilizes a unique speech-to-intent model, bypassing the need for the traditional speech-to-text approach. This allows the solution to operate fully offline, never needing to transmit information to the cloud, and therefore offering a more private solution. **Figure 1** compares conventional speech models to that of Fluent.ai.

Possible Applications

Fluent.ai primarily works with original equipment manufacturers (OEMs) and original device manufacturers (ODMs) to embed its technology in a variety of products, working within industries such as manufacturing, retail, smart home, and more. Applications of the technology run the gamut, as Fluent.ai's

technology is made to be nimble and applicable for everything from wearables to robotics (**Figure 2**).

This year, Fluent.ai attended the all-digital CES, and introduced a Speaker ID function with its technology. This enables the tech to provide personalized responses to a user depending on who speaks the command. For example, if two different users ask a device to play their favorite music, two different kinds of music will play based on what it knows about each individual's preferences.

Today's consumers are looking for increasingly personalized experiences, and interacting with their technology is no exception. This has led to an increasing demand for speaker ID tech to be integrated into more household items. Fluent.ai's offerings are making strides over the hurdles of traditional speech technology, creating a robust solution in today's market that overcomes many challenges.

Profiling Voices

In today's economy, consumers are continuing to look for more personalized experiences—with more than 80% of people indicating they're more likely to make a purchase when offered a personalized experience.[2] This same desire and mentality can be applied across industries. With this in mind, personalized technology is also on the rise. Consumers are continuing to hunt for technology that understands their personal preferences and reacts accordingly. Speech technology is no exception. Speaker ID has recently risen in popularity, with major players such as Apple and Samsung beginning to scratch the surface of this. However, developing an offering such as speaker ID doesn't come without its hurdles.

One of the greatest challenges for speaker ID technology is ensuring that the device responds accordingly to each unique user's requests. This

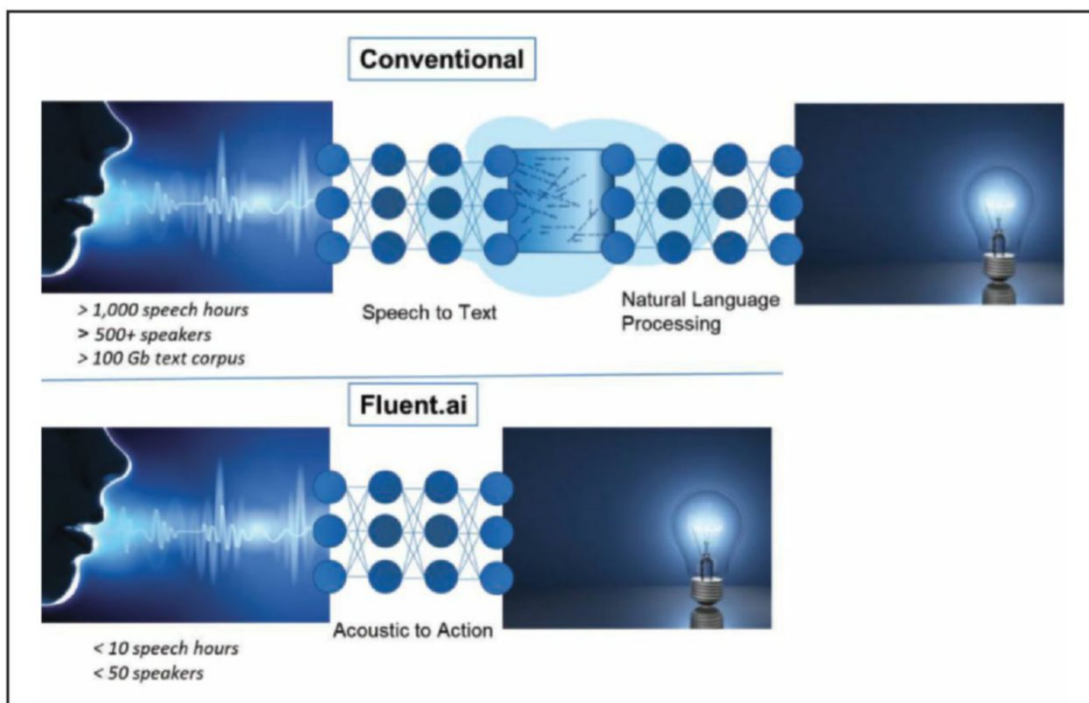


Figure 1: This is a comparison of a conventional speech model vs. that of Fluent.ai.

requires detailed efforts to ensure that each voice is correctly profiled to respond appropriately. All of Fluent.ai's products are based on cutting-edge neural network technologies. For training the models, a large number of audio files are used from different speakers—spanning across languages, dialects, accents, and more. From this, the model learns the voice profile of a user from just a few utterances.

Once the profile is established, the neural network is able to leverage this model in order to provide personalized responses. As an added benefit, because the models are initially trained on data recorded through a variety of different front ends, these models do not have a dependency on any particular signal processing front end. Furthermore, the registration of a new user would happen on the final device that the technology is embedded on, therefore the registration would be done on the front end of this device, as opposed to the software.

Low Power

While it may not be immediately top of mind for a consumer, low-power solutions are extremely

beneficial. This isn't something an end user often thinks about, until they find the battery of their device quickly dies, ultimately leading to a great deal of frustration. This makes it imperative for OEMs to seek out low-power solutions to help ensure satisfaction from their end users, no matter the industry. For artificial intelligence solutions, the latest edge technology is all about consuming the least amount of power while creating a solution that is optimized for peak performance. Furthermore, a low-power solution is more cost-effective, making it more attractive to utilize. Fluent.ai's products are always developed to run on low power, and its speaker ID is no exception. This enables the technology to be nimbler and more applicable across industries—previously a pain point of adoption.[3]

The speaker ID model works in conjunction with Fluent.ai's wakeword model. For example, when a user speaks the programmed wakeword, such as "computer," the system recognizes if the correct wakeword is being spoken by a known or registered user. To maintain a low-power solution, the team designed a neural network architecture in which

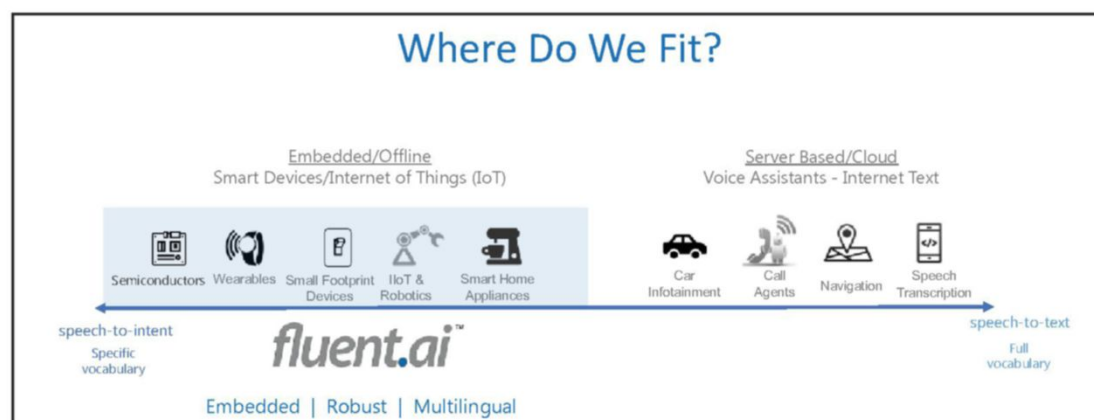
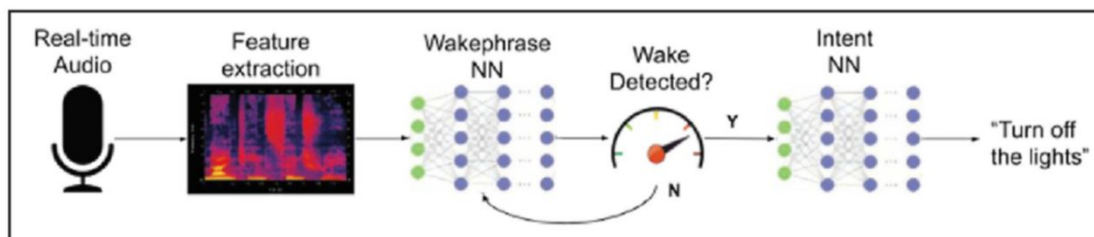


Figure 2: Fluent.ai's possible applications

Figure 3: Fluent μ Core consists of the wakeword and automatic intent recognition engines.



the wakeword and speaker ID networks share layers in order to achieve parallel recognition of both the wakeword and speaker ID simultaneously. By layering these models together, the software is able to operate with less computing power because they can draw upon similar models to achieve the goal of responding to a command.

Edge Processing

AI on the edge has been a hot button topic lately, with adoption accelerating over the last few years. As artificial intelligence continues to touch nearly every industry in some capacity today, edge processing is beginning to proliferate too. Fluent.ai uses a series of steps to optimize the neural networks for edge processing.

Its proprietary neural network architectures use less computing power as compared to the industry. Furthermore, the team utilizes several neural compression and mathematical optimization techniques, which reduce the size and computing requirements of Fluent.ai's networks. And, Fluent.ai has its own runtime library—Fluent μ Core—that is optimized for an increasing number of hardware platforms. This enables the company's technology to have even greater applications. The Fluent μ Core is able to

perform streaming neural processing, which results in smaller memory requirements, and therefore it can run more capable networks on memory and CPU-constrained edge devices.

With all of this in mind, Fluent.ai leveraged edge processing to help develop its speaker ID technology. This lent itself to creating the neural network architecture that enabled the company to layer the wakeword and speaker ID models, and ultimately create an efficient and low power solution (**Figure 3**).

Multilingual

Currently, there are more than 6,500 languages spoken worldwide. However, there are still many hurdles that mainstream technology needs to overcome to be more accessible to greater populations. Within speech technology, being understood is one of the greatest pain points for an end user. Mainstream options still have trouble understanding stutters and accents, as well as bilingual users and voices of people of color. While the industry is working to address these challenges, they are typically retroactive solutions, instead of solutions that are built into the technology at the start.

Similar challenges can be found in speaker ID technology, but Fluent.ai's approach ensures that this tech is more widely accessible. All of the company's offerings are trained to understand hundreds of languages, accents, speech impediments, and more, ensuring that the solution will respond accurately no matter the speaker's background. Fluent.ai's speaker ID solution is no exception, specifically programmed with its technology that can recognize any language, accent, impediment, and more. As the demand for more personalized user experiences rises, the call for technology to be more inclusive will also rise.

While the company only recently introduced the tech at CES 2020, Fluent.ai's speaker ID technology has a variety of potential consumer applications, including everything from system preferences in business meeting rooms, to playing a user's favorite music on their smart home device. As end users continue to look for personalized solutions, speaker ID technology will continue to rise in popularity. Through Fluent.ai's technology, OEMs can help deliver on this demand by integrating the embedded solution into their offerings.

About the Author

Vikrant Singh Tomar is the founder and CTO of Fluent.ai, a speech recognition company based in Montreal, Canada. Vikrant has more than a decade of experience in speech recognition and machine learning, and holds a PhD in automatic speech recognition and machine learning from McGill University. He has worked at Nuance Communications and Vestec Inc. in the past.



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True Wireless Earbuds Evolving Toward Hearables

By
J. Martins
(Editor-in-Chief)

Sweden-based
Earin's latest A-3
TWS earbuds (Image
courtesy of Earin)

There's a lot happening in the true wireless stereo (TWS) category and this market update article intends to provide a quick overview of the state-of-the-art technologies and platforms in the space, as well as some reference approaches in product design.

According to Strategy Analytics, the global market for true wireless stereo (TWS) earbud devices is expected to grow faster than smartphones did 10 years ago (70% CAGR vs. 60%). As the market research company notes, the category evolved from 1 million units of Apple AirPods sold in 2016 to 15 million in 2018, and selling a staggering 60 million units in 2019, with the introduction of the AirPods Pro.

Strategy Analytics says that total global Bluetooth headset sales have reached volumes over 300 million units in 2020 and that TWS Bluetooth headset sales are now more than half that (170 million). Juniper Research, in a report published in mid-2020, estimated sales of 260 million units for 2020.

Other analysts estimate that Apple sold close to 85 million AirPods in 2020, taking as much as 70% of the total revenue in the TWS category, and 55% market share in units sold.

This also means that Apple's AirPods business alone could already represent between 12 to 18 billion in sales (no one knows for certain, since Apple doesn't disclose those sales figures). Anyway, even based on the lowest estimate, that represents a larger business in itself than most technology companies.

No wonder suddenly every audio and technology company is paying serious attention to this product segment pioneered by companies such as Bragi or Doppler Labs, only 6 years ago. And even more impressively, Apple's designs are actually just great implementations of the basic concept, first translating the convenience of true wireless stereo, and in the second iteration (AirPods Pro) introducing active noise cancelling and hear-through, starting to evolve toward hearing enhancement. A very powerful platform, with loads of cutting-edge technology that does the essential very well, in typical Apple fashion.

But the AirPods Pro also surprised and excited the industry by introducing a new concept. Since the AirPods Pro are based on Apple's own dedicated H1 headphone chip, the products are able to receive firmware updates that, among other things, introduced spatial audio support. The update also paved the way to new accessibility features, adding what Apple called "Headphone Accommodations." Basically a way to amplify soft sounds and tune audio to help music, movies, phone calls, and podcasts sound crisper and clearer. This accessibility feature is also implemented when the AirPods Pro Transparency mode is active, helping users to better recognize important sounds and dialog.

And once again, by showing that TWS products can evolve in features, Apple gave the starting shot to a race that will pave the way to real hearing augmentation and hearables.

TWS Earbuds Are Just Starting to Evolve

According to Futuresource Consulting, this is a market in its infancy, with the potential to quickly challenge the hearing aid industry. As Futuresource's latest research details, the so-called "global hearables market" (including all TWS earbuds, hearing aids and Personal Sound Amplification Products or PSAPs) represents (conservatively) 139 million units sold in 2019, with a market value of \$19 billion US, forecasted to reach approximately \$30 billion in 2021. In stark contrast, Hearing Aids sell approximately 20 million units annually (volumes decreased sharply in 2020 due to the pandemic restrictions). According to the latest 2020 quarterly tracker report from Futuresource Consulting, the overall headphones market achieved year-on-year growth in value and units, which could have been much bigger if it wasn't for supply-chain constraints that were unable to meet growing demand. And Futuresource estimates that

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Apple brought Spatial Audio to AirPods Pro with a firmware update, changing the way the market perceived the whole product category.



The extraordinary growth in the true wireless stereo earbuds is attracting all brands to enter the segment. Pictured are the latest KEF Mu3 Noise Cancelling TWS design.



Anticipating hearables, WS Audiology's launched Signia Active Pro earbud-style hearing aids.

TWS has been a key factor in that growth, accounting for more than 45% of total headphones shipped.

Excited by that market potential, consumer electronic and technology players are firing all cylinders in edge AI, sensor fusion, and better audio front ends for advanced signal processing. Research efforts are feeding fast-paced innovations in voice recognition, active noise canceling (ANC) and augmentation, while everyone is also researching new ideas, such as spatial audio and health applications. Of course, consumers can hardly keep up with all the possibilities.

A very useful summary of trends in this space resulted from research conducted by Austrian company ams AG in Spring 2020, with 2,000-plus consumers around the world. "Comfortable, Smart, High-performing Earbuds Survey," focused on audio earbud uses, habits, and preferences, and concluded that comfort is still a key focus for users. But the survey also identified the benefits of active noise cancellation and volume-limiting technology as top earbud innovations now mentioned by consumers.

Consumers also mentioned 3D/spatial sound and long battery life as criteria for selection, which means that there is now a much better awareness of the category. Regarding design, ams studied user preferences between loose fit and tight-fitting in-ear earbuds depending on activity. The ams survey showed earbuds are an integral part of daily life for many consumers. Users in the US and some European countries reported that they tend to wear earbuds all day—while making calls, listening to music, or exercising.

Around 45% of the users surveyed by ams revealed they use tight-fitting, in-ear earbuds because they are perceived to stay in place (67%) and feel more secure when exercising (51%). However, consumers noted tight-fitting (occluded) earbuds are not comfortable for extended periods or they do not fit correctly (44% across all regions surveyed). Respondents also reported that they want to be aware of their surroundings, and that volume-limiting technology to prevent hearing loss or damage is important across the world with 56% requesting this feature.

Clearly, research shows that as consumers are becoming more familiar with new functionalities, such as ANC and hearing augmentation modes, the more they tend to consider them when choosing their next purchase. And users are now much more aware of form factors, being clear that loose-fit designs will be generally more well-tolerated for longer use. Still, users want the best of both worlds: loose-fit earbuds with advanced ANC and processing features.

Whatever the reason for considering the possibilities or choosing one design over another, research also shows that TWS earbuds will have short renewal cycles and that consumers are willing to investigate new designs regularly.

Bluetooth LE Audio Transition

Probably the greatest enabling technologies in the true wireless space will result from the implementation of LE (low energy) Audio.

Isochronous channels (ISO) over Bluetooth LE is the foundation for LE Audio, introduced in the latest Bluetooth LE 5.2 specification. With ISO, several low latency and synchronized channels can be set up to stream audio over Bluetooth LE to several sinks, such

as a pair of wireless earbuds... or many. The audio experience is also extended thanks to significantly reduced power consumption compared to Classic Audio, and improved audio quality due to new LC3 Codec, complemented by the mentioned ability to stream audio to an unlimited number of devices.

With these updates, the next generation of Bluetooth audio will enable new use cases, such as consumers sharing wireless audio (Personal Audio Sharing), and accessing location-based wireless audio sources, such as screens in public venues broadcasting wireless audio. Bluetooth 5.2 is now starting to be implemented in multiple source and sink devices, and will enable generally more reliable true wireless experiences, thanks to Multi-Stream Audio and the new, high-quality, power efficient, LC3 audio codec.

As Vincent Nallatamby, the founder and CEO of Tempow states, LE Audio “will significantly improve the performance of Bluetooth Audio and enable a new generation of Wireless Earbuds capable of substantial optimization from both a technical and user experience perspective.” The challenge now, as he noted in early 2020, is to match the availability of source devices with LE Audio support—smartphones, tablets, laptops, etc.—before the latest Bluetooth chipsets supporting LE Audio start moving millions.

The leading chipset and IP vendors are all now releasing the needed solutions. Since September 2020, the Bluetooth Special Interest Group (SIG) started qualifying products using the new LE Audio specification, originally reported in January 2020.

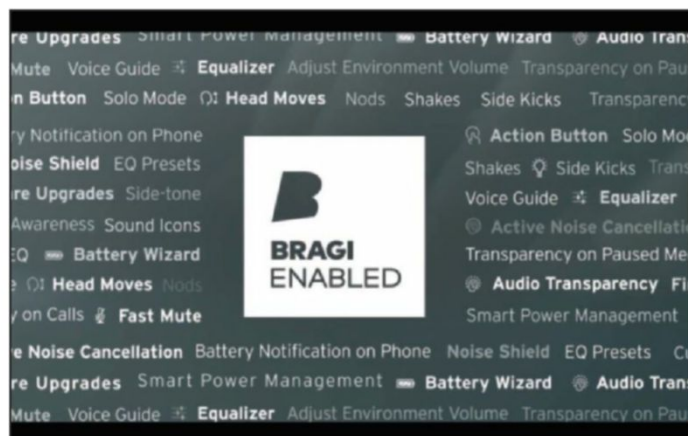
CEVA announced in November 2020 that its RivieraWaves Bluetooth Low Energy (LE) 5.2 Platform received Bluetooth SIG Qualification, enabling its licensees to boost product development with fast time-to-market, benefiting from expedited end-product qualification process.

The Next Big Thing

In the middle of this key transition, and with several innovations and new platforms being announced, lots of new options are currently available to developers and brands to differentiate their designs. Apple’s competition is heating up from Amazon, Google, and Microsoft, all of which entered the TWS space in 2020, while Samsung, Xiaomi, and Huawei continue to aggressively expand their business. “While the rapid growth of the Chinese market is prominent, Chinese players are expected to pick up further in 2021. Not only are global brands but the emerging Chinese local brands in low-end products will also command attention,” states Liz Lee, Senior Analyst at Counterpoint Research.

“New models are ceaselessly released with various features and functions, including noise cancellation, bio-sensing, waterproofing, and integrated AI voice assistants in broader price ranges from low-cost entry to high-end premium,” she adds. “Based on future product strategies focused on voice communications from global tech companies and their efforts to create high-added value, the true wireless hearables are just becoming one of the daily necessities among consumers.”

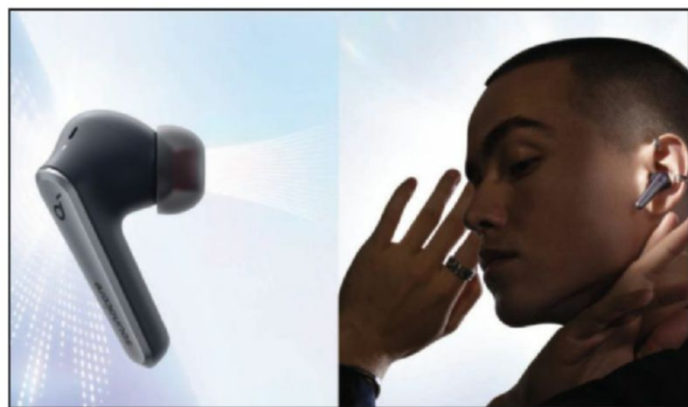
And SAR Insight & Consulting predicts that this trend will inevitably disrupt the hearing aids industry, with a direct to consumer (DTC) and over-the-counter (OTC) approach being an opportunity too large to miss for consumer electronics companies.



From a pioneer in the true wireless space, Bragi GmbH now offers Bragi OS, a specialized operating system for hearables, enabling manufacturers to add software apps and services such as voice activation, voice biometric, audio post processing and music services.



Scottish brand RHA introduced TrueControl ANC, a new cutting-edge true wireless stereo earbuds design, using Qualcomm’s aptX and AAC Bluetooth streaming with advanced active—and adjustable—noise cancellation, advanced connectivity, and a 20-hour battery life.



Anker Innovations’ Soundcore Liberty Air 2 Pro features advanced active noise cancellation (ANC), transparency mode, and uses the company’s PureNote multi-coated driver technology. This is a mid-priced true wireless ANC alternative with truly premium features and technology.



The new HP Elite Wireless Earbuds, developed in collaboration with hearable pioneers Nuheara, combine ANC with personalization and customization features, using the ams AS3460 Digital Augmented Hearing solution that powers the Nuheara IQbuds2 MAX.



Earin, a technology company from Malmö, Sweden, and pioneer of the original true wireless concept, unveiled its third-generation of premium earbuds, the A-3—a more mature Bluetooth 5 product that combines a new loose-fit design with improved audio quality.



The results of an extended collaboration, Klipsch teamed up with Dirac's digital audio processing experts, and the pioneers in hearables and creators of the Bragi Software Suite with embedded AI, to create the new Klipsch T5 II True Wireless ANC Earphones.

Offering an apparently conventional TWS design, which survived multiple, similarly looking iterations, Australian startup Nuheara finally achieved its goal of standing out in a crowded consumer space, with the ability to enhance the user's hearing ability. Nuheara's latest IQbuds2 MAX Smart Hearing Earbuds (\$399) are finally selling in volume.

A bridge toward truly smart hearables, the Nuheara IQbuds2 MAX product enjoys critical acclaim thanks to a combination of great sound quality, advanced ANC and personalized hearing enhancement.

Nuheara IQbuds2 MAX earbuds use a powerful triple-core DSP system and enable even more noise control options, benefiting from broadband noise attenuation performed by ANC technology from Austrian firm ams AG. The product delivers an affordable option for consumers who are not ready for hearing aids, but who are looking for improved listening, and looking to enjoy a music listening experience as good as that of conventional earphones. The design also persuaded HP to license the technology from Nuheara, resulting in the introduction of the HP Elite Wireless Earbuds.

And consumer TWS earbuds are starting to appeal to hearing aid companies, which understand how the category is changing consumers' perception to surpass existing stigmas. As a result, WS Audiology's hearing-aid brand Signia is now targeting a new generation of users with its Signia Active Pro earbud-style hearing aids that also offer high-tech audiological benefits. According to the company's research, "Two out of three people would prefer to wear earbuds to hear better, instead of the most stylish high-tech hearing aids previously available."

The new Signia Active Pro earbuds combine standard TWS features with a connected app that enables users to customize their experiences, adjusting intelligibility to the level and type of ambient noise. But the app also offers 24/7 support and connects the user to the full remote service of Signia TeleCare, enhancing follow-up audiology appointments with real-world usage insights.

Spatial Audio

In such a fast-paced product segment, where innovation and differentiation are essential, manufacturers are trying to leverage existing chipsets and platforms to quickly bring new designs to market, while introducing new features. While ANC is considered a key feature, as well as personalization tools, it seems many brands are now following Apple's lead and are all working on earbuds with Spatial Audio—enabling an immersive experience while watching movies in Dolby Atmos, playing games, or listening to music services that are starting to offer Dolby and MPEG-H spatial formats.

Swedish digital audio processing pioneer Dirac, already offers solutions for spatial and 3D audio reproduction in smartphones and conventional headphones, and is teaming up with many SoC vendors, design studios, developers, and manufacturers to bring TWS implementations with spatial audio to market.

Dirac's spatial audio solution, which leverages the company's patented Dynamic HRTF (head-related transfer function) technology, is available for direct integration into digital signal processors (DSP) and Bluetooth chipsets from Qualcomm, BES Technic, and MediaTek. And Dirac also offers speaker optimization possibilities to brands that adopt the processing suite.

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ZUMI SYSTEMS



The ZUMI CS-A86 is a single board audio spectrum measurement instrument. The CS-A86 operates autonomously to execute advanced measurement sequences, apply pass-fail criteria and log all data. The device is aimed at implementation in automated test equipment for production and development. The primary function is to obtain frequency response function measurements for devices under test (DUT) that require testing of signal paths especially involving filter structures or other signal conditioning elements. The built in capability to configure DUT's via serial interfaces allows a comprehensive set of quality control tests to be completed without user input.

Features

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

Applications

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

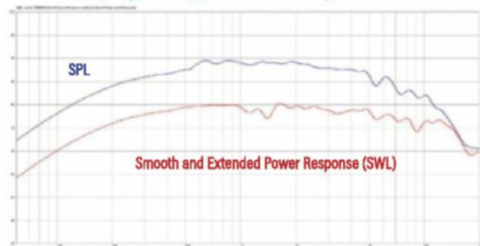
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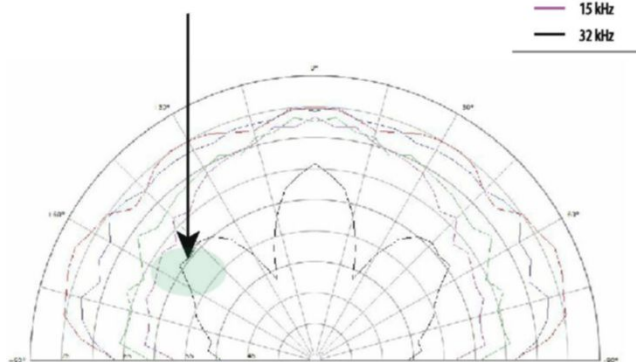


Freq. Resp.	130 Hz - 20 kHz
Pwr. Handling	10 Watts
SPL (1W/1m)	80 dB
FS	129 Hz
DCR	4 Ohms
Dim. (mm)	46 (OD) x 25.8 (h)

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ams worked with Padmate to perfect the PaMu Quiet earbuds, a unique solution in the space that uses dual chips, combining the ams AS3460 Digital Augmented Hearing device with the standard Qualcomm QCC5124 Bluetooth System-on-Chip.

Dirac announced projects with Klipsch and RHA, and also teamed up with Swedish manufacturer and product designer Sudio to deliver next-generation TWS earbuds, scheduled for release in Summer 2021. To enable spatial audio, these upcoming Sudio TWS headphones will boast a dual-core processor chipset that features Dirac's spatial audio solution, embedded directly into the devices chipset. This enables spatial audio independent of playback device or media player.

To meet the same demand for TWS earbuds, CEVA teamed up with VisiSonics, a developer of patented 3D spatial audio technologies. The company is working closely with CEVA to develop a comprehensive embedded 3D spatial audio solution using its RealSpace 3D audio software, optimized for CEVA's low power audio and sensor hub DSPs, together with CEVA's MotionEngine head-tracking algorithms

running on its BNO080 nine-axis System in Package (SiP). The CEVA partnership with VisiSonics also meets the demand from manufacturers requiring completely new designs, with a high precision real-time 3D audio solution for System-on-Chip (SoC) vendors, OEMs, and ODMs. The CEVA/Hillcrest Labs nine-axis SiP enables rapid development of sensor-enabled Augmented Reality (AR), immersive, and spatial audio applications. And the solution also enables users to take advantage of a configurable sensor fusion platform to add intuitive user interface, gesture control, and activity tracking.

The Platforms

Just prior to this edition's publication, CEVA unveiled Bluebud, a highly-integrated wireless audio platform aimed at standardizing DSP-enabled Bluetooth audio IP for applications including TWS earbuds. The new Bluebud solution already fully supports Bluetooth 5.2 and LE Audio design and is ready to support brands in the transition from Bluetooth Classic.

As an integrated hardware and software IP platform, Bluebud aims to solve the technology complexities, providing a drop-in solution for system companies. CEVA designed Bluebud as a turnkey hardware and software platform that is particularly well suited to support developers and brands in dealing with TWS earbuds. Bluebud offers advanced, value-added software and toolkits optimized for the onboard CEVA-BX1 DSP, including audio codecs, voice assistants, spatial audio, in-ear detection, tap control, activity classification, and full support for TensorFlow Lite Micro machine learning models.

The platform combines CEVA's RivieraWaves Bluetooth 5.2 IP, with both Classic Audio and LE Audio, and enables end-product differentiation via support for host-level application software, which can run on the CEVA-BX1 DSP or an optional host CPU.

And specifically targeting ANC and augmented hearing, Austrian high-performance sensor manufacturer ams offers a fully optimized digital ANC solution, based on its ams AS3460 Digital Augmented Hearing digital audio processor and companion chip, providing what the company claims to be the world's best performing noise cancellation for loose- and closed-fit earbuds—up to 40dB of noise attenuation in the loose-fit type wireless earbuds over a broad range of audible frequencies. This solution includes ams' unique Adaptive Leakage Compensation technology and can be combined with sensors and leading Bluetooth SoCs, such as those from Qualcomm, BES, and MediaTek.

Tempow, the Paris, France-based Bluetooth startup, announced a partnership with Zhuhai Jieli Technology, a leading chipset manufacturer in the TWS Bluetooth SoC segment. Jieli Technology has joined Tempow's chipset partner program to deploy TempowOS—an operating system specific for true wireless earbuds—on its flagship AC897N and AD697N extreme low power Bluetooth TWS chipsets.

Jieli's solutions offer support for some of the most thought-after features in new TWS products, such as ANC, and will be compatible and pre-integrated with TempowOS to offer advanced product features such as multi-source (the ability to connect the earbuds at the same time to a laptop and smartphone), fast pairing, and gesture customization.

Tempow is licensing TempowOS as a reliable operating system



Cambridge Audio's latest Melomania Touch Earbuds are a recent design based on the Qualcomm QCCC3020 Bluetooth 5 Audio SoC.

for TWS designs with differentiating features and a platform that can evolve and adapt to consumer trends. Tempow offers its unique patented dual-A2DP architecture on top of Jieli's SoCs, and adds the possibility of firmware updates, ensuring that wireless earbud designs are always evolving with more features.

And no overview of TWS platforms would be complete without mentioning Qualcomm, the company that currently powers virtually all the big-brand TWS designs in the market—from Bang & Olufsen to Jabra, Sennheiser, and Yamaha. In the past two years, Qualcomm has quickly updated its family of SoCs for Bluetooth TWS earbuds development.

Qualcomm has enabled mobile and CE brands to create products able to compete with Apple's massively successful AirPods, including the AirPods Pro with ANC and spatial audio. Building on the success of the existing QCC5100 and QCC30xx ultra-low-power Bluetooth SoCs, Qualcomm is delivering improved truly wireless support with its Qualcomm TrueWireless Mirroring technology. This technology is available with the Qualcomm QCC514X (premium tier) and Qualcomm QCC304X (entry/mid-tier) SoC platforms, available to manufacturers. Both new SoCs have integrated dedicated hardware for Hybrid Qualcomm (Hybrid ANC) and voice assistant support. The QCC514x and QCC304x SoCs are also optimized to deliver longer playtimes (up to 13 hours based on 65mAh battery, depending on settings and other factors), even with ANC enabled.

The more recent Qualcomm QCC305x SoCs are designed specifically to help brands differentiate across a range of tiers, with full end-to-end support for the new Bluetooth LE Audio standard. The QCC305x series also adds support for ultra-low-power digital ANC technology, Qualcomm Adaptive ANC, Qualcomm TrueWireless Mirroring, and wake-word activated voice assistants. Other premium audio technologies include Qualcomm aptX Adaptive at up to 96kHz, for high-quality listening and low latency streaming while watching videos or gaming; and Qualcomm aptX Voice and Qualcomm cVc Echo Cancellation and Noise Suppression for improved voice clarity on calls.

Qualcomm also offers a vast range of embedded software options for differentiation through its Qualcomm Extension Program. Recent



To enable manufacturers to stay ahead of the curve by bringing innovative new features with Bluetooth 5.2 to market quickly, Qualcomm made available its QCC5141 Occluded Earbud Reference Design with Qualcomm TrueWireless Mirroring and Qualcomm Hybrid ANC.

examples targeting the TWS space include 7 Sensing Software, offering its Hearing Augmentation and Pure Voice solutions running on the premium-tier Qualcomm QCC5100 Bluetooth SoC family. And also, Jacoti, an audio technology company focused on hearing enhancement, brings medical-grade algorithms and software to add accessible, accurate personalized hearing assistance.

In November 2020, Qualcomm announced its latest QCC5141 Occluded Earbud Reference Design Platform to allow manufacturers to stay ahead of the curve by bringing innovative new features to market quickly with Bluetooth 5.2, with Qualcomm TrueWireless Mirroring and Qualcomm Hybrid ANC.

This new Qualcomm reference design platform is designed for the high-end premium segment with support for advanced features such as Hybrid ANC with hardware leak-through, Qualcomm aptX Adaptive audio codec support, and implementations of voice services, with activation of Amazon Voice Services and support for Google Audio Accessories. 

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The Evolution of True Wireless Stereo

By
Jean-Marc Luneau
(7 Sensing Software)



In this article, we will look at some meaningful new features and improvements that can help True Wireless Stereo (TWS) earbud brands differentiate their products and make them stand out in today's crowded market.

True Wireless Stereo (TWS) earbuds have seen an incredible growth these past few years and have evolved from poorly connected gadgets to reliable must-have accessories for virtually any activity: listening to music, consuming other media, calling on the go, and teleconferencing, with new use cases coming (e.g., vital signs monitoring and hearing enhancement).

After connectivity markedly improved over the past years, the primary differentiating factors for TWS earbuds now include:

- Music and media playback quality (with ANC and 3D audio)
- Voice calling quality (both transmitting TX and receiving Rx)
- Battery life

Many brands have been fighting to achieve the greatest amount of noise cancellation and the best playback sound—and there is still a lot of work invested into delivering proper acoustics via design, material choices, and software tuning. All mid- and high-end devices claim great sound quality with renowned acoustic brands sound signatures. Alas, it is still very common to find devices with large drivers or multiple balanced armature (BA) speakers and mediocre sound reproduction. Properly tuning an ANC engine not to degrade music has also proven

hard to achieve even on premium devices.

However, battery life was extended with latest chipsets and the introduction of Bluetooth 5. And the industry is making more effort on codecs (LC3), while extra sensors are also being integrated for ease of use and battery savings pursued (e.g., with optical proximity sensors for off-ear detection).

Despite some improvements since the launch of the first TWS earbuds, the voice calling use case on current devices is also not exempt from flaws. New technologies offer opportunities for significant improvements.

Beyond mastering those must-have features, there is also a clear need for differentiation as all brands offer more or less the same features in each price range. TWS earbuds have great potential to move from being a mostly passive gadget to becoming an always-on active device offering real personal sensing enhancements.

Crystal-Clear Voice Calling

Voice Calling has improved in the mid- and high-end segments, but there is still a very significant margin to do much better even on premium reference models, such as Apple AirPods Pro and Huawei Freebuds Pro that still let too much noise pass through while not always preserving your own voice well. And there is nothing worse than being on a conference call and feeling that your audience

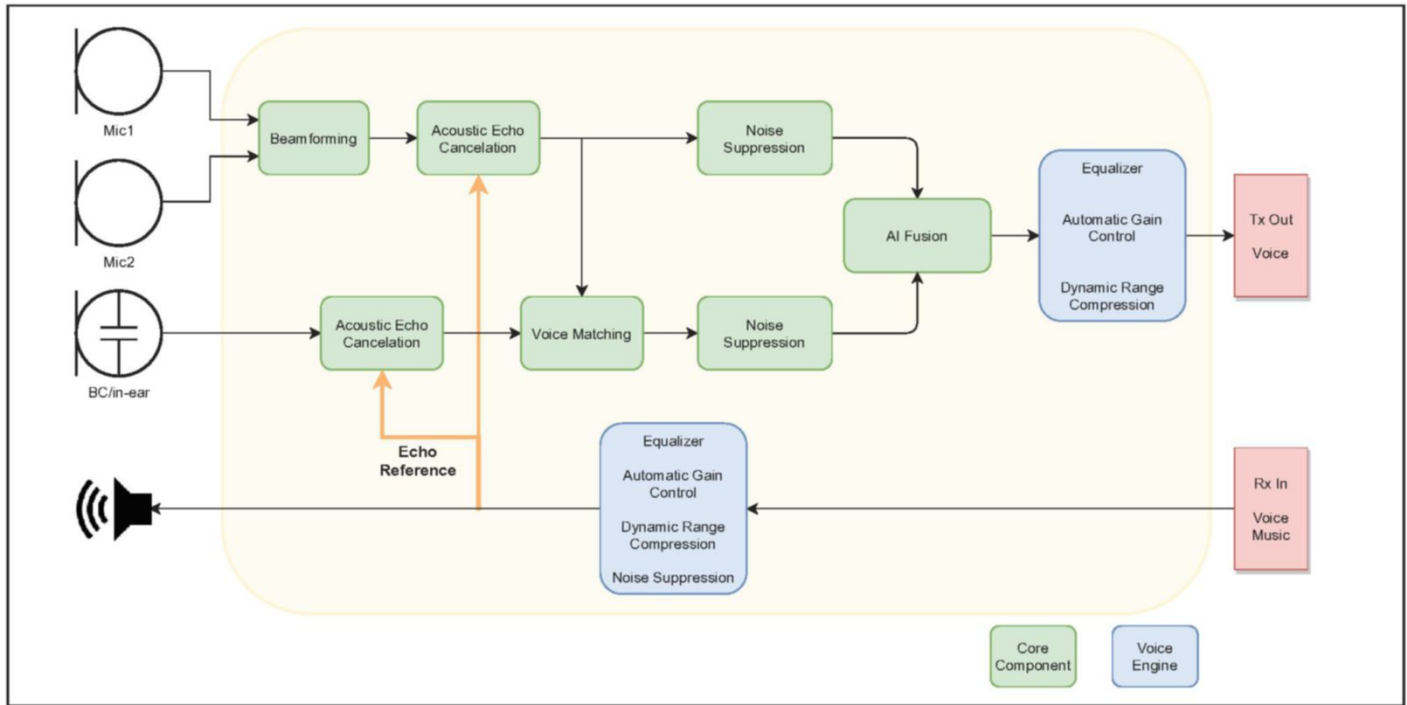


Figure 1: Pure Voice block diagram

does not properly hear you and your message does not come through.

On mid- and low-end devices, voice calling quality has not improved much since the first TWS earbuds because pre-integrated standard platform solutions are used for cost reasons in that segment, leaving the owner with a pair of earbuds that can barely be used to make a phone call.

In this pandemic era, when so many of us work from home, a long-lasting and dependable device is crucial as we spend our days talking with our teams and customers on Microsoft Teams, Zoom, Slack, and others. Users must be able to rely on their earbuds to pick up their voice and filter out background noise, including the most transient sounds such as background voices.

Areas of Improvement for Voice Calling

For every new pair of earbuds, reviewers and consumers have accepted that there is no easy solution, that background noise is always a problem when trying to be properly heard while using earbuds.

When outdoors, wind noise is one of the biggest challenges, especially if no protective mesh or mechanical solution was implemented on the earbud. Wind noise may be one of the most problematic noise types during a voice call (or while listening to music) outside.

Front ends for voice assistants have improved greatly on smartphones and smart speakers. However, consistent voice capture is still rare in TWS earbuds, making it a challenge to reliably interact with your device in all environments.

Today, voice calling is still one of the most power-

hungry applications on an earbud, so footprints are critical to leave room for new use cases and the processing of sensor data for new differentiating features.

7 Sensing's Pure Voice

Compared to existing voice calling solutions, the novelty of Pure Voice is to perfectly leverage the in-ear inward facing microphones and/or bone vibration sensors present in mid- and high-end earbuds. Pure Voice offers excellent voice preservation and dramatically removes background noise (including wind) and noise in speech while keeping very low footprints. **Figure 1** details how the Pure Voice technology works.

The Pure Voice technology uses in-ear microphones and/or Bone Vibration Sensors for voice pick-up. The in-ear microphones are used for feedback noise cancellation and are getting more popular with the increasing presence of hybrid ANC. Bone vibration sensors are also used on high-end earbuds of brands such as Apple, Huawei, and Samsung.

The Pure Voice algorithm automatically adapts to all existing/known bone vibration sensors and in-ear microphones. Neither technologies nor sensors are easy to deal with, as they capture sound in a very nonlinear and distorted way. They also capture all sorts of other noises (e.g., scratching or teeth sounds), and their integration can sometimes create a lot of complexity.

Bone sensors are particularly adapted to open fit or leaky designs with large vents although Pure Voice supports all earbud designs with any of the two technologies.

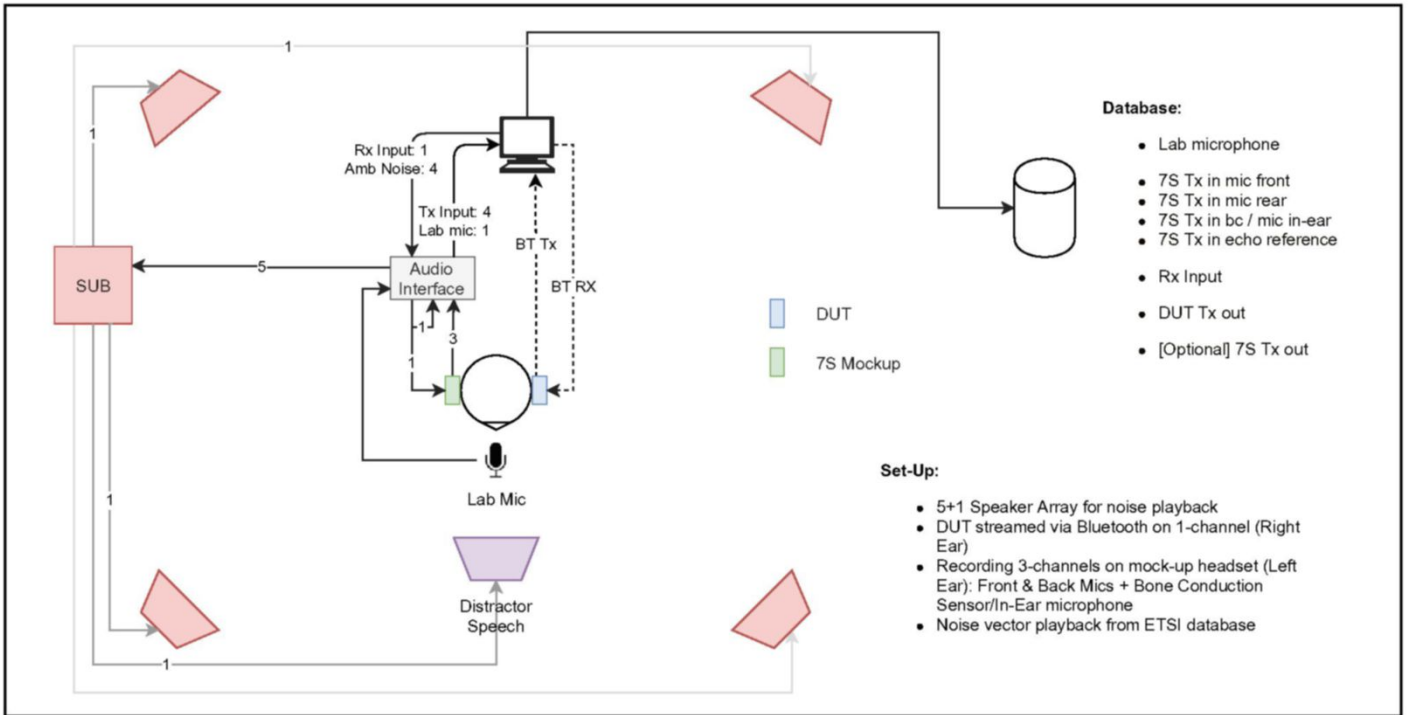


Figure 2: The lab setup used to test Pure Voice

Pure Voice Performance Benchmarks

7 Sensing performed multiples tests to confirm the performance improvements brought by Pure Voice (using devices with two external microphones with one in-ear microphone) over standard devices.

Tests were performed with car noise and pub noise, comparing Apple AirPods Pro and Huawei Freebuds Pro earbuds. During the benchmark tests, one of 7 Sensing's development earbuds was used on one ear, while a commercially available earbud was used on the other ear (**Figure 2**).

The results are very significant. Voice preservation was near perfect while offering almost full and instantaneous noise suppression.

In the first test, 7 Sensing pitted Apple AirPods Pro vs. 7 Sensing Pure Voice. The Apple AirPods Pro have two external microphones, one Bone Vibration Sensor, and one in-ear microphone per earbud. **Figure 3** shows that while the AirPods Pro were not able to remove more noise without degrading speech, Pure Voice removed most of the background noise while still having much more speech content preserved.

Figure 4 shows that with pub noise, which is renowned for being difficult, AirPods Pro maintained a good amount of speech at the cost of not removing much noise. Pure Voice, without any convergence time, removed significantly more background noise while still preserving all the speech content.

In this next test, 7 Sensing pitted Huawei Freebuds Pro vs. 7 Sensing Pure Voice. The Huawei Freebuds Pro have two external microphones, one Bone Vibration Sensor, and one in-ear microphone per earbud. **Figure 5** shows the Freebuds Pro did a good job at removing noise in speech; however, Pure Voice on the right removed all the noise with no convergence required for the noise suppression

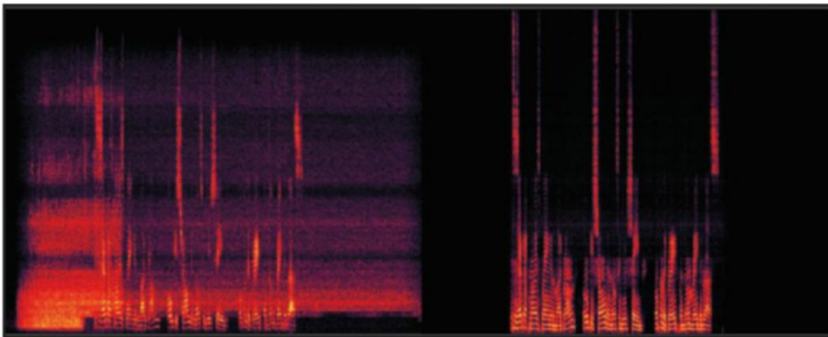


Figure 3: 7 Sensing pitted Apple AirPods Pro vs. 7 Sensing Pure Voice using car noise. The left graph shows the Apple AirPods Pro and the right side shows the Pure Voice.

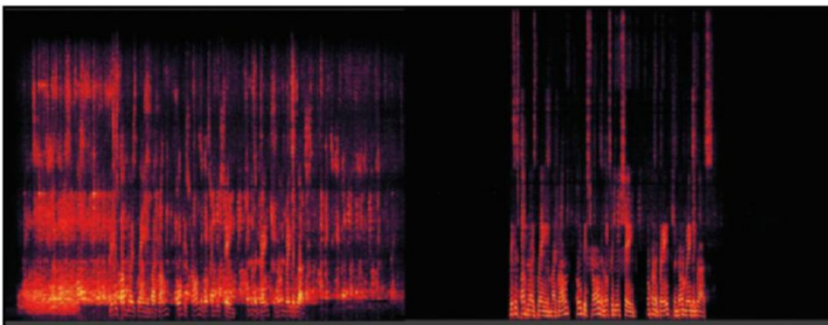


Figure 4: 7 Sensing pitted Apple AirPods Pro vs. 7 Sensing Pure Voice using pub noise. The left graph shows the Apple AirPods Pro and the right side shows the Pure Voice.

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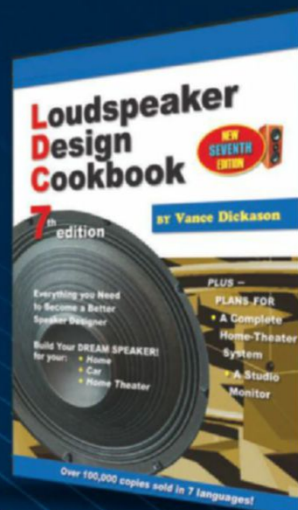
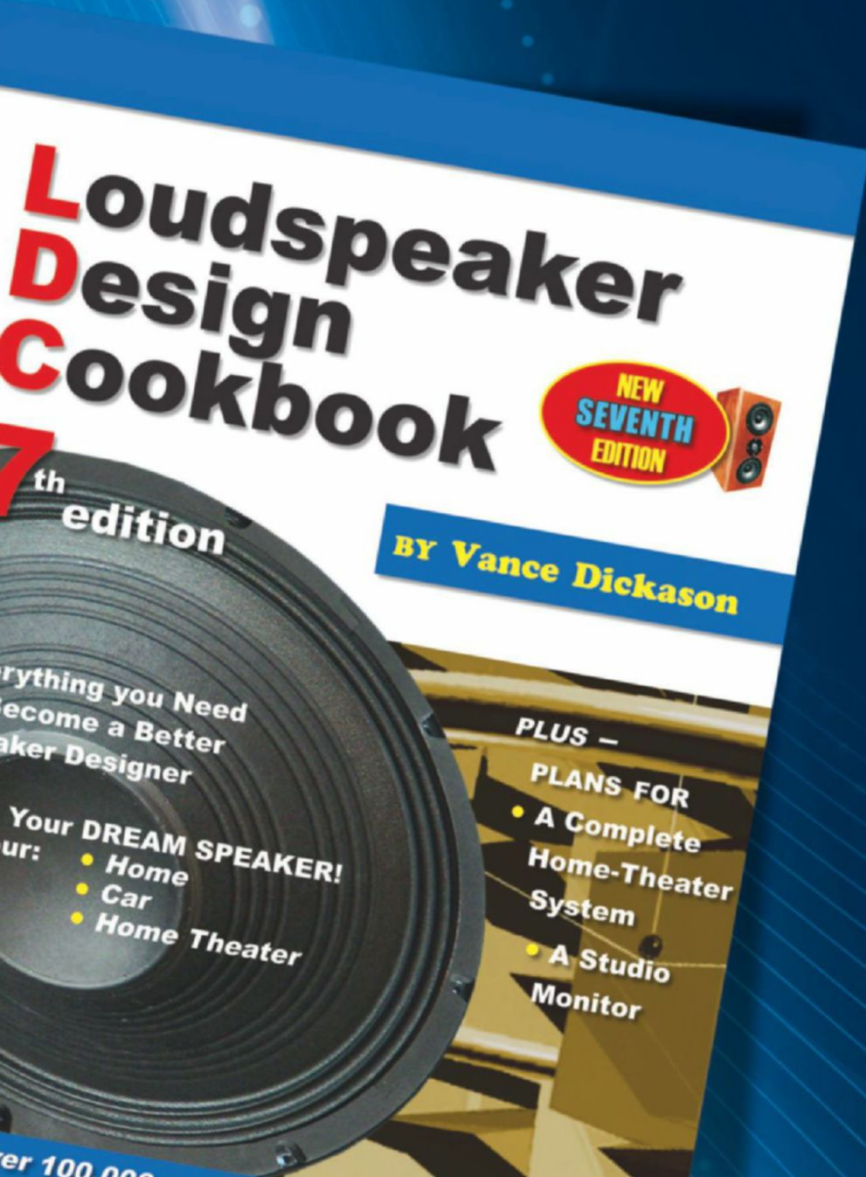
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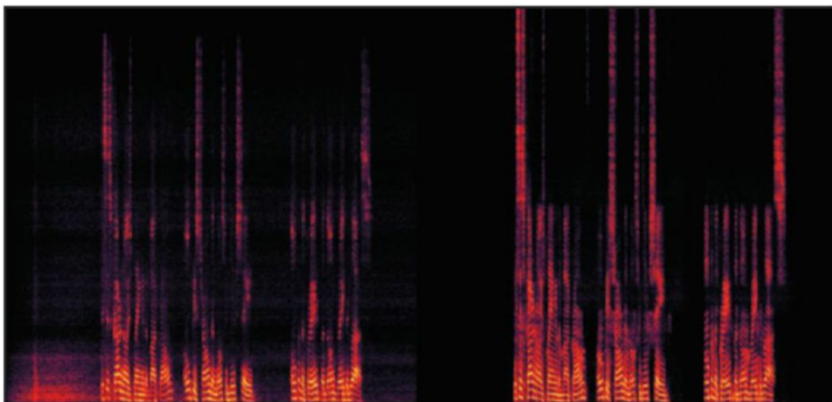


Figure 5: 7 Sensing pitted Huawei Freebuds Pro vs. 7 Sensing Pure Voice using car noise. The left graph shows the Huawei Freebuds Pro and the right side shows the results with Pure Voice.

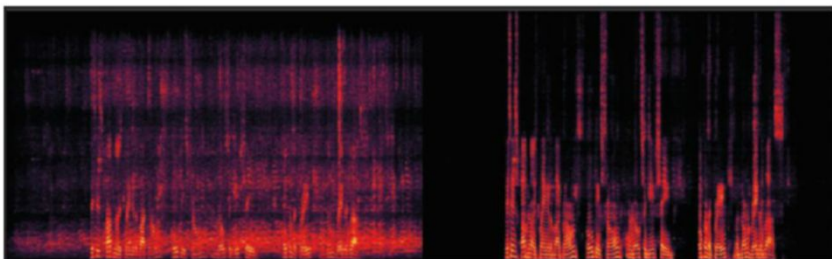


Figure 6: 7 Sensing pitted Huawei Freebuds Pro vs. 7 Sensing Pure Voice using pub noise. The left graph shows the Huawei Freebuds Pro and the right side shows the results with Pure Voice.

algorithm. Pure Voice also shows much more speech preservation in the mid-range.

Figure 6 shows that with pub noise, the Freebuds Pro were not able to remove background noise enough in the low frequencies while heavily smoothing noise and speech in the higher bands. Pure Voice was able to clean the low frequencies efficiently and still maintaining high frequency content.

Super Hearing: Beyond ANC and Transparency

Due to the fact that the comfort of TWS earbuds has improved and they have become trendy fashion accessories, many users now wear them for a large part of their day, opening up the opportunity for a new use case: better hearing.

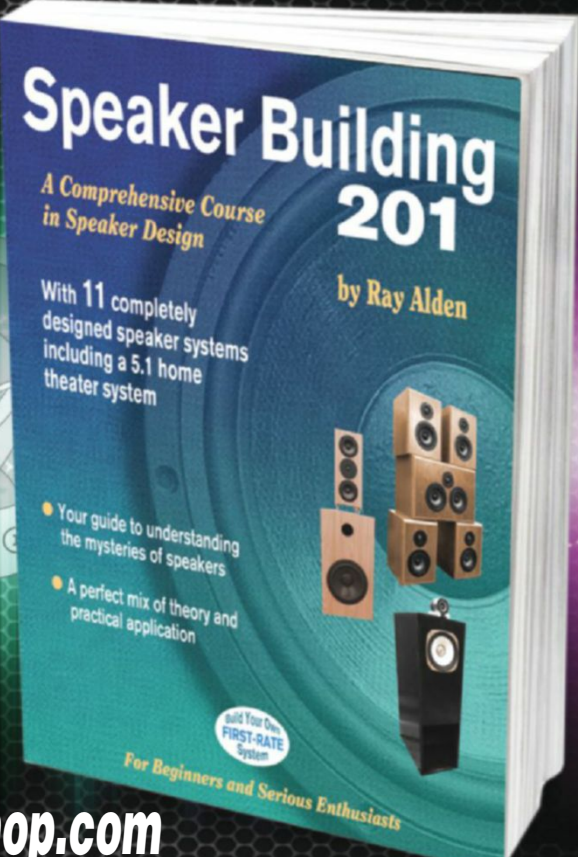
Some earbud products provide a “transparency” or “hear through” feature, however, even normal hearing people regularly face very noisy environments where conversations are challenging. Without transforming them into medical hearing aids, earbuds can greatly improve listening conditions to make conversations crisp and clear.

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thanks to progress in hardware latency, design, and tuning, it is offering ever stronger noise cancellation. However, users sometimes desire the opposite effect of being able to hear and socialize while keeping their earbuds in. Premium earbuds today offer adaptive noise cancellation, setting the sound scene somewhere between transparency and isolation.

Today, transparency, or hear through, is at best

compensating the occlusion effect to offer almost normal hearing. Transparency has been present since the first hearables (Bragi Dash/Headphone), but seldom reaches a real natural feeling of not wearing earbuds. In transparency mode, no distinction is made between noise and information/content—thus yielding no improvement of a user's existing hearing capabilities.

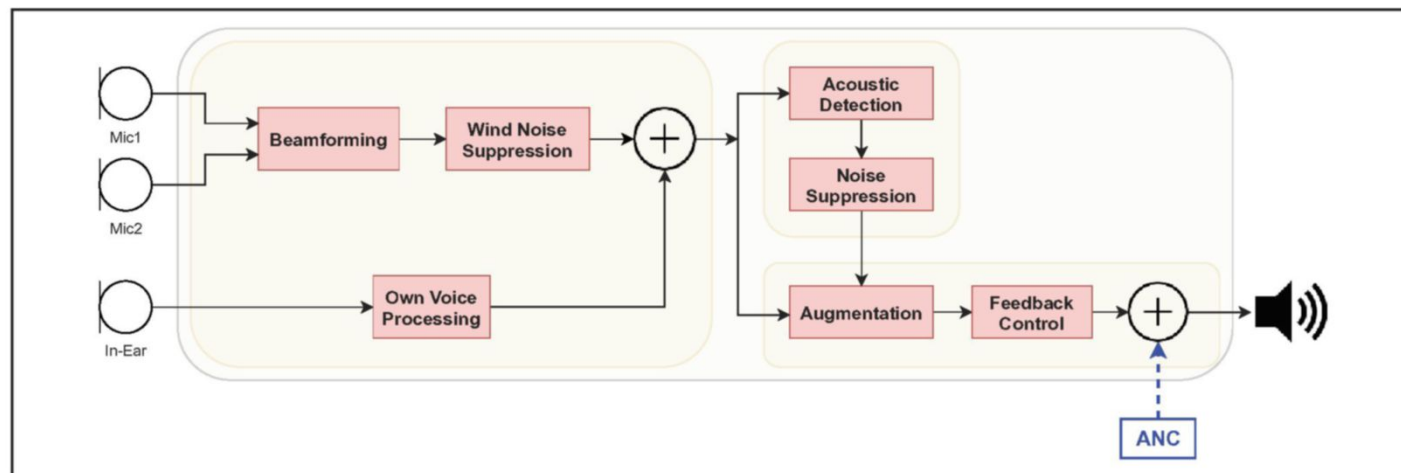


Figure 7: The Hearing Augmentation block diagram

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What if a TWS earbud could actually enhance the wearer’s capabilities and give him or her super-hearing? By combining techniques used in expensive medical-grade hearing aids with modern highly integrated and powerful consumer TWS chips, this has become a real possibility.

Amplification, noise reduction, and speech

enhancements add high-end differentiating features to an earbud product in a crowded market—yet without the stigma of wearing a medical device.

Market Traction for Augmented Hearing Features

Some key devices have already started implementing features to hear better. Among the top contenders are the Nuheara IQ Buds2 Max, the Apple AirPods Pro, and the Huawei Freebuds Pro—however, each one comes with its own limitations.

The Nuheara IQ Buds2 Max offer beamforming, noise reduction, and amplification. However, they offer limited conversation and speech enhancement features, and they don’t offer any voice perception improvement, which is crucial when wearing that kind of device nearly all day. They remain the most compelling offer feature-wise on the market, albeit they are positioned in the hearing assistance segment, or so-called up-and-coming over-the-counter (OTC) hearing devices.

With the accessibility features on iPhone released with iOS 14, Apple AirPods Pro offer tailored amplification without speech enhancement, beamforming, or noise reduction. AirPods Pro can thus be turned into personal sound amplifiers without any stigma of wearing an assistance device.

With their Voice Mode, the Huawei Freebuds Pro offer a conversation mode with limited noise reduction. However, they don’t offer beamforming or any kind of amplification.

Noise Cancellation/ANC is for music & video immersion and isolation Transparency is for ambient awareness Hearing Augmentation - or advanced transparency – is to hear more, hear better			
Features and Benefits	Hearing Augmentation	Apple AirPods Pro ⁽¹⁾	Huawei Freebuds Pro ⁽²⁾
Amplification for hearing more	👍	👍	👎
Smart Noise reduction for comfort and increased awareness	👍	👎	👎
Beamforming for greater focus	👍	👎	👎
Low Latency for naturalness	👍	👍	👍
Voice Focus for clearer conversations in noise	👍	👎	👍
Voice Detect for automatic switch to Voice Focus	👍	👎	👎
Low Footprints for all-day wear	👍	---	---

• (1) Apple AirPods Pro on iOS14 with Headphone Hearing Accommodation accessibility
 • (2) Huawei Freebuds Pro on AI Life with Voice Mode

Figure 8: 7 Sensing compared its Hearing Augmentation’s features and benefits to that of Apple’s AirPods Pro and Huawei’s Freebuds Pro.

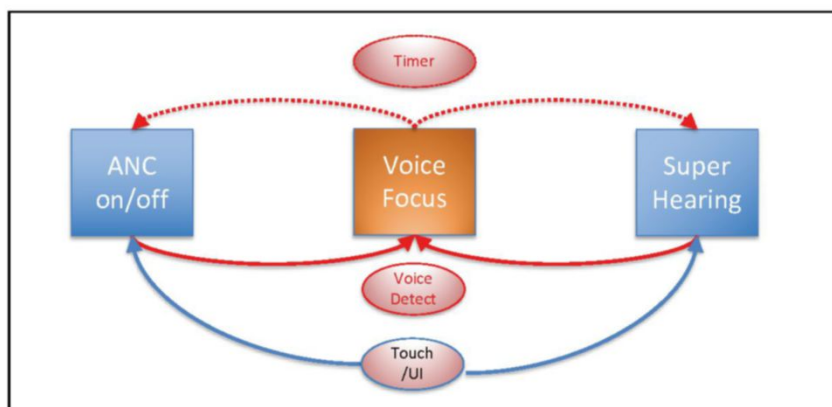


Figure 9: 7 Sensing’s Hearing Augmentation automatically senses mode changes.

About the Author

Jean-Marc Luneau is an audio DSP engineer focusing on engineering and product management. With more than 15 years of experience in research and development, he has worked on speech, music compression and voice enhancement, and biomedical imaging at Fraunhofer IIS, EPFL, Aalborg University and NXP Software, before co-founding 7 Sensing Software in 2017 in Leuven, Belgium.



7 Sensing Software

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7 Sensing Hearing Augmentation (HA)

To take the capabilities of TWS earbuds to the next level, 7 Sensing offers its Hearing Augmentation (HA) software solution. This suite of signal processing algorithms makes it possible to not just block out the world—as with ANC—but instead control exactly what the users want to hear for extra content awareness and elevated conversations, whenever they wish.

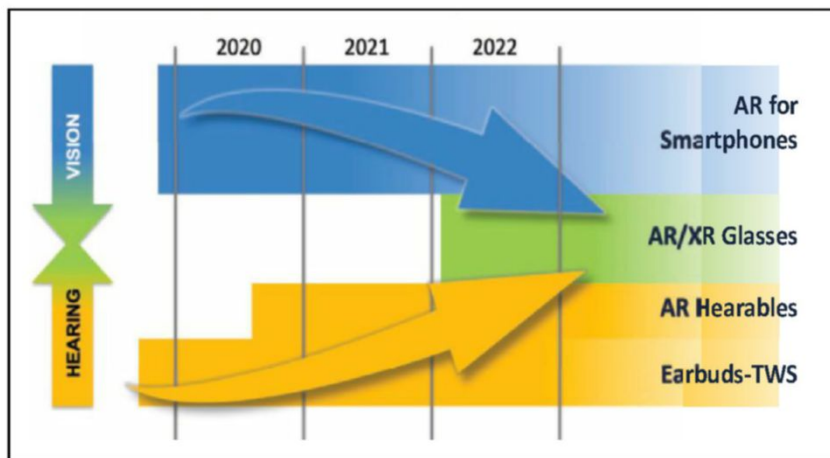
An earbud with 7 Sensing’s HA allows for new designs with super hearing capabilities, where users are able to comfortably have a conversation with other people in a noisy restaurant. Likewise, wearing the same design, users can enjoy their favorite music while jogging and still be able to hear sounds around them for safety awareness, tune in to a lecture for total concentration, or enjoy a music concert at safe levels.

HA automatically switches from one mode to the other. If the earbud picks up the voice of the user indicating the start of a conversation, it can automatically pause music playback and switch to Hearing Augmentation.

Combining passive isolation and noise cancellation, HA is layered upon and mixed into the existing audio path and takes advantage from maximum isolation available on the earbud. This is usually a combination of ANC in low frequencies (< 1kHz) and passive attenuation (or insertion loss) above. **Figure 7** demonstrates how the HA process works.

7 Sensing offers two HA modes. The first mode is Super Hearing for extended perception with controllable amplification, beamforming and real-time noise reduction. The second mode is Voice Focus, which is ideal for conversations. It enhances the speech spectrum to narrow down your listening sphere for absorbed discussions. To demonstrate HA's capabilities, 7 Sensing compared its features and benefits to that of Apple's AirPods Pro and Huawei's Freebuds Pro (**Figure 8**).

In a quest to keep Hearing Augmentation simple to use, 7 Sensing also implemented an automatic mode change with a dedicated Voice Detect feature. Voice Detect makes any earbud mode—Music (ANC on/off), Transparency, or Super Hearing—automatically switch to Voice Focus as the owner starts speaking. And after some time without speaking, the previous mode automatically resumes. An example of



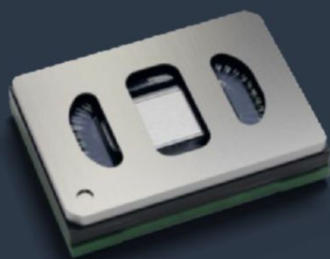
7 Sensing's Hearing Augmentation mode changes is shown in **Figure 9**.

Figure 10: 7 Sensing's technology has positioned itself to explore the exciting future of Augmented Sensing in earbuds, headsets, and augmented reality (AR) devices.

Future Uses

Combined with its research on audio features for TWS earbuds, 7 Sensing is also working in augmented sensing. The company has a strong team dedicated to Machine Learning—exploring applications with vision, depth, spectral and light sensing (**Figure 10**). The technology complementarity of AI/Machine Learning, Vision, and Audio ideally positions 7 Sensing to explore the exciting future of Augmented Sensing in earbuds, headsets, and augmented reality devices. 

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One Traditional Way to Increase Loudspeaker Directionality

Since their first use in acoustic phonographs, a great deal of research has been done on acoustic horns. Much of it has focused on the effect of different flare profiles and rates upon the variation of efficiency with frequency. In this article, Richard Honeycutt discusses the use of acoustic horns to increase loudspeaker directionality.

By
Richard Honeycutt

Although the first acoustic horns (used in early phonographs) were primarily employed to increase sound level by increasing efficiency, they have been used in the last eight decades or so mainly to control directivity. Particularly since the advent of the “constant directivity” (CD) horn in the late 1970s, horn speakers have been the go-to solution for directional control of sound radiation.

Horns

An acoustic horn is a device in which the sound is channeled through a gradually expanding cross-sectional area from the driver end (throat) to the radiating end (mouth). At very low frequencies, the horn is essentially acoustically transparent, and the driver acts as if the horn were not present. Then above a certain cut-off frequency, the horn begins to act as an acoustic transformer, converting the low acoustic impedance (low-pressure, high-volume velocity) mouth termination into a high acoustic impedance (high-pressure, low-volume velocity) at the throat. This transformation greatly increases the efficiency of the radiating system, often by 12dB or more, and reduces excursion, and thus excursion-related distortions (the most common kind).

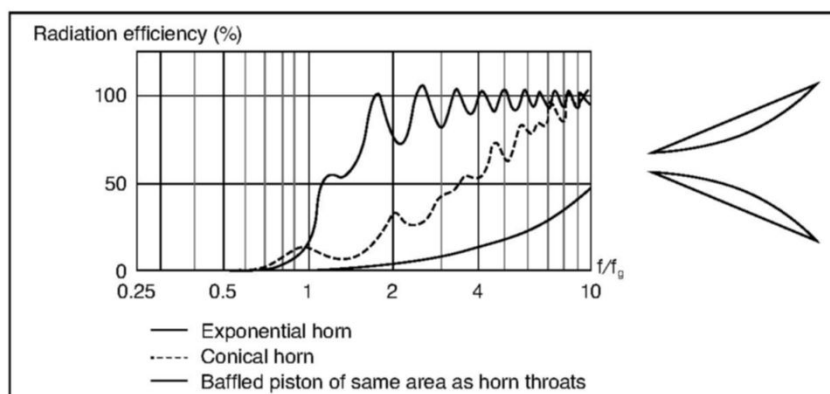


Figure 1: This graph compares the radiation efficiency of different horns.

In between the mouth and the throat, the manner in which the cross-sectional area increases as the sound travels down the horn affects both the cut-off frequency and the directional radiation. **Figure 1** [1] shows plan views of a conical and an exponential horn, along with graphs of their radiation efficiency vs. normalized frequency. Other than to note that the conical horn requires a much higher frequency to produce its impedance transformation, we will not dwell much on the loading effects of horns: That is a subject for a separate treatment. However, we will examine the horn’s directional radiation.

Since their first use in acoustic phonographs (**Photo 1**), a great deal of research has been done on acoustic horns. Much of it has focused on the effect different flare profiles and rates have upon the variation of efficiency with frequency. Many of the earliest horns had a conical flare (**Photo 2**), for ease of manufacture. However, manufacturers soon discovered that other flares produced better bass, due to having better loading in the three octaves above the cut-off frequency. Several flares were explored, beginning with the seminal paper by Arthur Gordon Webster [2], and included conical, exponential, and hyperbolic (which is really a family of flares). In 1941, Vincent Salmon published a paper [3] in which he combined the most widely used flares—exponential and hyperbolic—into a family that he dubbed “Hypex horns,” a name subsequently trademarked by the Jensen Manufacturing Co. (**Figure 2**).

Analyzing horns’ directional radiation can be discussed in three frequency regions (**Figure 3**). [4] In the lowest region, radiation changes from omnidirectional to directional exactly as it would for a loudspeaker cone the size of the horn mouth. In the middle region, the horn directs sound as one would expect by examining its walls: fairly constant directivity for flat walls and increasing directivity with frequency for curved flares. There are two non-obvious features to be aware of here,



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Photo 1: The Victrola acoustic phonographs used horns to increase the loudness.



Photo 2: This Edison phonograph used a conical horn with a flared mouth.



though. The first is that just above the mouth control region, the directivity may narrow for about half an octave or so; this is called “waist-banding.” The second is that if the horn walls are not straight in profile, the horn will beam at higher frequencies.

Essentially, for about the first half-wavelength of travel, a sound wave conforms to the surface of the horn walls. Then, even if the walls diverge, as in an exponential horn, the sound waves no longer follow the walls. Thus, horns with very slow expansions at the throat will be much more directional at high frequencies than at low frequencies. Such horns may have wide exit angles (the angle formed by the walls at the horn’s mouth).

In the past, when horn driver diaphragms were relatively heavy—aluminum vs. modern titanium diaphragms, for example—the driver’s high-frequency response tended to drop off above a few kilohertz.

Figure 4 and **Figure 5** show a comparison of drivers with aluminum vs. titanium diaphragms. Beaming was

actually useful in this case, since it boosted on-axis high-frequency response.

The so-called radial horn invented by Paul Klipsch (as the K-5-J midrange horn [5]) exploits this fact by maintaining a constant conical flare in the horizontal plane, with an extreme deviation from conical in the vertical plane. The overall expansion of the flare is exponential. Thus, the horizontal directivity is almost constant with frequency, but the horn beams very significantly in the vertical plane. Such horns are valuable in many applications where a 10° or 20° vertical radiation is adequate, but a 90° or wider horizontal directivity is needed.

The top region of horn directivity is that produced by the directional characteristics of the throat, generally as defined by the opening where the compression driver is mounted. Thus, a horn with a 25mm driver can have a wider high-frequency radiation pattern than will a horn with a 50mm driver. Transition to this region takes place somewhat above the frequency where the driver’s circumference equals the wavelength: about 5kHz for a 25mm driver or 2.5kHz for a 50mm driver. Of course, the frequency at which the transition actually occurs is determined by the wall angle of the horn (at the throat) as well as by the driver opening.

In order to improve the constancy of both horizontal and vertical directivity, reversed-flare horns [6], “conical-exponential or CE horns [7], and Manta-Ray horns [8]—all dubbed “constant-directivity or CD horns—were introduced. These designs succeeded in improving directivity control, but at the expense of introducing discontinuities in the flare profile that

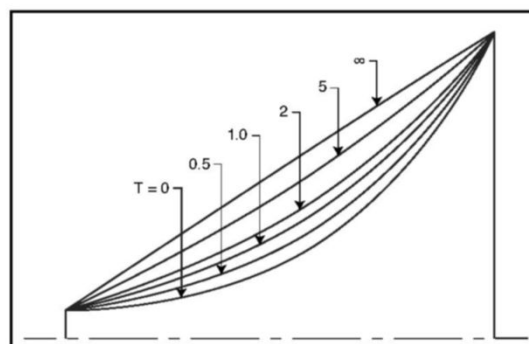


Figure 2: These flares comprise examples of the Hypex horn family: the curve marked $T = \infty$ represents the conical horn; $T=0$ represents the exponential horn, and the other flares are various hyperbolic horns.

resulted in frequency-response anomalies.

Oblate Spheroidal [9] and Tractrix [10] flare profiles have also been investigated, as has the effect of cross-sectional shape (circular, rectangular, oval, etc.). In each case, the guidelines given above concerning directionality still apply. It is important to keep in mind that no horn provides absolutely constant directivity. **Photo 3** illustrates the polar patterns of a two-horn speaker having a rated frequency range of 250Hz to 20kHz. Note the narrowing of the directivity at higher frequencies.

One way of increasing the high-frequency coverage angle is by the use of “bullets” in front of the driver’s phase plug. (The phase plug is a component of the driver that ensures that energy from all portions of the diaphragm exits the driver in phase.) Simulations and measurements of horn operation indicate that these devices introduce frequency-response irregularities.[11]

Where multiple horns are arrayed to cover the same frequency range, interference effects will cause response irregularities in the overlap zones. The closer together the horn drivers are located, the higher the frequency at which these irregularities will

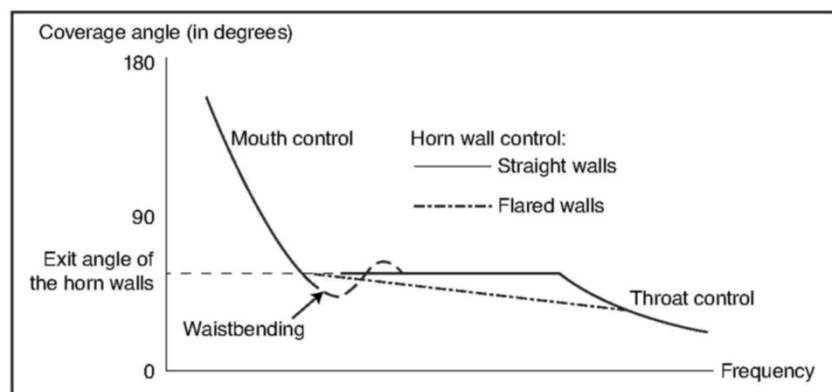


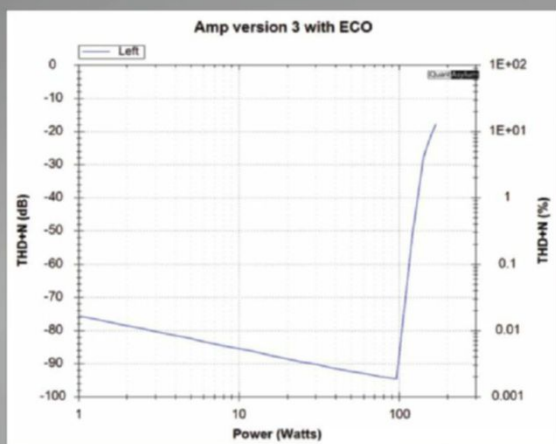
Figure 3: The directional coverage of horns can be discussed in three frequency ranges.

take place, and thus, the less objectionable they will be. Many designers deliberately underlap the horn patterns in areas where both horns will cover the audience, reasoning that slight droops in the level (meaning lower direct-to-reverberant sound ratio for the audience) are less objectionable than the response irregularities resulting from interference effects.

Near and Far Fields

The behavior of the sound field radiated by any source depends upon the listening distance. Generally, listening distances can be separated into the near field and the far field. The near field extends from the surface of the radiator to the greater of these distances:

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$$\left[\frac{L^2}{4\lambda} \right]$$

where L is the longest dimension of the source: the distance, L , and the wavelength λ are all measured in the same units. In the near field, the sound level from the source decreases only slowly with distance. In the

far field, extending from $L^2/4\lambda$ to infinity, the sound level from the source decreases at a rate of 6dB for each doubling of distance if the source is outdoors. Inside, room effects cause a slower decrease—somewhere between 3dB and 6dB per doubling of distance. These effects are important for performance measurement, in which the microphone should be in the far field for all frequencies of concern, for accurate results. They are also important in the evaluation and application of speaker arrays.

Rated Directivity and the Real World

Horns and even complete speaker systems including horns are often rated in terms of vertical and horizontal “dispersion.” First, the term “dispersion” is inappropriate here: Properly used, “dispersion” means the frequency-dependent velocity of waves in a waveguide or medium. The proper term for the angular spread of sound produced by a loudspeaker is “directivity.”

Commonly, directivity is expressed as $H \times V$, where H is the horizontal directivity angle in degrees and V is the vertical angle. A common example is $90^\circ \times 40^\circ$, the rating for many “speakers on a stick” used for portable live-sound applications.

As you might have deduced from Photo 3, even if we knew the frequency for which the rated directivity is valid, we would still lack valid information about the speaker’s directional response. For a portable speaker, this may not be very important, since the operator can set up the speaker(s) “by guess and by gosh”, then listen to recorded music on the system as (s)he walks the venue.

Installed speakers are another story. Even if the system designer/installer carefully analyzes the geometry of the venue and projects the $H \times V$ directivity accordingly, the results are unlikely to be as desired.

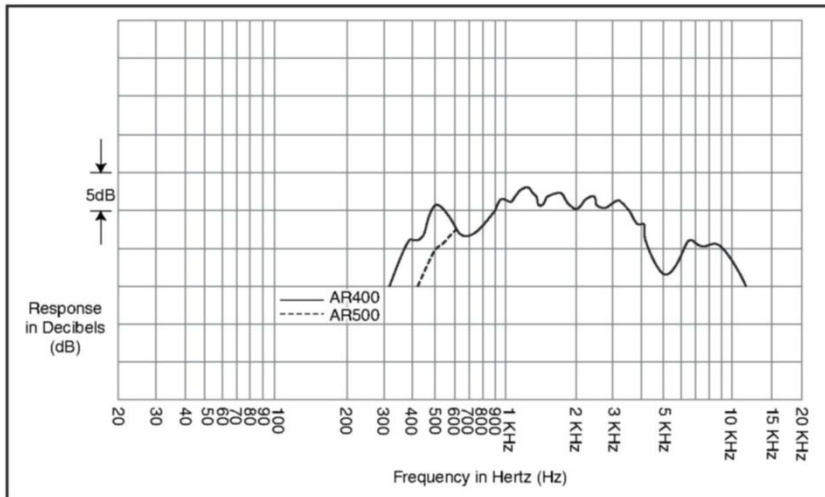


Figure 4: Note that the response of this aluminum-diaphragm horn driver drops off above 4kHz. Response curves are shown for two different horns with the same driver.

About the Author

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



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Beginning in the early 1980s, the Prohs Global Mapping program was used to predict loudspeaker coverage in the venue audience area. This program was developed into the Prohs-Harris Design (PHD) program for the DOS operating system.

Today, there are several excellent acoustical/electroacoustical modeling programs that can predict speaker coverage across the frequency range of at least 100Hz to 16,000Hz. These programs use measured speaker data that manufacturers provide in either the EASE or CLF format, and thus permit designers and installers to be quite accurate in mounting speakers to cover the audience evenly. 

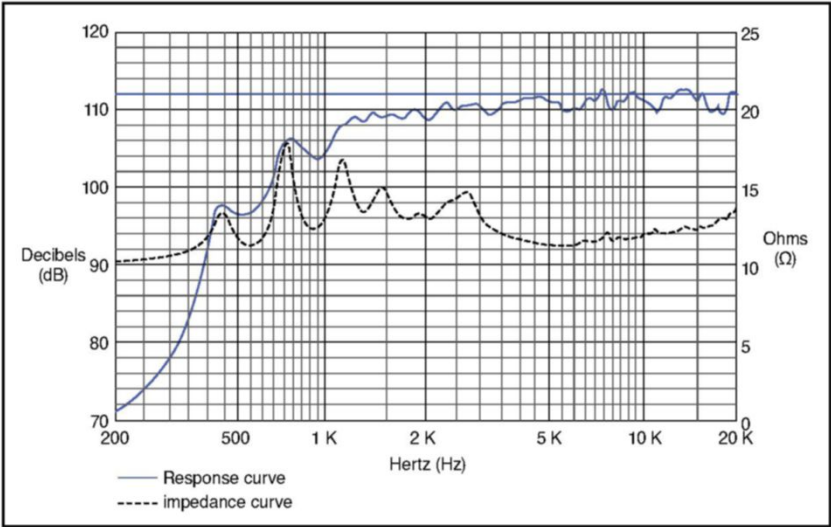


Figure 5: Note this titanium-diaphragm driver’s flatter frequency response above 4kHz.

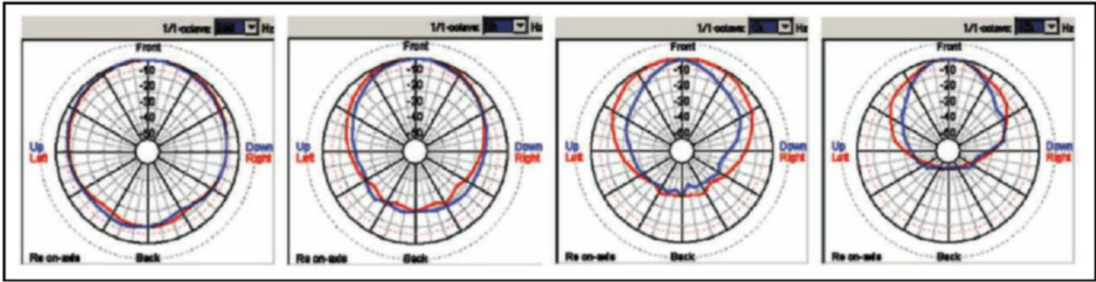
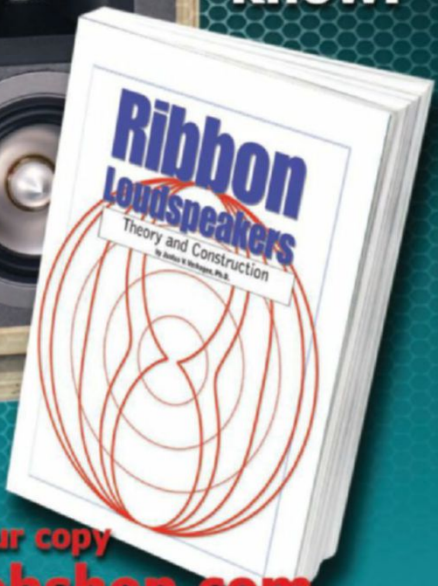


Photo 3: Polar patterns are shown for a two-horn vertical array at 250Hz, 1kHz, 4kHz, and 16kHz.

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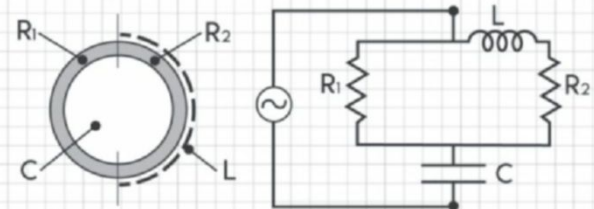


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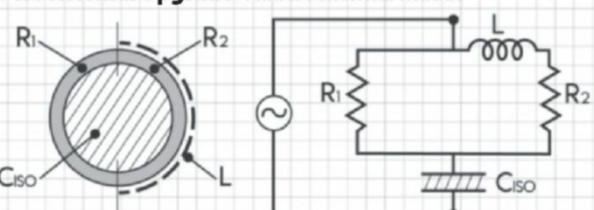
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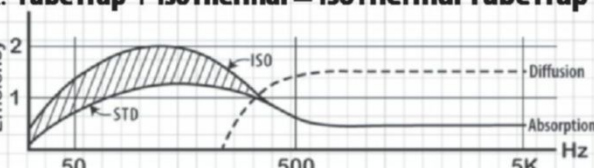
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Improving the Input Noise Model for a High Output Current Dual Headphone Driver

By
Michael Steffes

In this article, our audio expert address how to improve the input noise model for a line driver using a TINA model, enabling a more accurate prediction of integrated output noise.

An audio version of an earlier asymmetrical digital subscriber line (ADSL) driver was introduced in 2004 as the TPA6120A2 (Reference 1). This device was later put into a much smaller package and the price reduced for the audio market. However, it never provided a distinct simulation model but we can use the TINA model for the earlier THS6012 (Reference 2) to test and correct its input noise model. This will allow a more accurate prediction of integrated output noise over the audio band under any set of external conditions.

Lacking any other information, it is usually safe

to assume a reasonable looking spot noise plot over frequency, like that shown in **Figure 1** (Reference 2, Figure 16), was probably measured and has more veracity than later simulation models. It is normal for current feedback amplifiers (CFA) to have relatively high inverting input current noise. The higher input current noise on the non-inverting input (I_{n+}) indicates a diode input Class AB buffer across the inputs. This reduces the input voltage noise where a base input Class AB buffer would have much lower non-inverting input current noise but slightly higher voltage noise. The diode input buffer is a good choice for higher gain, lower source impedance designs.

The voltage noise shown in Figure 1 suggests a flatband value of $1.7\text{nV}/\sqrt{\text{Hz}}$ with a $1/f$ corner frequency at 100Hz . Testing the THS6012 TINA model for input voltage noise yields a $1.97\text{nV}/\sqrt{\text{Hz}}$ flatband with a 75Hz $1/f$ corner. While a bit high, it is not possible to add a negating external voltage noise to reduce this flatband. It is possible to modify the transistor model's internal to the THS6012 model to reduce this voltage noise but that is beyond the scope of this development. To test the model input current noise, simply place two current meters into each input and run an output spot noise simulation. Both appear to be very low relative to the measured plot as shown in **Figure 2**.

These can both be tuned to match the measured plot of Figure 1 using an external current noise macro. Those external noise terms will RMS with the internal model shown in Figure 2. **Table 1** shows the required external parameters to adjust Figure 2 to Figure 1 by solving for the correct RMS values at 100kHz and 10Hz . This is simply taking the square root of the target value squared minus the internal model value squared.

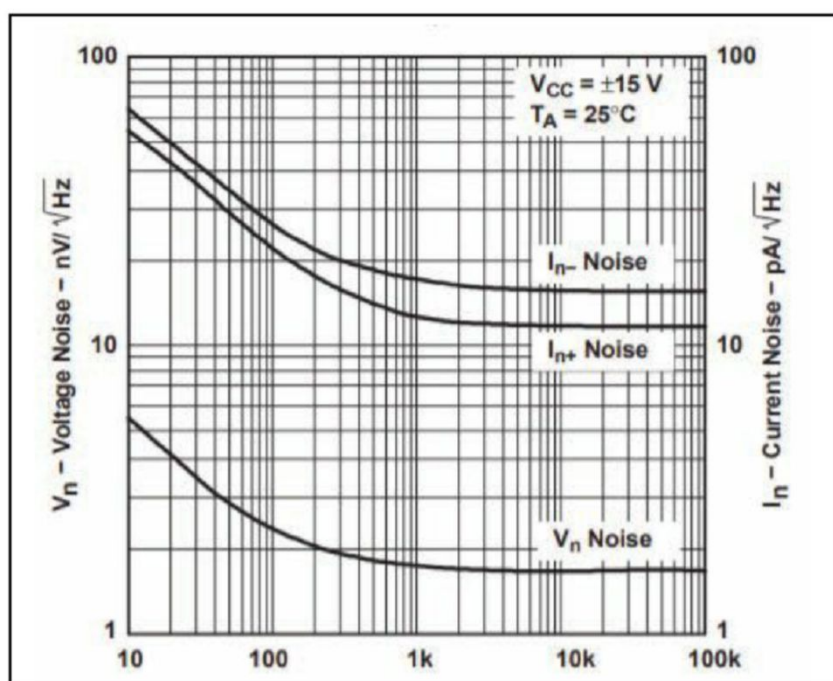


Figure 1: THS6012 Input noise plots from the datasheet

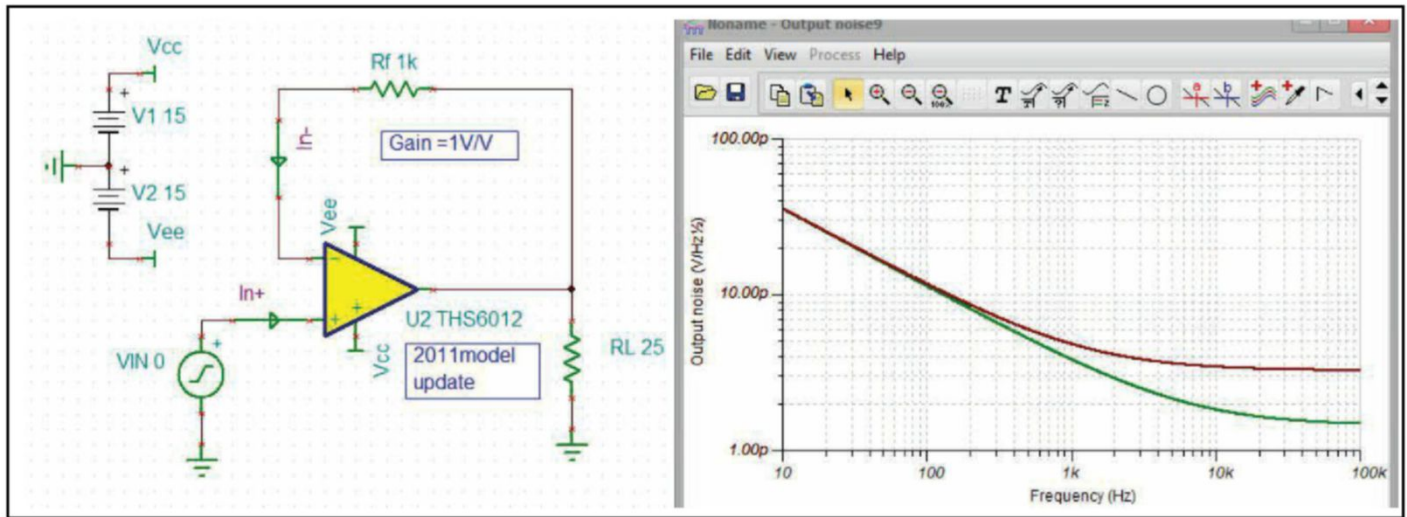


Figure 2: Simulated input current noise for the THS6012 TINA model

The current noise macro is adjusted (Reference 3) going into the netlist and setting the flatband value, the low-frequency specification point (10Hz here), and the current noise value at that low frequency. These macros (Reference 4) were adjusted to those red values in Table 1 and added externally to the core THS6012 model to produce the input current noise simulation of **Figure 3**—matching the measured results of Figure 1.

Now what does this mean for the noise specification in the TPA6120A2 datasheet? That 0.9 μ V_{RMS} is specified at a gain of 1 with no further information. Going back to the original release files yielded the AP test setup of **Figure 4** for this measured output noise. The simulated 1.26 μ V_{RMS} integrated noise from 22Hz to 22kHz is a bit higher than the reported 0.9 μ V_{RMS} but more accurate than the 0.67 μ V_{RMS} using just the original THS6012 model.

Comparing Measured SNR to the Corrected Simulation Model

The reported THD +N curves in the TPA6120 datasheet (Reference 1, Figure 1) are likely Audio Precision measurements. Taking the 10V_{pp} (3.53V_{RMS}) curve at 10kHz shows an SNR dominated result of -108dBc given the very light load of 10k Ω . Converting

	Non-Inverting In+	Inverting In-
100kHz Model Value	1.5pA/Hz	3.3pA/Hz
100kHz Target Value	11.5pA/Hz	16pA/Hz
100kHz External Value	11.4pA/Hz	15.7pA/Hz
10Hz Model Value	36pA/Hz	36pA/Hz
10Hz Target Value	55pA/Hz	62pA/Hz
10Hz External Value	41pA/Hz	50pA/Hz

Table 1: Required external current noise terms to adjust model to datasheet

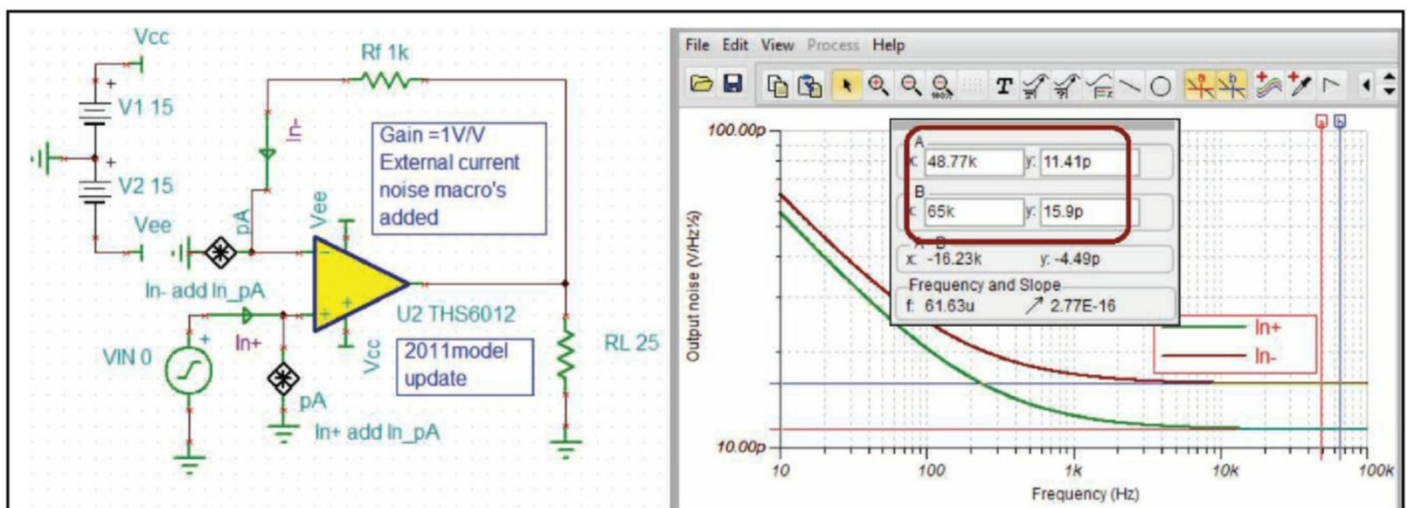


Figure 3: Input current noise including external macros to adjust model to measured data.

that SNR to an implied integrated noise over the stated 80kHz span gives a $14\mu\text{V}_{\text{RMS}}$ integrated noise through 80kHz. Using the corrected input current

noise model and the stated conditions provided for Reference 1, Figure 1 gives the integrated noise of **Figure 5**, closely matching that result.

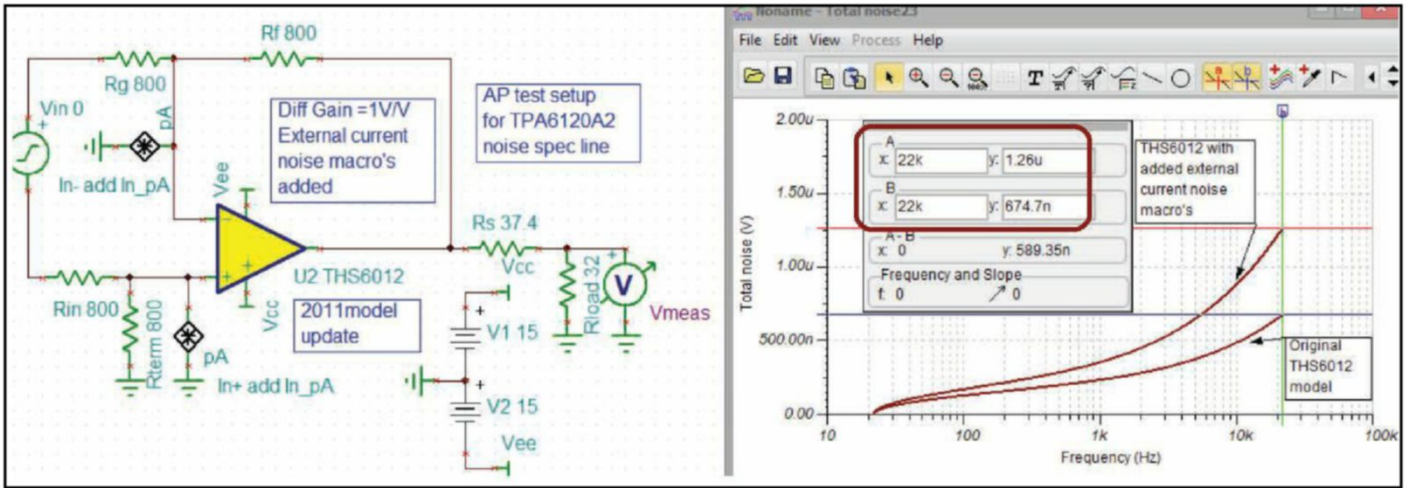


Figure 4: Integrated noise comparison with 1kΩ RF at gain of 1V/V

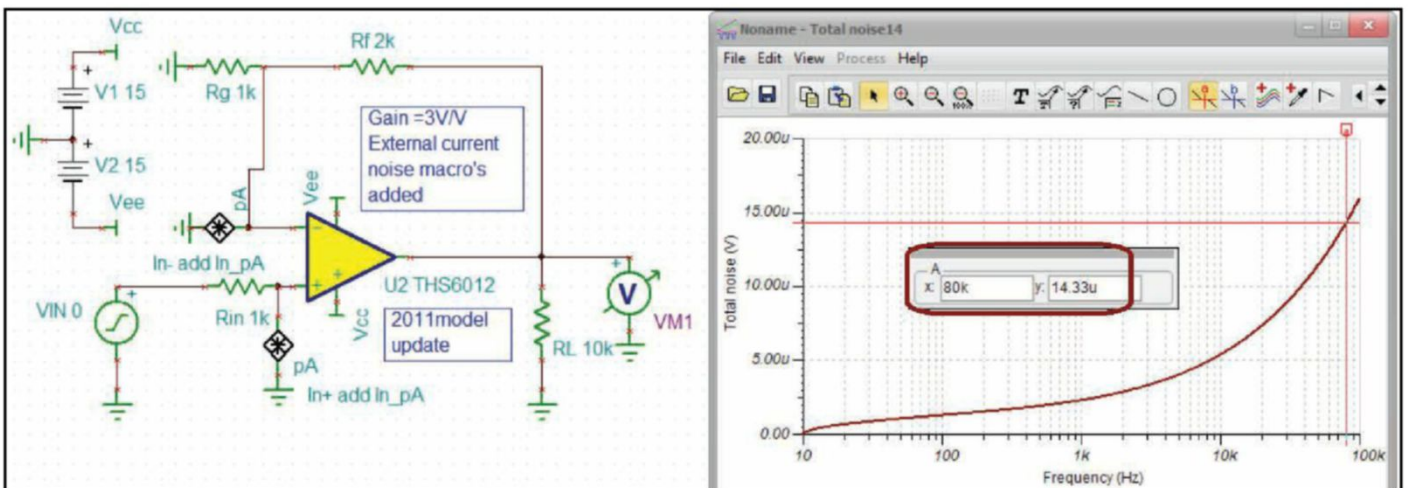


Figure 5: Corrected noise simulation for AP measurements of Reference 1, Figure 1

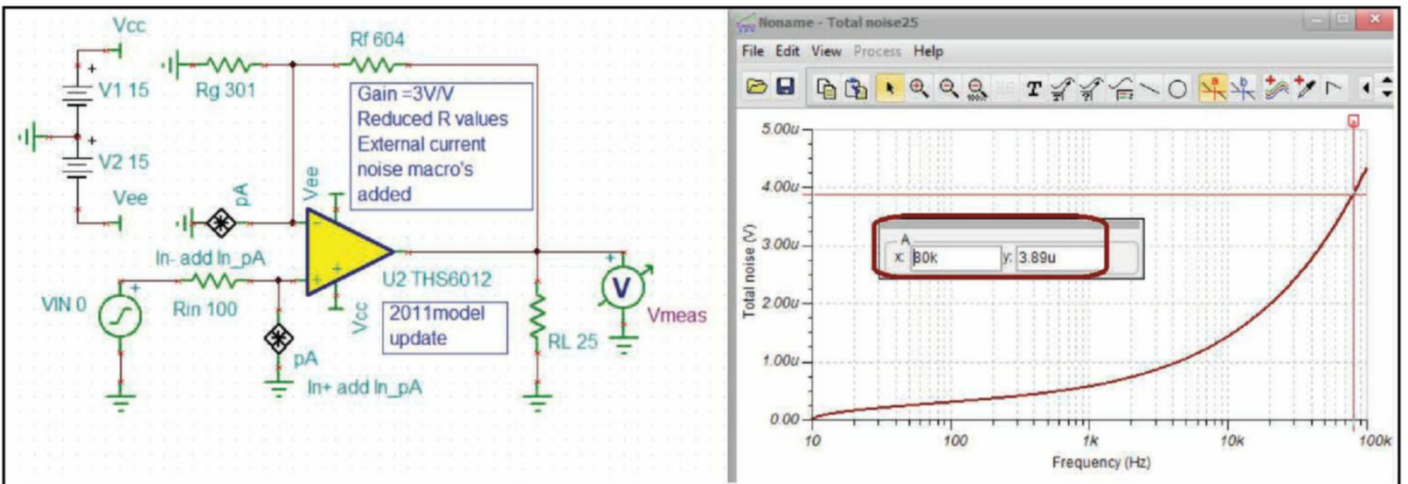


Figure 6: Lower noise implementation for the test circuit of Reference 1, Figure 1

Since the THS6012 TINA model was erroneously low on its input current noise, the TPA6120A2 introduction was likely unaware how much degradation those higher R values in Figure 5 were causing. Adjusting those R values to a more reasonable (lower noise) level gives the integrated output noise of **Figure 6**. This $3.89\mu V_{RMS}$ through 80kHz would have improved their measured data in Reference 1, Figure 1 ($10V_{pp}$ output) from -108dBc to -119dBc !!! Running a small signal bandwidth (SSBW) simulation for the circuit Figure 6 gives a very flat response with >120Mhz bandwidth.

The TPA6120A2 has proven very popular as a highly linear audio headphone driver. The THS6012 simulation model provides a good substitute model for design simulation. Adding the correct external current noise macros will yield much more accurate spot or integrated noise predictions. These noise corrections are unnecessary when the TPA6120A2 is imbedded inside another op-amp loop as a high power output stage (Reference 5) as that overall loop will correct the noise and offsets for the imbedded output stage. 

About the Author

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



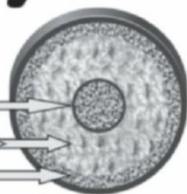
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Measuring Vibration with Piezoelectric Sensors (Part 1)

The Research

By
George Ntanavaras



Several years ago, George Ntanavaras designed a system consisting of the ACH-01 accelerometer and an amplifier with an integrated analog signal processor. Over the years, he has been looking for other possible solutions concerning the vibration measurements. In this article, he describes his research for the design, the construction of such a system, and presents the results and the measurements.

Several years ago in this magazine, I published an article describing the design and operation of a system consisting of the ACH-01 accelerometer and an amplifier with an integrated analog signal processor (see Resources). The cost of the system is very reasonable and proved to be popular among the amateur and professional loudspeaker builders who wanted to investigate vibration issues especially in loudspeaker enclosures. I made the PCB and the front panel of this amplifier available to DIY builders and a lot of amplifiers have been built around the world.

The ACH-01 is a general-purpose, linear single-axis accelerometer housed in a small, rugged, flat package and manufactured by TE Connectivity. According to its datasheet, it is internally buffered for lower output impedance and has a very wide frequency response, typically from 2Hz to 20kHz, with the worst case -3dB response from 5Hz to 10 kHz. It can measure up to approximately 150g and has a high resonance frequency at 35kHz. The linearity is typically 0.1% with a maximum

deviation of 1%. Its output provides an analog voltage directly analogous to the acceleration that it measures. The ACH-01-03 version of the accelerometer is supplied with a shielded cable and a convenient connector.

The amplifier bias the internal junction field effect transistor (JFET) of the ACH with a current source and amplifies its output. It provides three different gain settings to properly adjust the output voltage level and a first-order high-pass filter with three selectable cut-off frequencies. It can be calibrated for each ACH, providing at its output accurate values for the acceleration, the velocity, and the excursion of the object being measured.

A couple of years ago, mostly because I wanted to study the subject in more detail and expand my knowledge, I began looking for other possible solutions concerning the vibration measurements. In this article, I will describe this long-time research for the design and construction of such a system and I will present the results and the measurements I took.

I used the ACH-01 as my reference accelerometer because I didn't have any other accelerometer with better specifications. I have to point out that I didn't have any intention of making a reference system or replacing the ACH accelerometer, which has served me so well for many years now. The system that I will present should be considered as an educational approach to the vibration measurements for amateur builders. The interest here is mainly in the differences existing between the measurements rather than absolute measurements, because the system that I present in this article cannot offer calibrated outputs.

A Simplified View of the Piezoelectric Accelerometer

The piezoelectric accelerometer is widely accepted as the best available sensor for the measurement of vibration. It employs the piezoelectric effect of certain materials to measure dynamic changes in mechanical variables. According to this effect, when a force is applied on the piezoelectric material, a change in its dimensions occurs, which results in the generation of electrical charge.

A simplified version of a piezoelectric accelerometer is shown in **Figure 1**. The active element is the piezoelectric material, which has its bottom side bonded to a rigid base, while a so-called seismic mass is attached to its top side. When the system is vibrating, according to Newton's second law, the force acting on the piezoelectric element is equal to the product of the acceleration and the seismic mass. Due to the piezoelectric effect, a charge output proportional to the applied force is generated. Since the seismic mass is constant, the charge output signal is proportional to the acceleration and is connected to the sensor output via a pair of electrodes.

Sometimes, the mass of the piezoelectric material itself is used as seismic mass, reducing the sensitivity but increasing the resonance frequency of the accelerometer.

Brüel & Kjær has an excellent handbook about the piezoelectric accelerometer. The handbook can be downloaded from the company's website (see Resources) and is highly recommended.

Testing the System

The first thing I had to do was to find a reliable way to test and measure the vibration sensors. I needed a device to produce a defined vibration level, which should be constant through the entire range of interested frequencies. This is

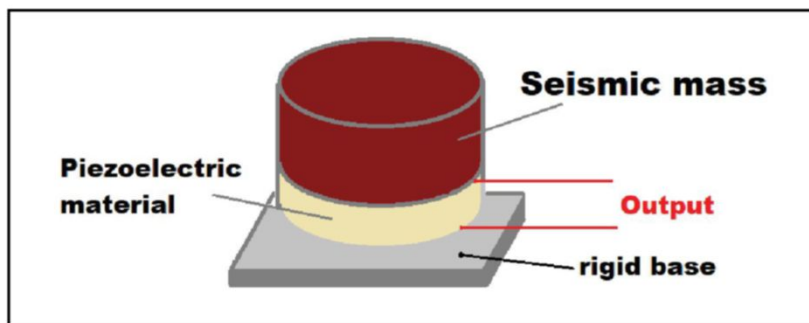


Figure 1: A simplified diagram of an accelerometer

not a problem if you have enough money to buy a mini-size Vibration Shaker or calibration device from any respectable manufacturer.

I decided to look for other solutions with an affordable cost and acceptable results. I had a Peerless driver available from an old subwoofer project, and I began the tests. When I started the project, I was interested in testing only at the low frequencies that could be covered by a subwoofer driver, but then I decided to expand the frequency range.

After a lot of experiments, I ended up with two separate testing systems—one for the frequency range between 5Hz and 300Hz, using a Peerless 830500 XLS 12" driver unit; and a second one for



Photo 1: The Peerless driver with the ACH placed on its cone

the frequency range between 200Hz and 2kHz, using a Visaton BS130 unit.

When a loudspeaker driver is used to test an accelerometer, one very difficult problem is finding an easy way to attach and remove the accelerometer from the cone without causing any damage. The Peerless driver's cone was made from a material to which I could attach the accelerometer and remove it without much difficulty or damage, using double-sided tape. But if it were done several times, after some point, the cone material would start to come off. I solved this by uniformly applying several layers of Loctite 401 instant adhesive to the center of the cone making it stiffer.

The Peerless driver is a very large and heavy unit, and I built a special contraption to support it in the vertical position as shown in **Photo 1**. The driver's frequency response was not very flat at the low frequencies since it was used without any enclosure. I equalized this response using a Linkwitz transform circuit. **Figure 2** shows the electronic diagram of the circuit using an OPA2604 op-amp. To calculate the values of the components, I measured the acceleration of the driver cone by

attaching the ACH sensor to its center. This was found to be similar to a second-order high-pass filter with $f_0=27\text{Hz}$ and $Q_0=0.43$. Then, I set the target response to $f_p=5\text{Hz}$ and $Q_p=0.7$. I used a spreadsheet that I found on the Linkwitz Lab's website (see Resources) to calculate the values of the components.

Photo 2 shows how I built the circuit. The power supply circuit is on the left PCB and the Linkwitz circuit is on the right PCB. The circuit applies a maximum boost of 25.8dB at 3.4Hz to achieve a flat response at the very low frequencies. This requires caution to keep the amplifier voltage that drives the subwoofer at low levels, otherwise the Peerless cone will make very large excursions that could damage it.

Additionally, I used a frequency response compensation file in ARTA software to make a few minor corrections to the response at the frequencies above 150Hz, resulting in an almost flat response as shown in **Figure 3**. The response was taken with the ACH accelerometer attached to the Peerless cone, and it is very flat from 7Hz to 300Hz, where the loudspeaker cone behaves like a rigid piston. The -3dB low-frequency point

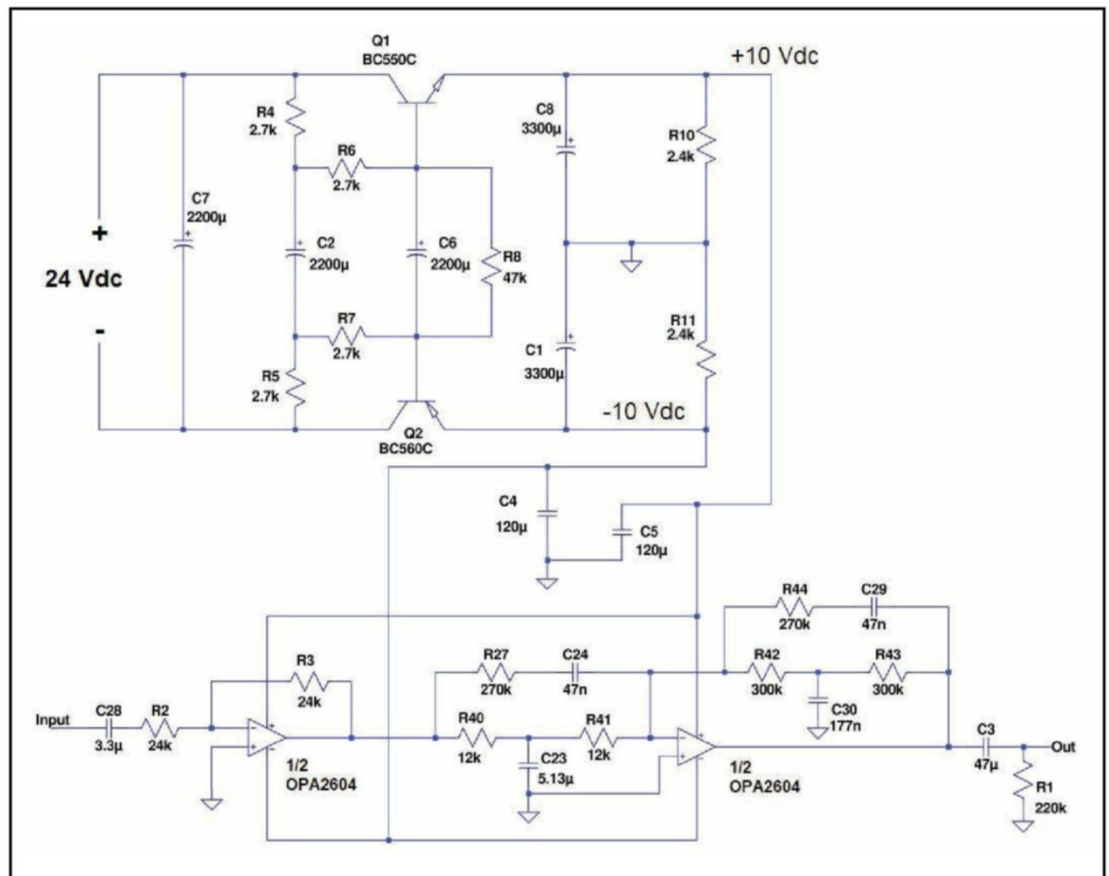


Figure 2: The Linkwitz transform circuit for the Peerless driver

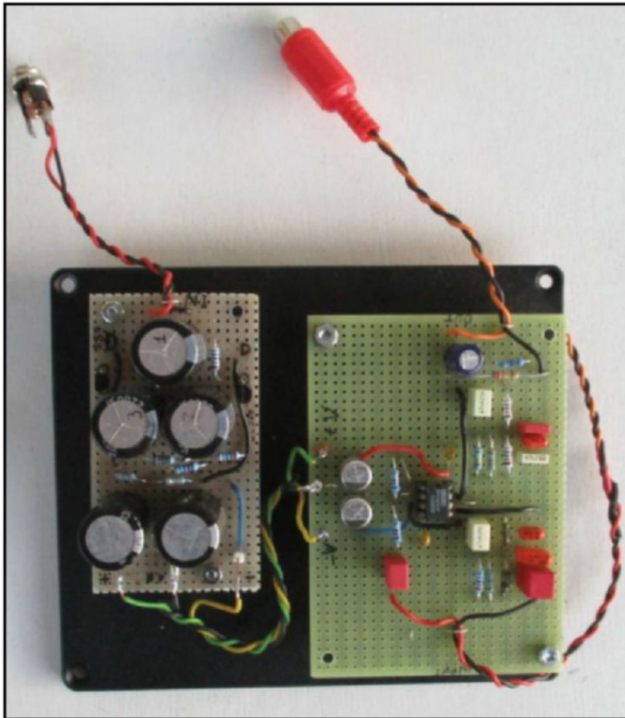


Photo 2: The equalization circuit of the Peerless driver

is at 5Hz. Beyond the frequency of 500Hz, the break-up resonance of the cone—which happens at 850Hz with a very large peak of almost 15dB—makes the system not useful for accurate measurements.

It is not easy to find a loudspeaker driver with flat response in the range between 200Hz up to a few kilohertz to test the sensors. The driver should have a small cone area, and this makes it very difficult to attach and remove the sensor from the cone. Another limitation is the weight of the accelerometer itself because it becomes comparative with the weight of the cone.

After a lot of thought and failed tests, I found a solution using the Visaton BS130 unit. Visaton describes the BS130 as a structure-borne sound converter for bass frequencies, but I will call it a “Bass Shaker.” The unit is firmly attached to a surface (e.g., a wooden panel), which could resonate, and when it oscillates, the vibrations are transmitted to this surface and they can be felt by the listener. It is used in home theaters, commercial movie theaters, arcade games, and so forth. It can be driven through a power amplifier like any other normal driver and its impedance is 4Ω.

I disassembled the Visaton BS130 (**Photo 3**), and I used only the internal vibrator. The oscillating plate is constructed from a very rigid material similar to the fiberglass that is used for the manufacturing the PCB. This makes the attachment and the removal of the sensor very easy.



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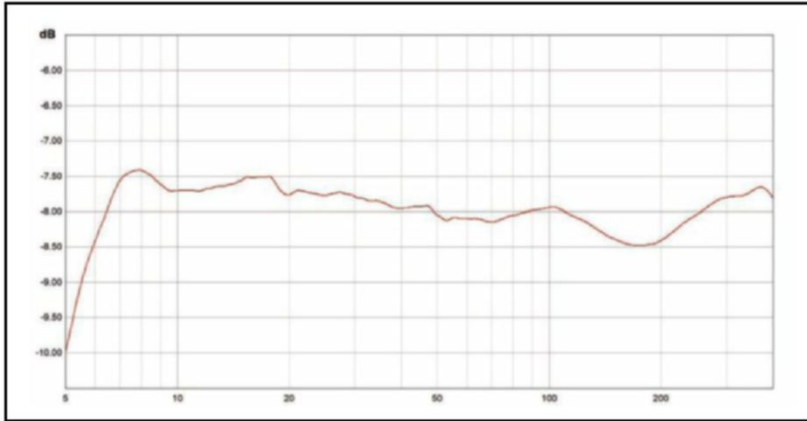


Figure 3: The Peerless cone acceleration after equalization, as measured with ACH



Photo 3: The disassembled Visaton BS-130 unit

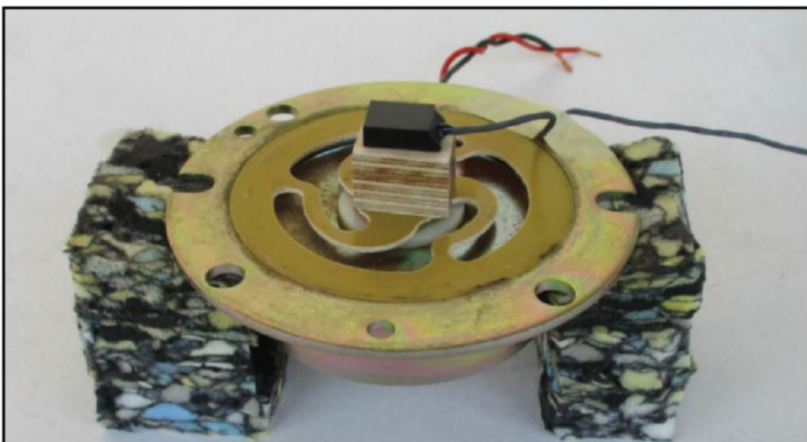


Photo 4: The Visaton BS-130 as modified for the measurements

I additionally placed a small piece of plywood with the dimensions 15mm × 15mm × 25mm to the center of the oscillating plate to attach the sensors using double-sided adhesive tape. The support base for the Bass Shaker is very critical and after some trials I found that the best one was made from a sound absorbing material. The final set-up is shown in **Photo 4**.

The acceleration response of the Bass Shaker with the ACH accelerometer attached (**Figure 4**) is reasonably flat within 0.5dB from 200Hz up to about 2.1kHz. Some minor resonances are shown at 450Hz and above 2.1kHz. Using these two systems, I had the ability to test the sensors with a reliable and repeatable way for the frequency range between 10Hz and 2kHz.

The Murata Shock Sensor

I began my experiments using the Murata shock sensor PKGS-00LDP1-R. This is a high sensitivity, SMD mounted, acceleration sensor with dimensions 6.4mm × 2.8mm × 1.20mm (**Photo 5**).

The sensor consists of a piezo ceramic element, which is stuck between two ceramic substrates. It is designed for hard-disk drive applications, used in personal computers and consumer audio-visual equipment. Its charge sensitivity is 0.84pC/g (±15%) where g is the earth's gravity and has a high nominal resonance frequency at 20kHz. The capacitance of the sensor is 770pF (±30%).

I used the circuit shown in **Figure 5** to amplify the sensor's signal. This circuit is based on the Jensen application note AS098 (see Resources) but I modified it to have unity gain.

I used a JFET in the input and a bipolar transistor buffers the output of the circuit. I built this circuit together with the sensor on a small PCB as shown in Figure 5b. The circuit was supplied by a 12V, very low noise power supply based on a capacitor multiplier.

I attached the PCB to the cone of the Peerless driver and I took an acceleration measurement. The result was very promising. The sensor had a good frequency response, almost similar to that of the ACH-01 sensor. But there was a serious problem—very, very large interference of the 50 Hz mains voltage and its harmonics was present in every measurement. Trying to find a solution, I shielded all around the PCB with an aluminum sheet and I connected it to the ground of the circuit. This gave much better results, but the interference was not totally eliminated.

I decided to continue looking for a better solution that would eliminate this mains interference. Also,

according to Murata's recommendation, I decided to design and build a charge amplifier circuit to use with the sensor.

The Charge Amplifier

A charge amplifier is a current integrator circuit that produces a voltage output proportional to the electrical charge flowing into its input. It consists of an operational amplifier (op-amp) in an inverting configuration with a feedback impedance that is a capacitor.

A practical circuit is shown in **Figure 6**. The piezoelectric sensor is directly connected to the inverting input. The charge Q produced by the sensor is proportional to the acceleration, and the circuit has an output voltage that is proportional to the acceleration measured by the sensor.

The resistor R_{FB} is needed to properly bias the op-amp.

The "charge gain" G of the amplifier is determined by the feedback capacitance C_{FB} and is given by the following formula:

$$G = \frac{1}{C_{FB}}$$

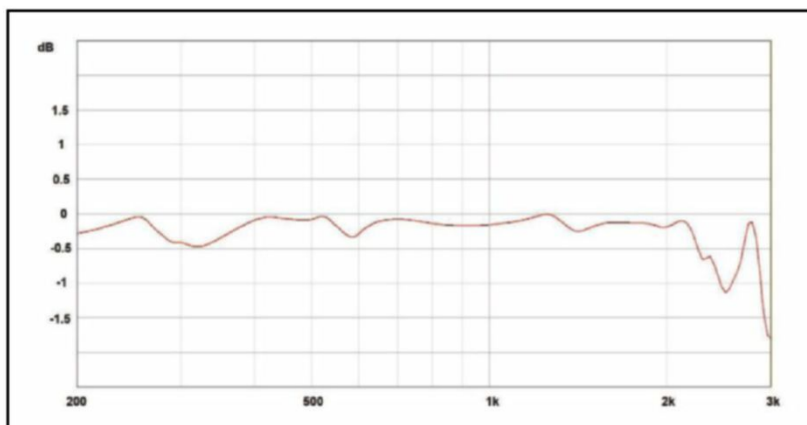


Figure 4: The Visaton Exciter cone acceleration was measured with the ACH accelerometer.



Photo 5: The Murata vibration sensor



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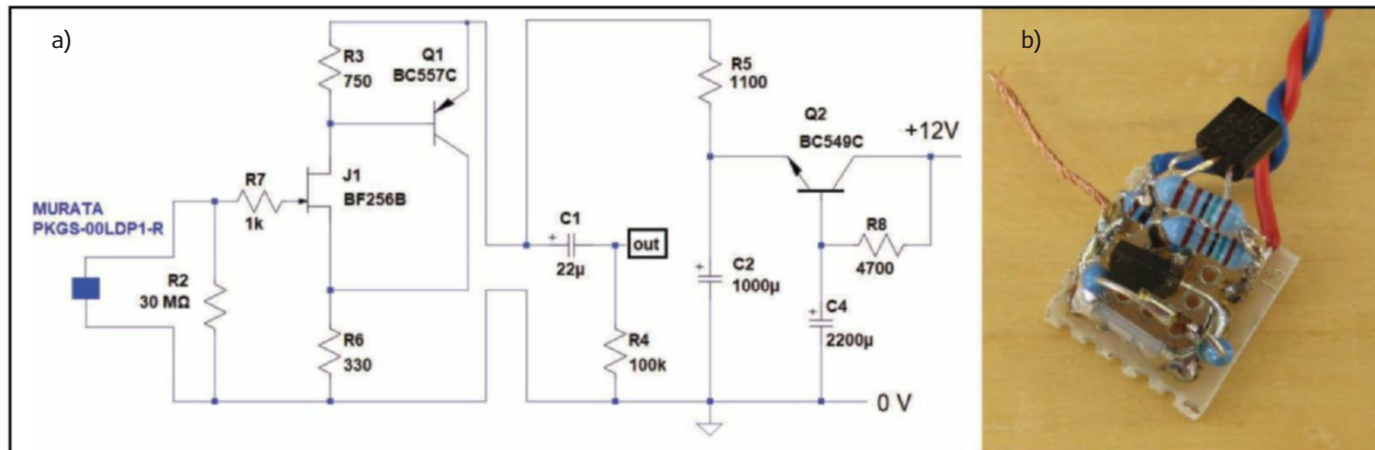
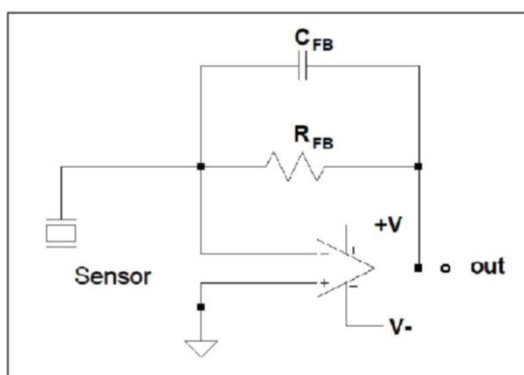


Figure 5: The schematic for the first piezoelectric sensor amplifier attempt (a) and the PCB (b)

Figure 6: A practical circuit for a charge amplifier



The formula indicates that the “charge gain” of the amplifier is independent of the cable capacitance or other input parasitic capacitance around the sensor.

The charge amplifier operates as a high-pass filter with the cut-off frequency F_{HP} given by the following formula:

$$F_{HP} = \frac{1}{(2 \times \pi \times R_{FB} \times C_{FB})}$$

Usually, the capacitance C_{FB} is a few hundred pFs and the resistor R_{FB} should have a very large value, in the order of a few tens of megohms, to achieve a low cut-off frequency. Very high value resistors are not very common, but this can be solved by using several resistors in series.

The input bias current of the op-amp flowing through this high value resistor will produce a large output offset voltage. This could be minimized by choosing an op-amp with very low input bias currents. A field-effect transistor (FET) input amplifier, which usually has an input bias current of a few pA (picoAmperes), will produce a very small offset of a few millivolts, as long as the resistor value is below a few hundred megohms. Then, this output offset can be filtered out very easily with AC coupling to the following stage.

Next Month

In the second part of the article, I will continue with the design and construction of the main piezoelectric sensor and I will present some other sensors that could be used for the vibration system. In the third and final part, I will present the design and construction of the piezoelectric charge amplifier and the measurements that will enable us to evaluate the whole system.

About the Author

George Ntanavaras graduated from the National Technical University, Athens, Greece, in 1986 with a degree in Electronic Engineering. He currently works as a Technical Consultant for Greek electronics companies. He is interested in the design of preamplifiers, active crossovers, power amplifiers, and most loudspeakers. He also enjoys listening to classical music.



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The Science and Physics of High-End Mic-Preamplifier Design (Part 3)

Considering the IC Options

By
Mike Meechan



In the previous two articles, Mike Meechan discussed the fundamental physics that play a part in microphone preamplifier design, and briefly explored its inhabitance in the practical world and how a good design can retain the fidelity of the mic signal along its path from transducer to output. At this stage, he considers constructing some of the theoretically-perfect building blocks of an active mic preamp.

In many respects, designing high-end pro audio equipment is much like the rapidly innovative engineering that goes into a Formula 1 race car. Both products rarely get a second chance in a live environment. So, a quote from the late and legendary engine designer Keith Duckworth sums up the unforgiving nature of both motorsports at its uppermost echelons and high-end pro audio design. "It was a different culture at Cosworth, a culture that demanded you don't do the calculations until you understand what the problem is." [1]

So let us look more at the practical considerations and what calculations and compromises these might entail. Changing tack from last month's (broadly) theoretical discourse, we now look at the more-practical real-life solutions. Recapping briefly, while costly, performance-wise, the JE-990 (as detailed last month) was well ahead of the game (manufactured first in 1979, and now in a more-recent reincarnation post-1989, integrated into various products of The John Hardy Co. [2]).

It is only in more recent years—20 years after the JE-990 debuted in 1980—that the very similar (John Hardy disputes the comparison) and competitively priced Graeme Cohen OM1556 design [3] made its commercial appearance. Key details included:

- Very low-noise balanced input/output encapsulated audio preamp hybrid
- DC coupled to eliminate C_{COUPLING}

- (Originally) dual super-matched transistor pair (LM394s) physically cross-connected in parallel
- I_E feeds output of each op-amp
- Input bias resistors be high and compatible
- Very low R_{GAIN} at 60dB gain maximizes noise performance
- 12-pin SIL construction
- SSM2015, 2016, and 2017 and THAT 1510/1512 are very similar

Table 1 shows various modern derivatives of the original Deane Jensen and Graeme Cohen designs. If we forego discrete transistor design, a simpler and often more economical/straightforward solution means consulting databooks before choosing from the various pro-audio ICs dedicated to low-noise amplification. However, convenience sometimes brings performance compromise—detractors have noted (somewhat unfairly, in my opinion) sonic deficiencies when fully-silicon solutions are implemented.

For several years, Analog Devices, after acquiring Precision Monolithics Inc.—which became Solid State Music (SSM) as a transitory company—produced a range of very highly regarded, versatile, pro-audio "solutions." Some of those solutions included low-noise integrated front-end solutions, most notably the SSM2015, the SSM2016, and the SSM2017 devices (which are all now obsolete and out of production). However, the more recently introduced

Device	THD+N @ 1kHz	Slew Rate	GBP	e_N	i_N	CMR	V _{SUPPLY}
AD797	0.00001%	20V/ μ s	110MHz (gain 60dB)	0.9nV/ $\sqrt{\text{Hz}}$	2pA/ $\sqrt{\text{Hz}}$	unspecified	$\pm 4.5\text{V}$ to 18V
INA103	0.0009%	15V/ μ s	100MHz	1nV/ $\sqrt{\text{Hz}}$	2pA/ $\sqrt{\text{Hz}}$ (@ 1kHz)	100dB	$\pm 4.5\text{V}$ to $\pm 25\text{V}$
MUSES03	0.0000%	35V/ μ s	12MHz	7.5nV/ $\sqrt{\text{Hz}}$	unspecified	90dB	$\pm 3.5\text{V}$ to $\pm 18\text{V}$
OPA1662/4	0.00006%	18V/ μ s	22MHz	3.3nV/ $\sqrt{\text{Hz}}$	1pA/ $\sqrt{\text{Hz}}$	106dB	$\pm 4.5\text{V}$ to 18V
OPA1656	0.00003%	24V/ μ s	53MHz	2.9nV/ $\sqrt{\text{Hz}}$	0.006pA/ $\sqrt{\text{Hz}}$	0.006%	$\pm 2.5\text{V}$ to $\pm 18\text{V}$
INA163*	0.0002%	15V/ μ s	80MHz (gain:40dB)	1nV/ $\sqrt{\text{Hz}}$	0.8pA/ $\sqrt{\text{Hz}}$ (@ 1kHz)	100dB	$\pm 4.5\text{V}$ to 18V
THAT1510 (mic pre IC)	0.005% (gain 60dB)	19V/ μ s	15MHz	1nV/ $\sqrt{\text{Hz}}$	2pA/ $\sqrt{\text{Hz}}$ (@ gain 60dB)	115dB (gain 60dB)	$\pm 5\text{V}$ to $\pm 22\text{V}$
Linear Technology LT1028	0.0018% (gain @ 60dB)	15V/ μ s	75MHz	0.85nV/ $\sqrt{\text{Hz}}$	1pA/ $\sqrt{\text{Hz}}$	120dB	$\pm 4.5\text{V}$ to $\pm 22\text{V}$
Elantec EL2125C	0.006%	185V/ μ s	175MHz	0.83nV/ $\sqrt{\text{Hz}}$	2.1pA/ $\sqrt{\text{Hz}}$	106dB	$\pm 2.5\text{V}$ to $\pm 15\text{V}$
SSM2275	0.001%	12V/ μ s		8MHz	7nV/ $\sqrt{\text{Hz}}$	<1pA/ $\sqrt{\text{Hz}}$	unspecified
Hardy 990C+ hybrid opamp**	0.0045%	18V/ μ s	50MHz	1.13nV/ $\sqrt{\text{Hz}}$	1pA/ $\sqrt{\text{Hz}}$	125dB (@ 60Hz), typically 100Hz	$\pm 12\text{V}$ to $\pm 24\text{V}$

* Instrumentation amplifier

** Figures are representative and depend on the source of the 990C and whether it is in its raw state (DIY JE-990 Discrete Operational Amplifier, www.hairballaudio.com)—the Audio Precision performance graphs are impressive for this variant—or built into a completed mature design (e.g., The John Hardy Co.'s M-1, M-2 and Jensen Twin Servo mic preamps from which the stated figures are quoted and the op-amp at $\pm 24\text{V}$ in order to maximize headroom). The servo circuitry runs at $\pm 15\text{V}$ for expediency. When the 990C is used in Hardy's various mic preamp cards (MPC-500C, MPC-600 and MPC-3000), the preamp must operate at the power supply voltages that are provided by those consoles, which is $\pm 18\text{V}$.

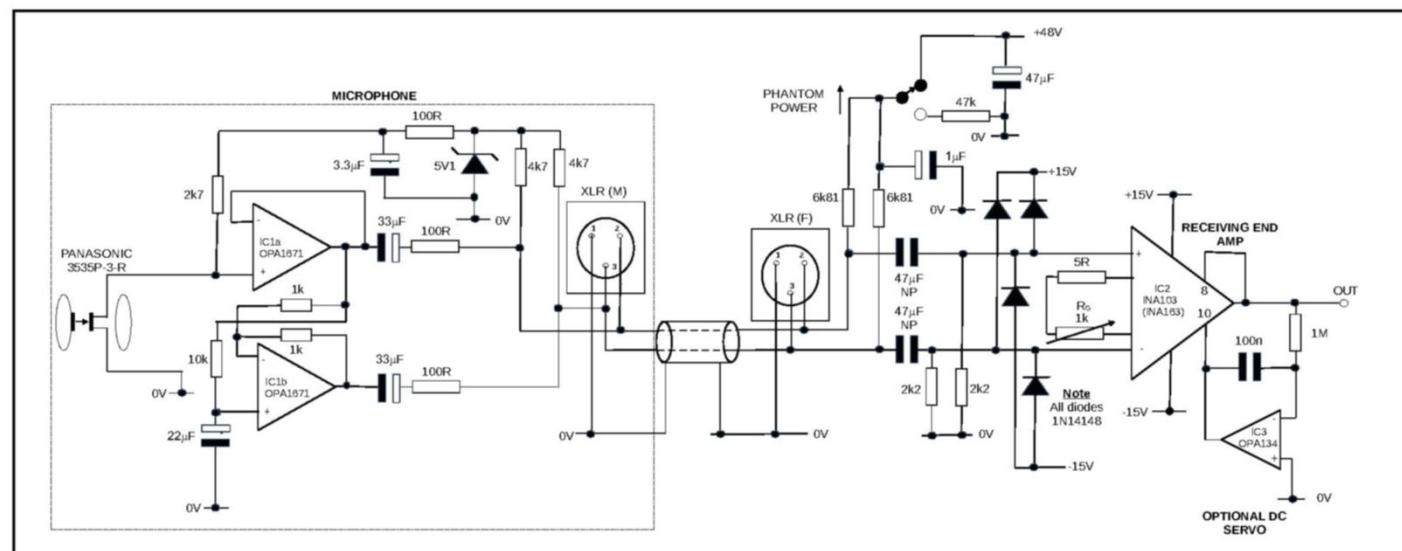
Table 1: Comparison of key audio performance parameters for various integrated mic preamp solutions

SSM2019 and the SSM2212 still find much favor in recent or revisited designs. Acclaimed designers often bemoan both the lack of development of discrete semiconductors, such as faster, higher-gain, higher-voltage transistors, and the need to find replacements for obsolete or obsolescent parts, which both sold well and exhibited exemplary performance in demanding pro audio fields, although pin-for-pin compatible ICs from various manufacturers (for certain products) do exist.

Burr-Brown's solution to the problem is the 16-pin dual-in-line INA103 "trans-impedance" instrumentation amplifier that can be used very successfully as a microphone preamplifier. This is

a low noise (1nV/ $\sqrt{\text{Hz}}$), low distortion (0.0009% at 1kHz with $G = 100$), high gain-bandwidth product (GBP) (100MHz with $G = 1000$) device having a DC common mode rejection ratio (CMRR) of 100dB and operation from supply rails up to $\pm 25\text{V}$ (meaning a very attractive single-ended output drive capability of +23.4dBu.) This is significant as many stand-alone analog-to-digital converters (ADC) are calibrated to need +24dBu to achieve 0dBFS. Furthermore, such topology allows access to the outputs from post- the main input stage, (via the "-Gain Drive" and "+Gain Drive" pins) providing a differential as well as a more conventional single-ended output. Interesting...

Figure 1: Typical (simplified) condenser internal mic capsule preamp (Source: Texas Instruments)



We can swing something like 22.86V_{RMS} or a rather healthy 64.66V_{pp} via THAT Corp.'s robust low-noise low-distortion bomb-proof differential high line-level driver (or a high-quality output transformer such as those of Jensen).

The corporation also offers the THAT 1510/1512

complimentary chipset integrated solution(s) [4]: paper specifications are impressive (large GBP, fast slew rate, very low distortion, low noise, and large output swing). Both are also pin-compatible with the obsolete SSM2017—and the SSM2019 (and other ICs). Another approach to the problem would directly

Preamp Type	EIN		Noise at 1kHz	Distortion at 1kHz		CMR (100Hz) *		CMR (1kHz)		CMR (10kHz)	
	RMS	Peak		Maximum Gain	Minimum Gain	Maximum Gain	Minimum Gain	Maximum Gain	Minimum Gain	Maximum Gain	Minimum Gain
THAT 1510	-128.3dBu	-124dB4	*(see note below)	0.0005%	0.005%	120dB	60dB	≈82dB	≈67dB	≈78dB	≈70dB
SSM2212	-128.3dBu	-124.5dB4	-87.4dBu	0.008%	0.015%	≈91dB	≈70dB	≈80dB	≈63dB	≈79dB	≈72dB
MPSA55 discrete*	-128.3dBu	-124dB4	-85.5dBu	0.04%	0.04%	79dB	55dB	83dB	70dB	79dB	55dB
Low-noise op-amp	-128.3dBu	-114dB4	-84dBu	0.04%	0.08%	72dB	70dB	72dB	53dB	71dB	45dB

* 1 nV/√Hz input noise (60dB gain), 3.4 nV/√Hz input noise (0dB gain), 2pA/√Hz at 60dB gain

** BC214C, ZTX214C, BC461, and 2N4401 low-noise transistor types returned similar measurement figures

Measurements taken in some instances using an Anritsu Network/Spectrum Analyzer MS420B, 10Hz to 30MHz, (or an HP3651A), a Boonton Audio Analyzer 1120, 10Hz to 140kHz, And some years back, a BBC ME-25/EP-14 tone source and noise meter, and a DIY ultra-low-noise/distortion temperature-stabilized Wien bridge signal generator, modified from an original John Linsley Hood design "Wien-bridge Oscillator with Low Harmonic Distortion: new way of using Wien network to give 0.001 % THD," which was published in Wireless World, May 1981, pp 51-53.

Table 2: Comparison of various key audio parameters of different input stage types

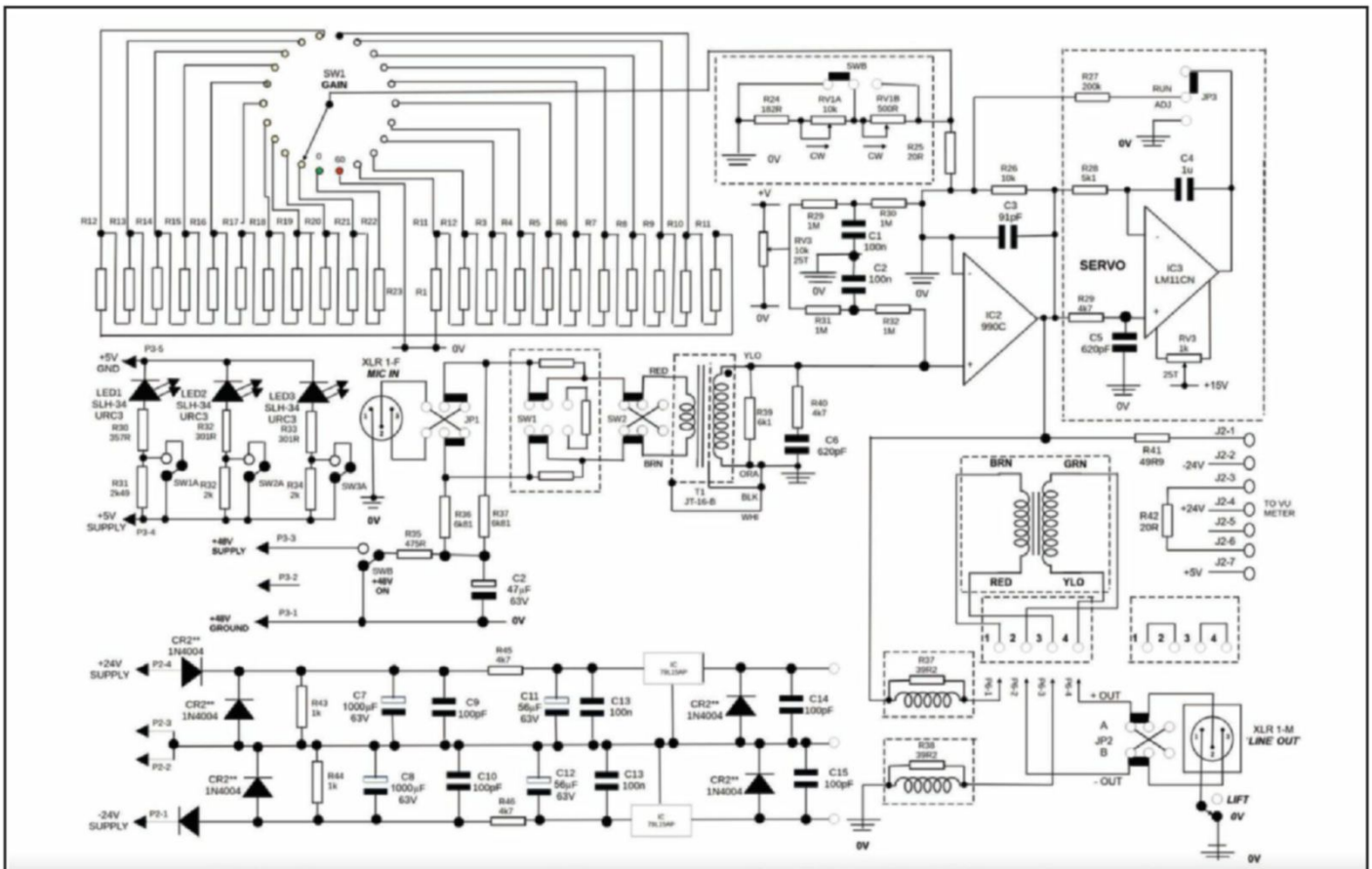


Figure 2: Hardy mic preamp modified to include a precision 24-way ELMA multi-way rotary switch giving precise 2.5dB incremental gain changes (derived from 990.pdf, johnhardyco.com, and hopefully with the various resistor values successfully reverse-engineered)

use op-amps in the input signal chain, which seems superfluous to fly in the face of what has been said up until now. However, the AD797 as chosen for the design is no ordinary op-amp device—as shown in Table 1. Other fine choices are Texas Instrument’s (TI) OPA1662 and the INA163. The THAT1510 scores highly too. **Figure 1** shows a typical condenser mic capsule preamplifier design.

General Rules About Noise (and Distortion)

The overall goal is to create athletic, accurate, transparent, linear-responding circuits. In other words, a neutral signal path—for acoustic and classical music recording. Here are some very basic rules:

- There can be Early effect distortion caused by the nonlinear collector base capacitance. This causes variations in the effective width of the transistor base, creating noticeable distortion at high voltage swings and high frequency.
- Coupling capacitors around op-amp output(s) can cause transient overload and distortion (and IMD is worse subjectively than THD).
- In cross-coupled topologies, low values of all R_{SERIES} reduce noise.
- Noise overall and crucially is dominated by diffusion noise in the input stage (whose noise will swamp all other stages) so the initial design parameters are crucial. Poisson or thermal noise and $1/f$ noise dominate the overall noise figure. Other types of noise fall away to such small figures as to be insignificant.
- Shot noise (Schottky or Poisson noise) is caused by random fluctuations in current—it results from the fact that the current is not a continuous flow but the sum of discrete pulses in time.
- Flicker noise dominates at low frequencies because of impurities in the conducting channels, and generation and regeneration due to I_{base} , etc. It is sometimes known as pink noise—equal noise power per octave—or $1/f$ noise. External static, hum, and mic stand, mixing desk or subway rumble, then self-generated noise, and air conditioning sound, all add to this low-frequency noise. Above the $1/f$ corner frequency—which varies typically from 2Hz to 3Hz to 100Hz—noise can be assumed to be of constant amplitude across the spectrum (white noise).
- Popcorn noise has all but been eliminated because raw material purity of semiconductors is

Step switch position	Calculated value	EIA E192 preferred value	E24 series/ parallel configuration	Value	Variance (%)
1	1k25	1k24	1k1+150R	1k25	0
2	938R	942R	1k2//4k3	938R2	0.019
3	704R	706R	680R+24R	704R	0
4	528R	530R	510R+18R	528R	0
5	396R	397R	330R+39R	369R	0
6	297R	298R	270R+27R	297R	0
7	223R	223R	220R+3R	223R3	0
8	167R	167R	120R+47R	167R	0
9	125R	124R	110R+15R	125R	0
10	93R8	94R2	120R//430R	93R82	0.019
11	70R4	70R6	68R+2R4	70R4	0
12	52R8	53R	51R+1R8	52R8	0
13	39R6	39R7	36R+3R6	39R6	0
14	29R7	29R8	2R7	29R7	0
15	22R3	22R3	18R+4R3	22R3	0
16	16R7	16R7	12+4R7	16R7	0
17	12R5	12R4	11R+1R5	12R5	0
18	9R38	9R42	12R//43R	9R38	0.019
19	7R04	7R06	7R5//110R	7R02	-0.266
20	5R28	5R3	5R6//91R	5R275	-0.088
21	3R96	3R97	6R2//11R	3R96	0.129
22	2R97	2R98	3R//300R	2R97	0.01
23	2R23	2R23	27R//13R	2R24	0.254

Table 3: Values of resistors to effect a precise 2.5dB gain change in a modified Hardy preamp

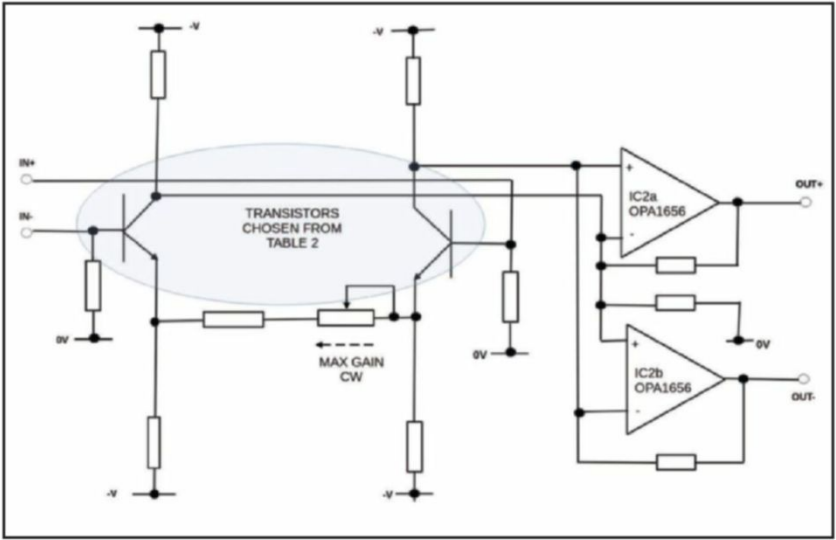


Figure 3: The cross-coupled input stage creates differential output (discrete vs. partially integrated).

high and modern-day production tests advanced sufficiently for them to be considered reliable.

- Distributed (parallel-connected) amplifiers—whether op-amp or transistor—minimize e_n because the outputs of all are summed (gain across each device provide, times number of devices, equals overall gain. Thus, the signal-to-noise improves according to the square root of the number of paralleled devices). The signal is summed in phase whilst those of noise are added in quadrature improving signal-to-noise ratio (SNR) because the signals are uncorrelated and can null each other. **Table 2** compares various key audio parameters of different input stage types. **Table 3** provides the values of resistors to effect precise 2.5dB gain change in a modified Hardy

preamplifier shown in **Figure 2** and **Figure 3** shows the cross-coupled input stage, which creates the differential (discrete vs. partially integrated).

Dividing the input voltage noise by the input current noise suggests that the Optimum Source Impedance (OSI) of the AD797 is 450Ω , (akin to the of low-noise bipolar transistors), implying that it can be used for direct connection to the microphone signal source. The only downside is it suffers from the same malaise as all other low-noise active designs—common mode signals typically in excess of the supply rails ($\pm 18V$) will blast the input. (Protection can only be afforded by having input leg series resistors. These worsen noise and CMR performance unless precautions are taken with the attenuator network topology).

Barring its dislike of input levels $>0dBu$, in all other respects, its performance is nothing short of outstanding—quite remarkable really, given that it is now decades old. The LT1128 is another possible contender in this marketplace (slightly better input noise, higher supply rails, and maximum permissible input voltage of $\pm 22V$, but compared to the AD797, [slightly] inferior distortion performance). Finally, the EL2125C provides a $0.83nV/\sqrt{Hz}$ and $2.1Pa/\sqrt{Hz}$ combination and is eminently suitable—as long as its minimum gain of 20dB can be tolerated.

Next month, we consider the digital revolution, wherein remote-controlled preamps are placed as close as possible to the microphone in a live sound environment, and use conventional relays or dedicated and specific ICs (which use FETs) to effect gain change.

About the Author

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



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The Best of Two Worlds

An Old-Fashioned Tube Amp and Multilevel Class-D Hybrid DIY Bass Guitar Amplifier Project

By
Jens Tybo Jensen

(Infineon Technologies)

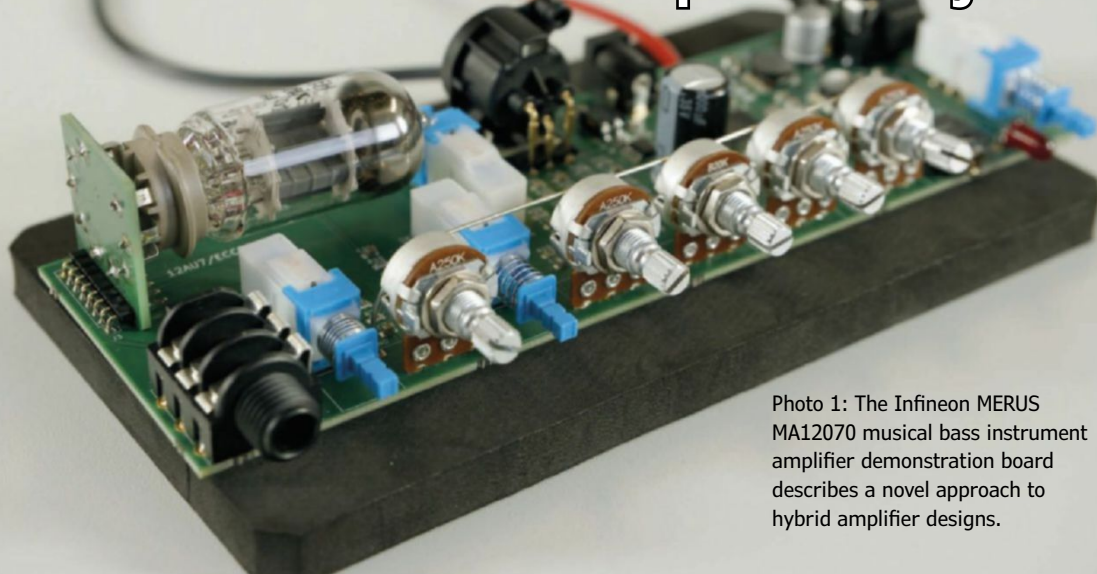


Photo 1: The Infineon MERUS MA12070 musical bass instrument amplifier demonstration board describes a novel approach to hybrid amplifier designs.

This article describes the practical and basic electrical design of a DC (AC adapter) or battery-powered, 60 W, professionally featured and ultra-efficient bass instrument amplifier. It is modelled after a classic vacuum-tube bass amplifier topology, while utilizing the exceptional audio quality and best in class efficiency of the Infineon MERUS™ amplifier technology to amplify every nuance of a genuine vacuum-tube preamplifier.

Since the advent of the modern Class-D amplifier, the goal of creating the “perfect” audio amplifier (i.e., an ideal electrical two-port network that produces a signal at the output port that is an exact replica of the signal applied to the input port but increased in magnitude) has moved increasingly closer.

As such, audio amplifiers have become a lot more affordable as well as smaller and lightweight, and more energy efficient and more reliable. Also, compared to earlier amplification classes, they exhibit much better figures of merit (e.g., lower harmonic distortion, lower noise, flatter frequency response, wider bandwidth, and excellent transient capabilities). In a nutshell, Class-D amplifiers have become the electronic equivalent of the “perfect magnifying glasses,” which enhance the original “image” with perfect sharpness and without any distortion or change of color or tone to the original.

Hey! What’s not to like? Perfect is great, right? Except that for some applications (e.g., amplifiers for musical instruments), perfect electrical amplification is not always enough.

In many of these applications, the audio amplifier plays an active and often very important role as an integral part of the instrument. Musicians often want the amplifier to act as a sound shaping tool. Rather than perfect amplification, a musician wants “color” and just the right amount of distortion and “growl” to be added to the original sound of the instrument.

When amplifying electric guitars and bass guitars, vacuum tube amplifiers (tube amps) are often regarded as superior in sound to solid-state amplifiers. The practical drawbacks from using vacuum tubes are obvious though: high power consumption, high voltages, questionable reliability, poor mechanical stability, big size, heavy weight, and high cost.

Digital emulation (DSP) of tube amplifiers has been designed to overcome many of these drawbacks, but while being very reliable, flexible, and comparable in sound, digital emulations seem to miss some characteristics and feel of a real tube amp.

Hybrid amplifiers combining vacuum tube and solid-state circuitry have been made. These amplifiers often cater to the low- to mid-price

segment of amplifiers, but the use of a tube is often more just for looks and sometimes not even electrically connected to the input signal circuit.

Designing an amplifier for musical instruments is quite a different task than designing a hi-fi amplifier. The general consensus for what constitutes a “good” hi-fi amplifier is that the amplifier should be made not to color the sound in any way: having very little total harmonic distortion, very flat frequency response, very wide bandwidth, and excellent transient capability.

The musician on the other hand wants the amplifier to exhibit the diametrically opposite qualities—to shape and create sounds rather than to reproduce sounds.

To exemplify, consider what makes the sound different between a piano and a guitar playing the same note, such as the A-flat, which has a fundamental frequency of 440Hz. A piano playing an A-flat note sounds very different to a guitar playing the same A-flat note. The harmonic spectrum of a piano is very different to the harmonic spectrum of a guitar: It distorts the fundamental 440Hz tone in a different way, thus creating a piano sound rather than a guitar sound.

For this reason, musicians often use tube-amplifiers for amplifying their instruments. Tube-amps sound fuller and more complex with more pleasant overtones. Especially when overdriven. This is due to the inherent high distortion/characteristics and somewhat slower dynamic behavior of the vacuum tube. Especially the triode vacuum tube where even-harmonic distortion components dominate the spectrum. The human ear generally likes even harmonic distortion components over uneven harmonics to the fundamental tone.

Instead of objective characteristics, such as total harmonic distortion (THD) and noise, you might hear musicians use different words to depict the ideal amplifier. You may hear them describe the amplifier as “more musical” or more “forgiving.”

I even saw a bass amplifier praised by a musician who wrote that his musical notes “sank into the power amp tubes like on a luxury pillow top mattress, with clean lower-mids and punchy but not too harsh upper-mids.” Not a very scientific description, but he clearly knows what he wants.

For these reasons, musicians often prefer old-fashioned cathode Class A tube-amplifiers. They frown upon Class-D amplifiers for being too harsh or dull

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sounding whereas tube-amps are described as having fuller and more complex sound with more pleasant overtones—especially when overdriven with a guitar.

So, what is going on here? Is this just a romantic preference for the technology from yesteryear, a modern form of “snake oil” or is there actually

something more to it? As it turns out, there are perfectly sound scientific explanations to the why musicians tend to prefer Class A “tube sound.”

As an example, the inherent distortion characteristics and the somewhat slower dynamic behavior of the vacuum tubes, especially with triode vacuum tubes, means that distortion of the even-harmonics components dominate the spectrum over uneven harmonics. The human ear generally prefers even-harmonic distortion components over uneven harmonics to the fundamental tone, which explains why the “tube sound” is preferred for the right guitar amp groove. It is obvious, therefore, that for some applications the older amplification techniques have distinct advantages.

Electrical Specifications	
Power Input	+24VDC, 3 A regulated, positive-tip 2.1mm DC-barrel
Total Idle Power Consumption (dominated by the tube)	2.7W @ 24V input
Power Input at 10WRMS Output	15.3W @ 24V input
Output Power	60 WRMS @ 5% THD 4Ω
Output Load Impedance	1× 4Ω to 16Ω, 1/4" jack
Instrument Input Sensitivity/ Impedance	25mV/1MΩ, 90mV/65kΩm 1/4" mono
AUX Input Sensitivity/Impedance	400mVrms/200mVrms, Mono/stereo, 4.7kΩ, 3.5mm (mini) Jack
Bass Control Range	±12dB @ 40Hz
Treble Control Range	±14dB @ 4kHz
Middle Control Range	11dB @ 380Hz
D.I. Output Level	True balanced: -6dBm ground lift
D.I. Output Frequency Response (-3dB)	55Hz to 8kHz, 6dB/octave

Table 1: Here are the amplifier’s electrical specifications.

Main Features of the MERUS™ MA12070 Amplifier

- Single classic tube sound bass instrument channel
- 1× 60W @ 4Ω speaker output
- Powered by off-the-shelf regulated Wall Adaptor (+24VDC, 3 A) or battery pack
- Unparalleled power efficiency—with very low power losses and cool operation.
- No external heatsink required (full volume, bass instrument playing)
- Dual 12AU7 triode Class-A pre-amp run directly on adaptor voltage for genuine “tube sound”
- Passive instrument input jack with -12dB input pad switch for active instruments
- Gain and volume controls for tube overdrive
- Bright switch for slap-bass sound
- Dynamic “breathe and feel” when fully dimed
- Classic three-band passive Baxandall tone control: treble, middle, bass
- Stereo 3.5mm AUX input for practicing/street accompaniment
- XLR D.I.: Frequency compensated, true balanced output for live PA/ mixing desk (phantom power safe)
- XLR D.I. ground switch
- Protection: Power supply reverse polarity and hot-plug, thermal, DC and speaker short-circuit
- Speaker jack output with low-cost LC-filter
- DC barrel input jack
- Low-idle power consumption even with vacuum tube (2.7W)
- Small dimensions: 185mm × 60 mm × 30 mm

The MERUS MA12070 Musical Bass Instrument Amplifier

Does that mean you have to choose sides? No. In fact as this article aims to describe, it is possible to get the best of both worlds (**Photo 1**).

The MERUS MA12070 amplifier utilizes the exceptional audio quality and superior power efficiency of Infineon’s MERUS Class D amplifier technology to amplify every nuance of a real 12AU7 triode vacuum tube preamplifier.

The 12AU7 triode tube is run at low plate voltage as a preamp stage, which excludes high voltage generation circuitry and makes the design a lot more accessible while leaving the heavy lifting of the final signal amplification to the much more power efficient integrated Class-D amplifier IC.

Compared to an original vintage tube amp, the power consumption, cost, size, and weight is dramatically reduced. Correspondingly, reliability and safety are also dramatically enhanced.

This amplifier platform can be scaled for different variants of power and features. As suggested here, it can be powered by a regulated +24VDC off-the-shelf regulated external power supply (wall adaptor). This could also be a mains supply or a battery-pack as well. The supply voltage and loudspeaker impedance determine the output power, and the circuit topology is very flexible for enabling or disabling features. **Table 1** shows the amplifier’s electrical specifications.

It can be used as a standalone unit or in combination with a loudspeaker in a combo format. This eases the requirements for the output filtering and due to the multilevel output, which inherently has less out of band noise—cheap ferrite beads and twisted speaker wires is often enough to pass EMC requirements.

The bare-minimum BOM cost, excluding switches, potentiometers, and jacks, is comparably low, which enables the design to be used in professionally-featured DIY products at a very competitive cost.

The Amplifier Topology

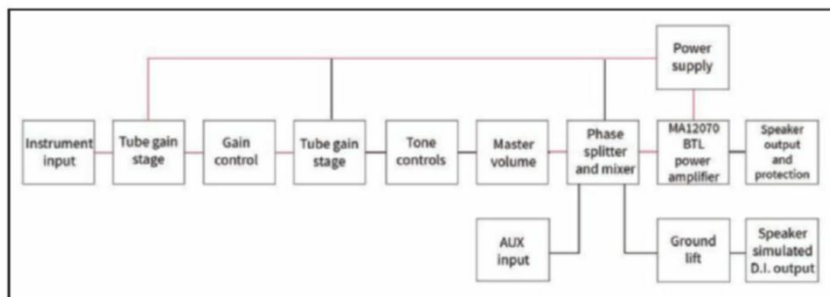
There are many tube amp designs and topologies from which to choose—and some designs have proven more successful than others. An American amplifier company from New Jersey relied on a certain topology though, and this is still regarded as one of the best and most classic tube bass amplifiers.

The MERUS MA12070 musical instrument bass amplifier project was inspired by the topology shown in **Figure 1**. The complete circuit for the amplifier is shown in **Figure 2**.

The Tube Preamplifier

The heart of the MERUS MA12070 musical instrument bass amplifier's tone-generating stage is the 12AU7 tube preamplifier. It relies on the excellent low plate voltage operating characteristics of this type of tube, which is available from several manufacturers. Passive filters and tone controls further shapes the frequency response of the preamp.

The 12AU7 triode has two identical triode sections build in. Both triode halves are used for AC



signal amplification. They are bot set up for low plate voltage operation. The two gain stages have been cascaded with a passive gain control placed between them and the signal amplitude from the first gain stage is fed into the second, thus overdriving the pre-amplifier for a complex distortion sound.

Figure 1: The MERUS MA12070 project was inspired by classical American tube bass amplifier construction.

Distortion and Dynamics in the Tube Preamplifier

There are several distortion generating elements in the 12AU7 preamplifier, including:

- Asymmetric gain of positive and negative halves of the grid input signal, due to the nonlinear nature of the triode characteristics.

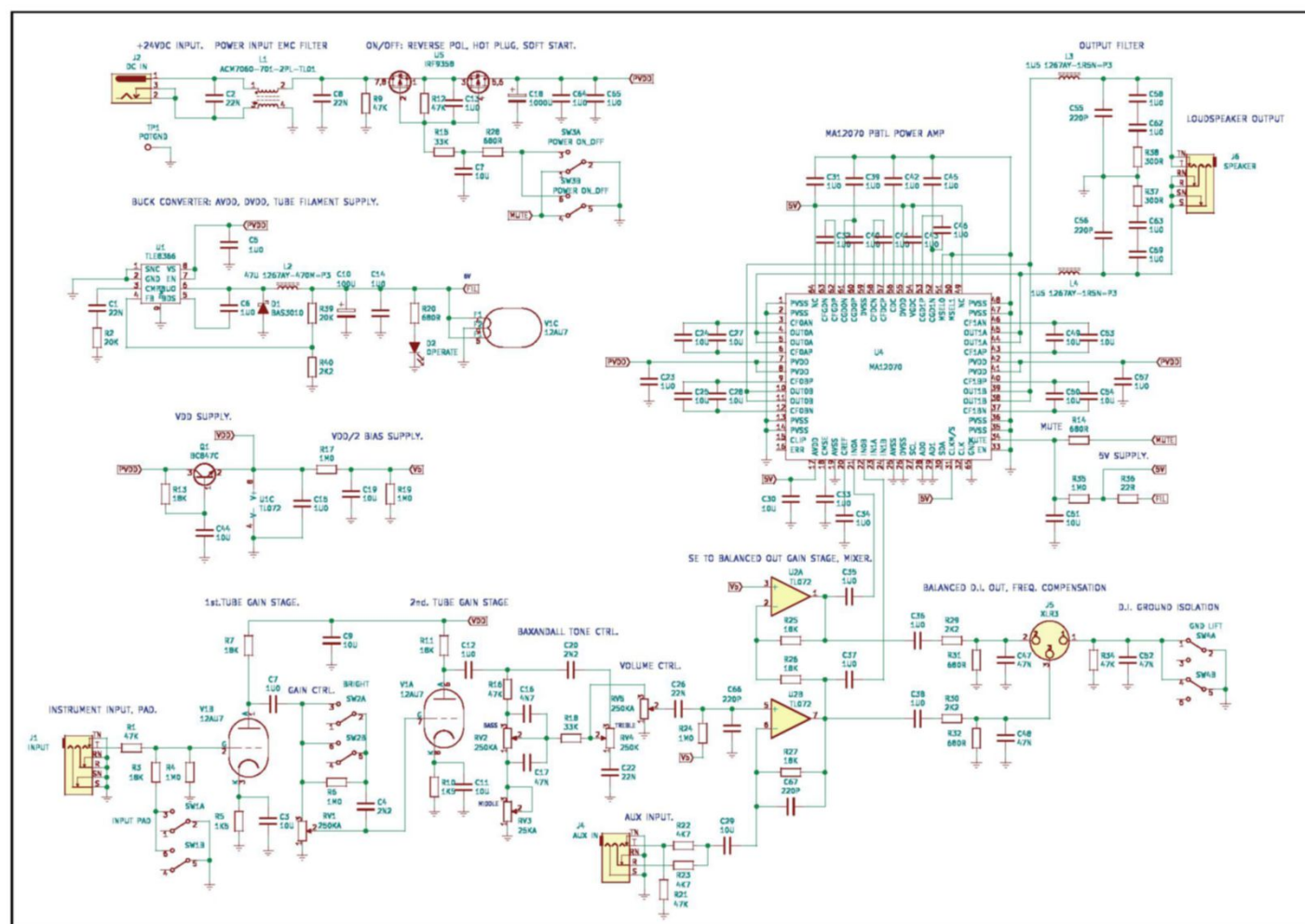
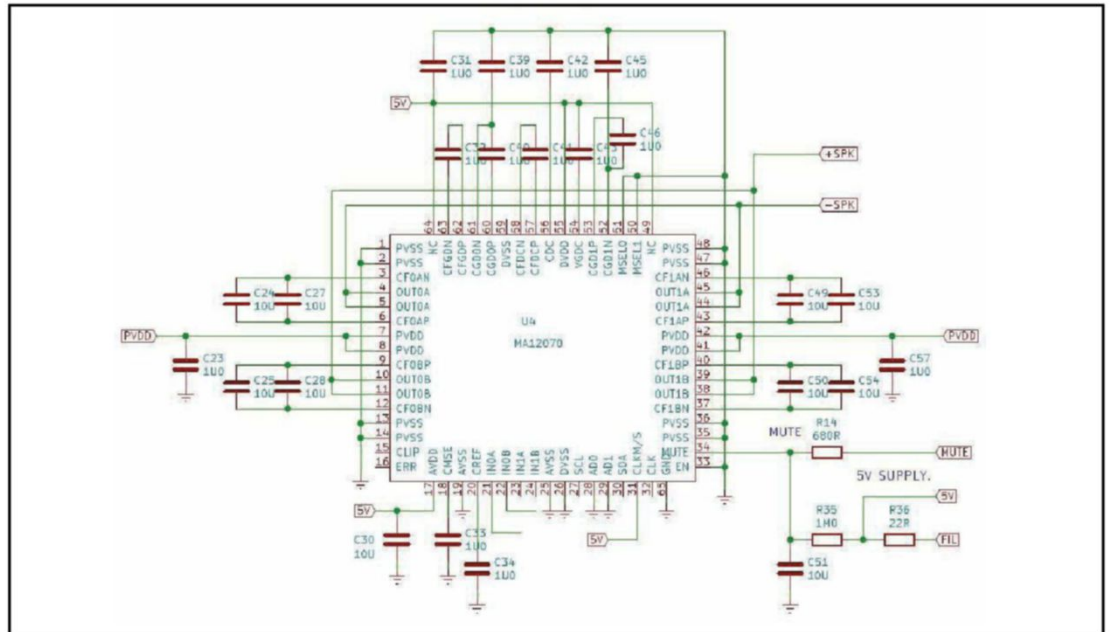


Figure 2: The musical instrument bass amplifier schematic with the MERUS MA12070 power amplifier

Figure 3: The MERUS MA12070 PBTL power amplifier from Infineon



Subtle even harmonic distortion components that rises with input amplitude.

- Grid-distortion, by driving the input grid of the triode positive with respect to its cathode. The grid starts to attract electrons from the cathode and input impedance falls rapidly. More abrupt and creates clipping of the lower part of the output waveform with strong even harmonic distortion components.
- Anode clipping by driving the grid so negative, that plate runs out of supply-voltage. The upper half of the output waveform is clipped with strong even harmonic distortion components.

The second gain stage alters the first gain stages AC-conditions (amplification) by loading of the falling input impedance of the second grid when overdriven. Thus, creating more asymmetric clipping and compression.

Combining positive grid and anode clipping distortion yields more un-even distortion components as the input amplitude rises (e.g., plucking a string on a bass guitar harder). That way the musician can vary the distortion content by different playing style, creating a very dynamic sound with varying harmonic content.

Other effects of clipping/dynamic compression in a cascaded configuration is blocking distortion and shifting bias point distortion all contributing to the dynamic feel of the amplifier.

Infineon's MA12070 Power Amplifier

The bass amplifier power stage is built around the Infineon MA12070 multilevel digital audio power amplifier IC with analog audio input. The output configuration is parallel bridge-tied load (PBTL) for maximum efficiency and low-device power dissipation (**Figure 3**).

With a supply voltage of +24V the MERUS MA12070 amplifier delivers 60WRMS into a 4Ω loudspeaker load and due to the high efficiency of the amplifier this is done without any external heatsinks.

Follow-Up Information

If you are interested in building your own bass guitar amplifier, the full documentation including an application note with further description of the circuit, BOM, schematics, PCB layouts, is available from Infineon in addition to a ready-made reference board called DEMO_BASSAMP_60W.

For more information about Infineon Class-D audio amplifier solutions, visit the Infineon website at: www.infineon.com/MERUS!

About the Author

After the Infineon acquisition of Class D Amplifier IC startup Merus Audio, where **Jens Tybo Jensen** was serving as VP Sales and Marketing, he became head of Marketing and Application Engineering for all Class D Audio Products within Infineon Technologies. Prior to his tenure with Merus Audio, he held a European Sales Director position with Knowles Electronics—a global supplier of high-performance micro-acoustic components and intelligent audio IC solutions. His global business development experience stems from a variety of start-ups and international hi-tech companies. He was co-founder and VP Sales at Nangate, Inc., which was acquired by Silvaco, Inc. He also served as VP Sales & Marketing for Exbit Technology A/S, which is now a part of Microsemi Corp. He holds a B.Sc.EE degree from the Technical University of Denmark and an Executive MBA from Copenhagen Business School.





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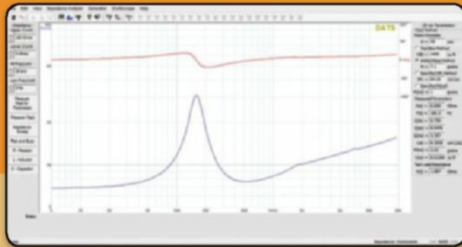
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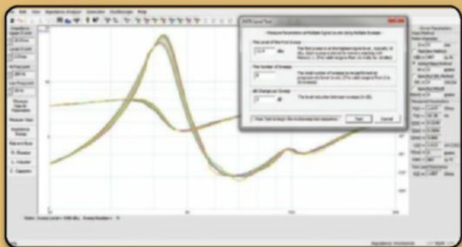
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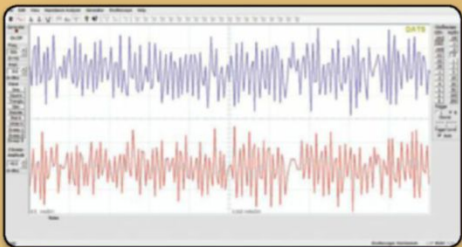
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- NEW DATS Linearity Test – easily compare multiple drive levels
- Rugged extruded aluminum housing
- Built-in precision calibration resistor
- Detachable test leads and USB cable
- On or under-desk mounting brackets

DATS V3
DAYTON AUDIO TEST SYSTEM

DAYTONAUDIO



RAVE REVIEWS!

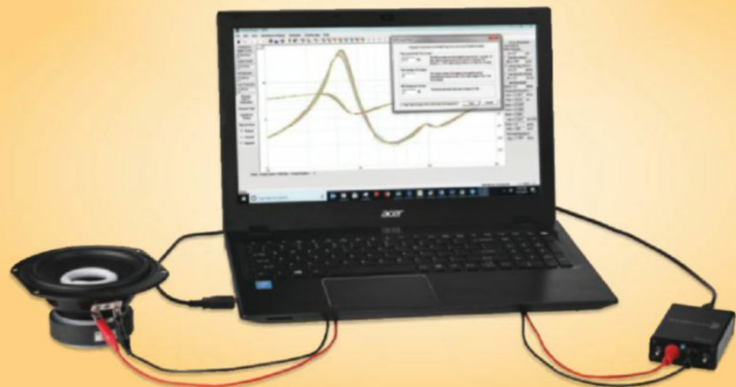
“Makes designing new speakers, and analyzing existing speakers, extremely easy and accurate.”

–Mark B.

Dayton Audio has taken the most complete and easy-to-use audio test system and improved it in almost every way when creating the DATS V3 Computer Based Audio Component Test System.

DATS (Dayton Audio Test System) V3 is the essential tool in any speaker designer's or audio engineer's toolbox, providing accurate, precise measurement data for any audio transducer or loudspeaker system. Save yourself time, money, and trial-and-error frustration by using DATS V3 to measure loudspeaker drivers, crossover networks, and complete audio systems!

DATS V3 is a unique, multipurpose test system that works with your PC and full-featured test software to measure the impedance response of a speaker driver and derive a complete set of Thiele/Small parameters that describe its operation. It's as easy as connecting a driver and starting the test!



RAVE REVIEWS

“This was simple to get up and running.”
–TigerTrading

“This is truly a tool you must have if you want to build subwoofer enclosures.”
–Dante B.



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