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ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Market Update

**From Home to the World
Microphones Get Essential Status
+ Voice Interface Systems**

By J. Martins

R&D Stories

**Why Thousands of
Virtual Microphones Matter:
The Impact of Acoustic Spatial Resolution
in a 3D Space**

By Erica Yorga (Nureva)

Audio Electronics

**The Science and Physics
of High-End Mic-Preamplifier Design**

By Mike Meehan

Fresh From the Bench

Widex Moment mRIC RD Hearing Aid

By Brent Butterworth

Audio Scope

**Design Considerations
Imbedding a
High-Output Buffer
Inside an Op-Amp Loop
By Michael Steffes**

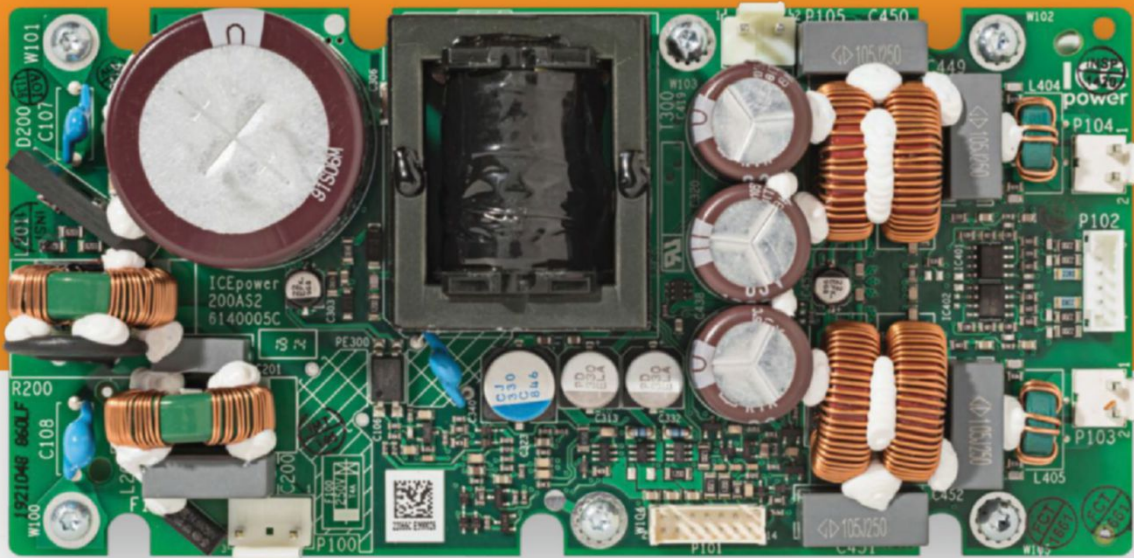
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Moving Coil/Moving Magnet
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(Part 1)
By Andrew C. Russell**

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Years**



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An Audio Renaissance

As the saying goes, “Without Input There Is No Output,” and without microphones there wouldn’t be audio recordings. Microphones were invented and perfected to serve basic communications and later for broadcasting—essentially connecting people. When Alexander Graham Bell patented the first microphone, in 1876, the concept was part of his basic wire transmission system. Microphone design quickly evolved to serve the needs of telephony, public address systems for concert halls and public events, and even hearing aids. But in fact microphones were almost simultaneously used for sound recording. The earliest known recordings of the human voice (phonograph recordings) made in 1857, used a basic transducer and a vibrating stylus that generated modulated lines of sound waves on sheets of paper. But it was with telephony and radio broadcasting that microphones were later perfected and finally allowed the first electrical recordings in the 1920s, in which a microphone converted the sound into an electrical signal.

Microphones are now once again connecting humanity. Ironically, at a time when everyone on the planet has a phone in their pocket, we are plugging in microphones to input our voices directly to computers in order to better interact remotely—because that is how we are able to keep connected and working during the pandemic.

Long before we would dream of speech recognition, we perfected microphones to new levels. We miniaturized them and created new topologies that are flawless in terms of fidelity. And yet, consumers sheltering at home are again revisiting the “radio sound” and the natural boost that enhances the classic voice proximity effect to either record podcasts, stream performances live from home, or simply to communicate leveraging the “high-fidelity” voice quality allowed (sometimes) on VoLTE networks or by VoIP connections.

The pandemic truly disrupted this segment of the audio industry in interesting ways. Even though most people have smartphones, headsets, wireless earbuds optimized for calls, even computers with built-in microphones for conferencing, millions reacted to the fact that they were spending more time at home by investing in large capsule microphones positioned with cool desktop mounts, or with articulated boom arms, shock mounts and pop filters—radio-studio style. Some microphone companies saw their classic broadcast-oriented microphones sell in quantities they could have never imagined. The familiar shape of the Shure SM7B showed up in the homes of so many famous singers and musicians that it almost became a 2020 icon.

And podcasting absolutely exploded, with people adopting the concept and taking it to new heights, while companies such as Spotify spent hundreds of millions of dollars in podcast content and related technologies. RØde Microphones, which had introduced in 2018 the extremely successful RØDECaster Pro Podcast integrated production studio, gradually expanded the system’s features and podcasting range of accessories in direct response to market demand. With people at home looking for even more products to complement their RØDECaster Pro podcasting consoles—from cables to color-coded rings—they sold like crazy.

Actually, as Edison Research recently revealed, apart from the explosive growth in podcasting (production and consumption), News/Talk content (broadcast and streamed) also increased.

And while humanity is learning new lessons from this global pandemic, using technology in ways that will make sense for the future, we are learning to value simple things like the ability to remotely control devices, or simply being able to sanitize equipment after use. And those are basic requirements that will remain valued from now on.

Truly interesting trends are addressed in this edition of *audioXpress*, where we discuss how big and small manufacturers have adapted in 2020 and redefined successful new strategies. 2020 was the year of the audio renaissance, someone wrote and I totally agree.

Ironically, a year ago I enthusiastically wrote in *audioXpress* how 2020 would be the International Year of Sound (#IYS2020 - www.sound2020.org)—an initiative proposed by the International Commission for Acoustics (ICA) in response to a UNESCO Resolution.

As it happened with everything in 2020, the International Year of Sound wasn’t as expected. We only have even more reasons to want to celebrate it. As soon as possible.

J. Martins
Editor-in-Chief

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All Electronics Corp.	39	Menlo Scientific, Ltd.	59
Audio Precision, Inc.	45	Nureva	29
Brüel & Kjær North America, Inc.	33	Parts Express International, Inc.	68
Celestion	23	Powersoft S.p.A.	11
COMSOL, Inc.	9	Profusion plc	2
GRAS Sound & Vibration	55	QuantAsylum USA, LLC	43
Hammond Manufacturing, Ltd.	3	Redco Audio	57
Hypex Electronics B.V.	51	Triad Magnetics	67
KAB Electro-Acoustics	57		

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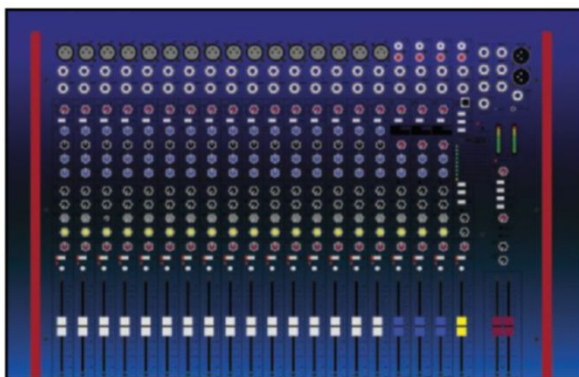
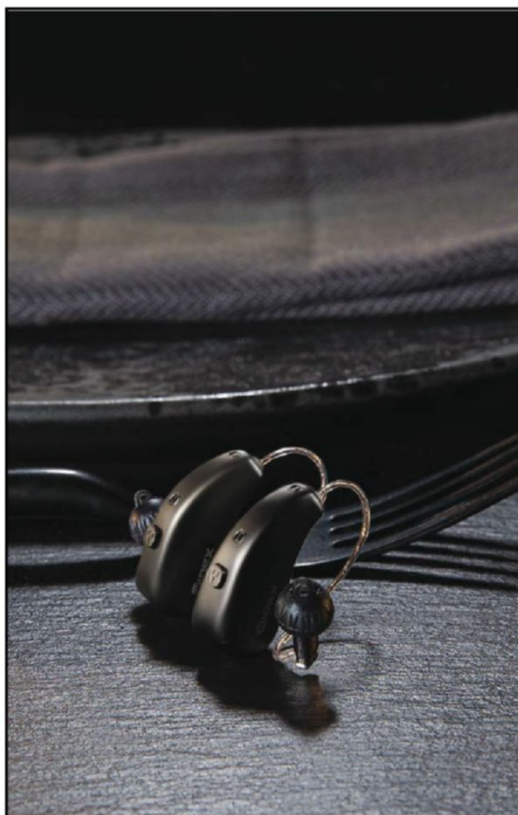


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● Features

34 Why Thousands of Virtual Microphones Matter The Impact of Acoustic Spatial Resolution in a 3D Space

By Erica Yorga (Nureva)

Thousands of virtual microphones. It's what Microphone Mist technology offers, and it sounds impressive. In this article, Erica Yorga explains what it really means and the impact thousands of virtual microphones actually have on your daily conference calls.

44 Widex Moment mRIC RD Hearing Aid

By Brent Butterworth

Can a hearing aid use DSP to deliver superior performance plus the convenience and versatility of the latest true wireless Bluetooth earphones? Brent Butterworth tests the Widex Moment to find out.

52 The Science and Physics of High-End Mic-Preamp Design (Part 1) Principal Mechanisms for Audio Designers: Engineering the Sound of Silence

By Mike Meechan

This series of articles investigates the principal mechanisms of noise, distortion, and interference that microphone preamplifier audio designers must understand, before exploring various circuit topologies used to maximize performance and minimize susceptibility. Finally, the ideal solution is considered.

60 X-Altra Moving Coil/Moving Magnet RIAA EQ Preamplifier (Part 1) Design Overview and MC Preamplifier

By Andrew C. Russell

For this DIY project, Andrew Russell shares the X-Altra Phono EQ Preamp, which is an accurate, very low noise, high-performance phono equalizer that will accept moving coil and moving magnet inputs.



● Market Updates—Microphones and Voice Interface Systems

MARKET UPDATE—MICROPHONES

8 From Home to the World Microphones Get Essential Status

By J. Martins

The topic for this Market Update couldn't be more relevant to the audio industry. Because if there was a product category that was completely disrupted in 2020, it was certainly microphones. Surprisingly, a lot of positive things happened for their manufacturers.

MARKET UPDATE—VOICE

22 Enabling Voice Capture and Developing for Voice Interfaces

By J. Martins

Because microphones are a critical front end for innovative applications in voice recognition and transcription—technologies that continue to show the same expansion trends in 2021—in this Market Update we explore what's new in voice capture and voice interface development.

● Departments

4 From the Editor's Desk

5 Client Index

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● Columns

AUDIO SCOPE

40 Design Considerations Imbedding a High-Output Buffer Inside an Op-Amp Loop

By Michael Steffes

From Home to the World

Microphones Get Essential Status

By
J. Martins
(Editor-in-Chief)

The topic for this market update article couldn't be more relevant to the audio industry. Because if there was a product category that was completely disrupted in 2020, it was certainly microphones. Surprisingly, a lot of positive things happened for manufacturers big and small, new and established. That's what this report underscores.

In February 2020, precisely when we were all starting to take notice that something was about to disrupt the whole world, *audioXpress* noted how the overall category was changing fast. As *audioXpress* reported in its second Market Update on this topic, the microphone market was consistently growing, particularly in professional applications, having exceeded \$1.8 billion US in 2019, according to market research published by Futuresource Consulting. And with the consumer space becoming increasingly vibrant and competitive, as technology was inspiring new use cases, and benefiting from the convergence with wireless audio and communications, the industry was seeing larger volumes than ever.

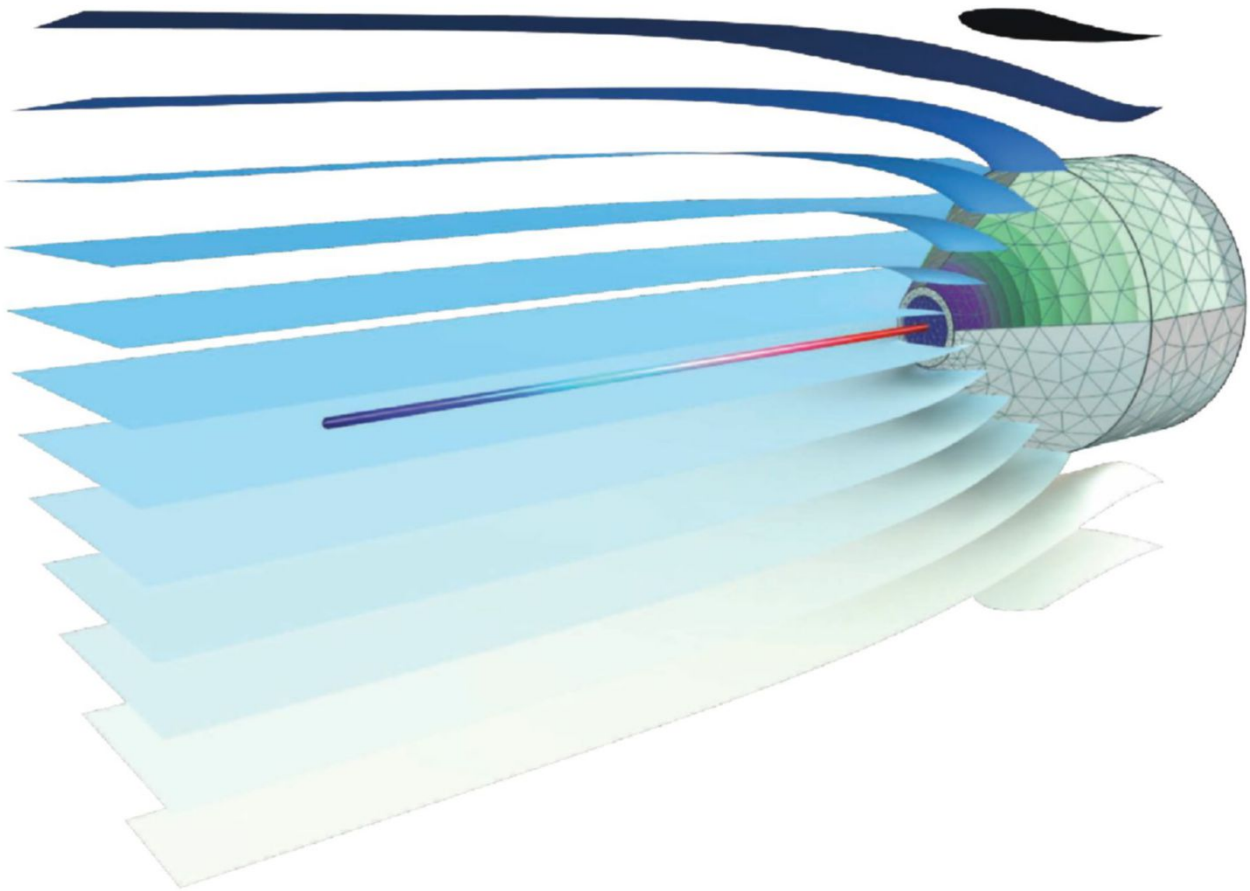
At the Winter NAMM Show in Anaheim, CA, in January 2020, all the microphone manufacturers in attendance introduced more options for studio recording, including more options for project and home studios. The trend for "virtual microphones" in the studio continued, with Townsend Labs and Slate Digital starting to gain momentum, with more musicians and producers recognizing the virtues of microphone modeling. Leading manufacturers (e.g., Sennheiser and Shure) had new, more affordable, and simpler to use wireless options for stage and

field production in general, and a few brands had taken notice of the popularity of podcasting, and had designed new models appealing specifically to that trending segment.

A few weeks before that—at CES 2020 in Vegas, NV—USB microphones for vlogging, podcasting, and gaming were a very noticeable trend, with even the gaming accessory brands introducing desktop microphones that actually resembled familiar large-diaphragm, radio-type microphones. But the consumer electronics market was overwhelmingly focused on voice recognition and voice control applications, where microphones are omnipresent and invisible but need to be more effective than ever—either when embedded on a simple trash can, or implemented in sophisticated true wireless stereo (TWS) earbud designs.

Some very successful brands on the crossroads between mainstream consumer, home studio, and vlogging (e.g., Rode Microphones and IK Multimedia, as well as Shure and Sennheiser), had so far been more focused on applications for the booming category of mobile devices, and had many new solutions dedicated for smartphones or tablets, designed to be used anywhere.

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



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

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

The COMSOL Multiphysics® software is used for simulating designs, devices, and processes in all fields of engineering, manufacturing, and scientific research. See how you can apply it to hearing aid design.

comsol.blog/cocktail-party-problem

With lots of people receptive to the idea of investing in a nice, high-quality microphone to use at home—preferably an instantly recognizable classic—Electro-Voice announced the introduction of the RE20-BLACK, a new low-reflection color option for the classic broadcast microphone.



And suddenly, with the pandemic restrictions, everything changed. Major live performances and world tours were abruptly cancelled. Entertainment venues shut down. Content production in general was stopped. Even commercial studios had to close temporarily, and rethink how they could safely handle sessions, if at all.

Office spaces all over the world were shut down and major investments in conference rooms and communication infrastructure for those buildings had to be canceled and adapted. In 2020, everyone was forced to work from home, quickly adapting daily workspaces to work remotely and communicate efficiently on the web. And even as the summer of 2020 allowed some easing on the lockdown restrictions, everyone understood that concepts had to be reimaged, allowing communications and work from anywhere to become a priority. And



The idea of repackaging existing microphones for use on Skype calls was already around in 2019, as Røde Microphones suggested in this promotional photo for its popular VideoMic NTG.

microphones were suddenly more important than ever to all of us.

The New Challenges

The microphone industry was quick to respond to these trends, reacting to the exploding demand for microphones that could be used at home for webcasting, meetings, podcasting, voiceovers, or general home studio use. But as no one had predicted how quickly the virus would spread globally and how long the health and safety restrictions would remain in place, no company was prepared for a whole new set of challenges.

It does make a difference when a radio or TV host speaks through a proper studio microphone, and the same is true with musicians and singers who seem to understand how important the microphone can be. Some singers can be really surprising in the way they play acoustic instruments and sing with a single microphone, always controlling their distance to the microphone. Inexperienced users—no matter if they are amateur singers trying to record a demo at home, or just talking to their friends on Skype or Zoom—don't know how different capsules react differently to distance, don't understand polar diagrams, and don't know what gain is until they turn it full up and hear the screaming feedback through their speakers... or their ears.

Using a microphone (e.g., the suddenly extremely popular Shure SMB7) requires the microphone to be connected to a proper professional microphone preamp and requires some experience to correctly set levels—which some musicians apparently are able to master (or they called for professional help).

Some users had the privilege of having home studios with proper room acoustics and the best available classic studio microphones—including some famous names who even have vintage Neumann U47s at home—but the large majority has no idea how to position and explore a large diaphragm condenser microphone with multiple polar patterns to their advantage, and that's how they could actually obtain better results from a less sensitive but extremely tolerant fixed cardioid, such as the SMB7.

As we could all see in those recorded or live streams with famous and regular guests appearing on TV calling from home, in most cases, people used whatever was most convenient and provided reliable results. And many of those Internet live-streaming appearances will now remain part of history.

One familiar vision that remains is the use of the familiar white Apple AirPods on notable ears—impressing positively for the clean and highly intelligible voice quality allowed by the built-in microphones, even under the challenging local



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EDM SUB



DV SUB



CINE SUB



M SUB



SCAN FOR MORE



Guests live on TV use Apple AirPods for both convenience and quality of the results. The AirPods' microphones and built-in voice optimizations allow for improved intelligibility and noise reduction, reducing undesired effects of room acoustics, even under the most difficult conditions of Internet streaming.

Bluetooth links and variable Internet conditions. According to the most conservative estimates, Apple AirPods and AirPods Pro with active noise cancellation (ANC) represented nearly one-third of the total headphone market sales in 2020.

audioXpress addressed the web conference and streaming communication challenges in a dedicated Market Update and feature articles, and will return to the topic in our April 2021 edition. We also intend to more closely explore the topic of microphone preamplifiers, which are now more relevant than ever.

Economic Impact

If there's one thing we don't know yet—and probably it will take some time to get a reliable picture—is the impact the pandemic has had on the overall segment. We know that large microphones manufacturers took a big hit in their most profitable product segment of large-scale multichannel wireless microphone systems for live production—as all the pro audio equipment companies also did. How much was the offset from other thriving application segments in the overall business, it's hard to estimate for now.

The strong off-axis rejection typical of a cardioid microphone, strong-close proximity effect, and excellent immunity to the surrounding acoustics, combined with a huge tolerance to strong SPL levels, made the Shure SM7B an ideal microphone for both singers and vocalists to use at home, but also for podcasters, gamers, and voice-over artists.



As we've discussed in our previous report, microphone companies typically characterize their business by market applications: pro, consumer, corporate, or industrial segments. Professional audio applications—stage, music, studio, and broadcast—actually didn't contribute much in 2020, and many products simply spilled over to other applications. Consumer applications, communications, business (conferencing), and industrial applications, all evolved positively, but again, frequently not serving the typically attributed applications.

The manufacturers' own commercial market segmentation by end application, such as microphones for voice (speech or singing), guitar (acoustic or electric), drums (with all the specific varieties from snares and cymbals to the bass drum), to sound recording for film and electronic news gathering (ENG), was again meaningless. Even if we did see strong investments in microphones specifically for studio recordings (studios have been able to keep going but hardly able to afford further investments), they were most probably not used in their traditional ways.

Of course, the two largest identified trends were the exponential increase in the sales of voice microphones, used in conferencing and communications, but also in microphones used for home studio recordings and live performance streaming, which tend to be predominantly acoustic.

And while in our previous report we highlighted emerging technology trends, such as the interesting adoption of software microphone emulations and the adoption of MEMS technologies for new application segments (e.g., test and measurement, environmental monitoring, and scientific research), all those have proven statistically irrelevant in 2020 given the complete disruption of the market, and the uncharacteristic distortions the pandemic has caused while those specific technology trends remain valid.

In 2019, as highlighted in our Market update, the corporate segment was the highest growing category, "outperforming all other applications by some distance," as market research firm Futuresource noted. All that changed in 2020. Large office spaces remained empty for most of the year and many will never return to their previous capacity use, while at the same time work from home and remote collaboration became the norm.

For much of 2020, microphone manufacturers have remained anxiously waiting for the return of those high-volume areas, while at the same time starting to plan and adapt their existing projects to whatever will be the market in 2021—knowing that installations have resumed with every easing of the pandemic restrictions.

While we cannot predict how this will evolve even in the next six to 10 months, we can pretty much expect that the dominant trends will remain and some will even be reinforced. Professional systems used for corporate communications and remote education will finally have time to adjust to the organizations' new way of working post-pandemic, and it will be interesting to see how these solutions will be transformed and evolve.

In the consumer space, aside from microphones meant as built-in components used in smartphones, laptops, smart speakers, smart TVs, and connected smart home devices, there are now increasingly new sophisticated microphone designs used in the desktop and even adapted with boom arms, podcast style. And as users have started to plan for more sophisticated and versatile home working spaces, there are also new interesting trends to adopt worn microphones with wireless transmitters, while a new category of fully integrated wireless systems is about to emerge and promises to become an interesting category.

Key Success Stories

One of the companies that was able to successfully leverage the last 12 months was undoubtedly Røde Microphones. The Australian manufacturer was already on a path to expansion and a large transformation, having initiated a very interesting evolution with new products and concepts targeting streamers, podcasters, vloggers, and new generations of media creators in general. And that strategy paid off in a gigantic way when the global pandemic hit.

Røde Microphones basically expanded its range of successful media creation microphones and solutions to respond to both the needs of remote work, while reinforcing packaging of complete sets available for the online retail channels.

As an example even targeting more professional applications, Røde expanded its Wireless GO range of field production solutions for reporters, videographers, and vloggers. This includes new microphone options for the Wireless GO ultra-compact digital wireless microphone transmitter, which already includes a built-in omnidirectional condenser capsule. Using its 3.5mm input, it becomes possible to connect an external microphone (e.g., the Røde Lavalier GO) for even greater flexibility. On the other extreme, the wireless receiver features mounts for all sorts of cameras, or clips to securely fit onto clothing or a camera strap. Audio is sent to the receiver using Røde's Series III 2.4GHz digital wireless transmission that delivers quality sound over 70m line-of-sight and is designed to work in locations with dense Wi-Fi and Bluetooth activity. All of this is packed into an incredibly tiny form factor—both the

transmitter and the receiver are super lightweight, weighing a minuscule 31g each—making it a perfect microphone for anyone from vloggers using compact smartphone rigs to filmmakers and reporters looking for a discreet, hassle-free wireless audio solution.

In 2020, Røde expanded the range with a fully white version (even the accessories are white!), and among many new options launched the Interview GO, an adaptor that instantly turns the Wireless GO transmitter into a lightweight wireless handheld mic for reporting to camera or interviewing in the field. The company was able to respond and adapt to the changing market trends because all its products are designed and made in Australia, in Røde's own state-of-the-art facilities in Sydney.

On a smaller scale on the US side, rounding out 2020, boutique microphone company Ear Trumpet Labs celebrated record sales, a triumph for small businesses everywhere during a year when the music industry has been one of the hardest hit.



Billie Joe Armstrong of Green Day, and Maren Morris perform with Shure SM7B microphones during the event "One World: Together At Home" presented by Global Citizen on April 18, 2020. (Photos by Getty Images)



Røde Microphones introduced white versions of the Wireless GO range, quickly adapting to changing market trends.

A mighty team of six powers Ear Trumpet Labs out of a small workshop in Portland, OR, where they handcraft each microphone. When the COVID-19 pandemic hit in early March, founder Philip Graham immediately closed the workshop and began formulating ways to continue production on the microphones with his team working from home.

After the initial shock and implementation of safety measures, the question then became how to effectively market and continue to sell microphones when musicians can't tour and perform. Known primarily for their live use, Ear Trumpet Labs began to see an uptick in users selecting the mics for their



Ear Trumpet Labs' microphones (e.g., the Edwina, Mabel, Louise and Josephine), originally designed for the stage, were endorsed by musicians appearing on countless live streaming sessions.

livestream concerts and home recording setups. Moving quickly, Ear Trumpet Labs built a page on its website dedicated to live streaming and home recording, and famous musicians endorsed Ear Trumpet Labs as a top option for those applications—proving that the quality of their products translates from the biggest stages to the most intimate living rooms.

Shure's Successful Pivot

Another remarkable success story is that of Shure. Remarkable because it explores consumers perception of its existing designs, repackaging them in a way that makes it even more appealing for current needs.

Forced to perform and stream from home, many musicians resorted to using a familiar but reliable microphone, the Shure SM7B, which quickly became one of the unexpected winners of the pandemic lockdown. Selling currently for \$399, the cardioid SM7B was already a great alternative for podcasting, voiceovers, or recording voice from home or untreated rooms. Because it's not a very sensitive microphone and it features a highly effective pop filter, protecting from explosive breath sounds, it allows very close proximity and still responds well without too much booming effect.

Inspired by the Shure SM7B's success, Shure cleverly introduced the new MV7, a dynamic microphone with both USB and XLR outputs for use with computers and professional interfaces alike. Basically, the MV7 expands on the famous cardioid studio microphone with software features. Users can connect via USB and explore additional setup features and Auto Level Mode within the complementary ShurePlus MOTIV app, which allows users to control vocal tone, and the distance-effect from the microphone.

Otherwise, the new MV7 retains the main features that made the Shure SM7B such an unexpected best-seller during the pandemic, allowing professional-level audio production results even at home.

Unexpectedly, Shure's existing (and recent) MOTIV USB microphone range—which apparently would be perfect for home users—did not sell as well, maybe also a consequence of the category being saturated with new and established alternatives at every price level. And if people had to be seen onscreen during Zoom calls, they might as well show up looking professional, by using a respected microphone like the SM7B, which screams "professional."

By offering a similar-looking design that sells for just \$249 US, Shure was able to offer the ideal response to this market trend, and at the same time meet the increasing demand for audio streaming

and recording solutions at home with the new MV7 Podcast Microphone. And the MV7 is even certified for use with the popular VoIP gaming solution, TeamSpeak.

Directly following the successful introduction of the MV7, Shure introduced the Shure MV5C USB microphone. Focused on improving voice audio quality and making video conferencing from home more effective, this microphone is a version of the already existing Shure MV5, which was optimized for microphone recording, supported by the ShurePlus MOTIV app for iOS platforms and offering DSP presets. But apart from having the same retro-cool look, the Shure MV5C (C for Conferencing) focuses on making life simple for people who need to enhance the audio quality of their own voices during meetings, allowing close proximity in radio or professional announcer-style, with an elegant, fully adjustable, desk-mount stand, selling for \$99 US.

Earthworks Audio Enters the USB Microphone Space

Another US company targeting podcasters, live streamers, gamers, and remote workers, Earthworks Audio recently entered the growing USB microphone space. Cautiously, Earthworks released ICON, a studio-quality USB microphone, and ICON PRO, a broadcast-quality XLR streaming microphone. Both products share the design and Earthworks partnered with another US company, Triad-Orbit, to create a custom desktop microphone stand.

The New Hampshire-based company had recently entered the highly competitive vocal microphone segment with the stunning Earthworks SR314 (handheld) and the SV33 (studio) models.

The original designs and unique qualities of these latest vocal microphones help place Earthworks firmly on the radar of an expanded market. But once the pandemic hit, the Earthworks production team quickly shifted focus to help isolated remote workers feel more connected, exploring successful elements from the company's latest vocal capsules.

"Whenever anyone records or broadcasts with an Earthworks mic, the reaction is always the same. You viscerally feel just like you're in the same room," says Gareth Krausser, COO of Earthworks. "That connection is needed now more than ever."

And so, "connection" became the guiding principle for the Earthworks development team. The goal was set to create elegant streaming microphones that look and feel as good as they sound.

The ICON condenser USB microphone is a cardioid design with an impressive stainless steel body that, like the SR314, is concurrently retro- and



Shure was able to quickly offer an ideal response to market trends with its new MV7 Podcast Microphone—a design akin to the iconic SM7B professional microphone.

futuristic-looking. And because a lot of podcasters and musicians already have invested in pro-audio interfaces with good-quality microphone preamps, Earthworks also launched the ICON PRO model, using a conventional XLR connector.

Both models directly appeal to the home user who needs to look good and sound great online, either recording a YouTube video or doing web interviews. The Earthworks ICON essentially offers studio-quality, consistent results with its integrated pop filter and the characteristic fast response (14 μ s rise time speed) of the company's designs. As Earthworks also details, the ICON USB was tuned specifically for an optimized sound signature of an expensive, larger diaphragm, condenser. And its off-axis rejection prevents room noise from entering the audio stream.



An ideal solution for working-from-home professionals, the Shure MV5C condenser cardioid microphone provides clear audio from any home office, without the need to wear headphones.

Useful features for these streaming applications include an integrated visual LED status indicator, intuitive analog mute and gain control, and headphone output with sidetone (allowing users to hear their own voice). Also important for these home applications, the ICON USB doesn't use foam, allowing for easy cleaning and sanitizing of the stainless steel windscreen.



Targeting podcasters, live streamers, gamers, and remote workers, Earthworks Audio released ICON, a studio-quality USB microphone, and ICON PRO, a broadcast-quality XLR streaming microphone.



Austrian Audio hit the jackpot with its OC818 studio microphone, which can be remotely controlled via its PolarPilot app (Android and iOS), allowing real-time wireless control over the microphone's polar patterns, high-pass filter settings, and attenuation pads, via Bluetooth. With this feature, users are able to explore fully different ways to capture a performance, and they can also work from a safe distance without the need to enter the vocal booth.

As part of the look, Earthworks partnered with Triad-Orbit to design and build a custom desktop microphone stand that is robust enough to hold and balance the stainless-steel microphone. For maximum flexibility, the integrated M2-R swivel ball joint can be disconnected and re-mounted on any studio mic stand or boom arm.

The ICON PRO microphone looks and feels similar to the ICON USB microphone but it is hand-tuned with an extended frequency response and the capsule has a faster Rise Time Speed of just 11.67µs. That extra speed makes the vocals even richer, fuller, and more detailed. Because it is an XLR broadcast microphone requiring 48V of phantom power, it offers extended headroom and dynamics. Both microphones are machined and hand-assembled in Wilton, NH. The ICON retails for \$349 and the ICON PRO sells for \$499.

New Microphones from Austria

A lot of innovation in microphones is coming from Austria these days. Austrian Audio opened its doors in 2017 following the closure of the AKG offices in Vienna. The new company started with a core team of previous AKG personnel from management, acoustics, electronics, test and measurement, mechanical design, RF/wireless, and software/firmware, all determined to create a complete catalog of products that would bring something new to the market, while being respectful of its heritage.

So far, Austrian Audio has successfully entered the studio recording market with the OC818 multi-pattern, dual-output, large diaphragm microphone, using a unique ceramic capsule design, combined with wireless remote control.

Following the company's first public presentation at Prolight+Sound in Frankfurt, Germany, in April 2019, Austrian Audio was present for the first time in the US at the NAMM Show 2020, showing the OC818 and the OC18, which is a fixed cardioid version, together with the company's first Hi-X50 on-ear, and Hi-X55 over-ear Hi-X (High Excursion) headphones.

And while the OC818 and the OC18 are not exactly affordable models that people would invest just to show on Zoom calls, many studios have found the microphones' innovative possibilities of wirelessly controlling polar patterns and remotely configure all the microphone settings at a safety distance to be appealing enough to invest in one or two units. Austrian Audio also saw a huge increase in demand for stereo configurations, motivating the company to offer a complete stereo set, containing a stereo adjustable bar, shock mounts, windscreens, mic clips, and cables—all in a carrying case with space for two optional OCR8 remote controls/Bluetooth dongles.

When Austrian Audio designed a microphone that

could be remotely controlled from a smartphone app, they were far from even dreaming that this would become such a valuable feature because of a dangerous virus that would force people to socially distance. But the feature was quickly found particularly relevant with people singing—helping singers, audio engineers, and technical staff to remain safe.

But when Harman relocated AKG manufacturing and operations overseas and closed offices in Vienna, not all the employees moved on to work at Austrian Audio. Some had previously left to start other companies—such as Lewitt Audio, founded by Roman Perschon in 2009, and which has recently seen a significant uptake in business.

Lewitt microphones, such as the very affordable but robustly built LCT 240 PRO—intended for home recording—found acceptance on a wider market looking for a versatile microphone for both voice and instruments. Great sounding on streaming, and offering professional sound quality results when used with a good quality preamp, this cardioid condenser microphone also benefited from the fact that it looks familiar enough to be impressive and is also a modern-looking take on AKG's C214 and C414. And for those who are looking for an extra level of quality but not willing to pay much more, Lewitt also offers the LCT 440 PURE cardioid condenser with an impressive 1" capsule and improved circuit design, minimizing background and self-noise.

Other AKG talent took the opportunity to establish their own companies and continue building upon the illustrious legacy of Viennese audio-engineering. Philipp Sonnleitner, a former software engineer and product manager at AKG, had a vision to create his own business by offering modern audio recording tools for people to use with their smartphones. In 2015, after countless conversations with musicians, he launched his company and introduced the mikme gold, his first product.

This compact microphone condensed everything one could find on a more traditional C414, including a 1" externally polarized condenser capsule, plus a circuit design that features a built-in 24-bit and 96kHz studio-grade digital recorder. All in a square-shaped block, half-the size of a C414. And of course, the mikme gold was totally controlled from a smartphone app.

That product allowed the company to expand, and take the concept further, evolving from the initial vision of an all-in-one device for musicians and "create great audio in the mobile space," to focus more on the consumer and expanding live streaming space.

Already in 2020, supported by successful crowdfunding campaigns, mikme introduced two new



concepts. One is called the mikme pocket, designed for vloggers, journalists, and YouTubers.

The pocket is essentially a compact wireless audio recorder and transmitter, allowing up to 3.5 hours of continuous use, and weighing just 97g. It can be used with the smartphone app or as a standalone audio recorder. The mikme app can wirelessly connect with up to 3 mikme devices simultaneously, automatically syncing smartphone recorded video with the sound recorded by those devices.

A second concept was also introduced, resulting from the changing environment of the pandemic, and is called mikme connect. This more affordable design was directly inspired by the needs of remote work sound and combines the existing mikme concept in a USB-powered microphone. The mikme connect uses a 2/3" electret condenser capsule and built-in digital processing (2x Cortex M4) to optimize the captured sound.

On its website, the company features three interesting videos that show the mikme connect compared with the sound recorded using Apple AirPods, the built-in microphone on a MacBook Pro, and the sound of a standard wired headset with microphones, as used for UC&C. Not surprisingly, the mikme, placed at a correct distance from the user (close) in a desktop stand, sounds how a professional

The affordable Lewitt LCT 240 and the successful LCT 440 Pure are both versatile cardioid condensers that proved well for home recording.



The mikme connect USB-powered microphone is an example of a new successful approach to microphones.



The DPA 4097 CORE Micro Shotgun was designed as a solution to social distancing protocols in interview situations.

cardioid microphone should sound, while effectively cancelling the room acoustics.

The surprising part (for some) is the realization that the second best sound comes from the (wireless) Apple AirPods, followed by the built-in microphone on the MacBook Pro, with the really bad sound originating with the “pro” headset.

Still, the mikme connect with a good quality desktop stand costs less than the AirPods Pro, and offers a low-latency, direct 3.5mm stereo jack output, with the only glitch being the use of a dated and inconvenient Micro-USB connector.

New Production Needs

As we mentioned, socially distancing is a new reality of the pandemic, and it’s easy to understand its impact on the use of microphones for different applications. Audio capturing quality directly depends on microphones being as close as possible to the

source, and unless the situation allows using a professional shotgun, it’s hard to maintain adequate distance when a microphone requires operational handling, such as in interviews or close action.

The microphone industry is still adapting to a completely new set of requirements, from sanitizing materials, to designing new accessories for handling microphones from a safe distance. Danish manufacturer DPA Microphones, a leading vendor in the professional content creation segment, was one of the first companies to adapt with a completely new product, adding the 4097 CORE Micro Shotgun Microphone, optimized to pick up speech from a distance. Incorporated into the company’s new 4097 CORE Interview Kit, the mic has been designed for use with a lightweight boom and windjammer, particularly useful for journalists in a socially distanced world. The design allows users to conduct interviews from up to two meters (6.5’) away.

For other situations, recording is increasingly demanding a set and forget solution that can be tweaked in post. And for those applications, using Ambisonics is an obvious option. As soon as the COVID-19 situation calmed down in China and production resumed, Hong-Kong-based miniDSP announced the launch of its new ambiMIK-1, an affordable First-Order Ambisonic (FOA) microphone with an embedded preamp, analog-to-digital converters (ADC), and an USB processor.

As a fully integrated FOA microphone, the miniDSP ambiMIK-1 is ideal for spatial audio and content production, offering a compact solution with high sample rates (up to 192kHz) in a portable package. With applications ranging from VR/AR and music recording, the ambiMIK-1 doesn’t require any bulky external preamp or XLR cables. A simple USB cable to a tablet/laptop and it’s ready to go!

Using four low-noise 14mm capsules in a slim aluminum body and nylon protective frame, the ambiMIK-1 package was engineered for durability and optimal performance without breaking the bank. The capsules feed an on-board four-channel microphone preamp and an ADC with 120dB dynamic range. The microphone has its own XMOS processor, providing audio recording support at sample rates up to 32-bit/192kHz.

Last but not least, miniDSP partnered with Dirac Research for the design of a VST plug-in specifically calibrated for the ambiMIK-1 with more than 1200 measurements. The controls include Yaw/Pitch/Roll/Output gain and Output Format (B format or Binaural).

For other manufacturers, the pandemic restrictions also gave them the opportunity to relaunch products that had so far not been



The new miniDSP ambiMIK-1 four-channel Ambisonics microphone enables spatial audio recordings and is also ideal for remote recordings.

recognized. With the need to socially distance, the use of wireless audio transmission is an obvious choice for any production.

Sennheiser offers the Memory Mic—an affordable solution to capture audio at a distance, or even remotely—but the product was not fully understood by potential users. Now, the company is able to offer its wireless Memory Mic and its free companion app as a solution for teachers, course instructors, and vloggers to use on streaming videos, conferencing, and online learning, and even for families to stay in touch. Working as an external microphone that is simply attached to the clothing with its magnetic clip, like a professional lavalier microphone, or placed in the most convenient location for sound capture, the Memory Mic is a reliable wireless solution for a yoga instructor who needs to produce video lessons, or anyone who needs freedom of movement while online.

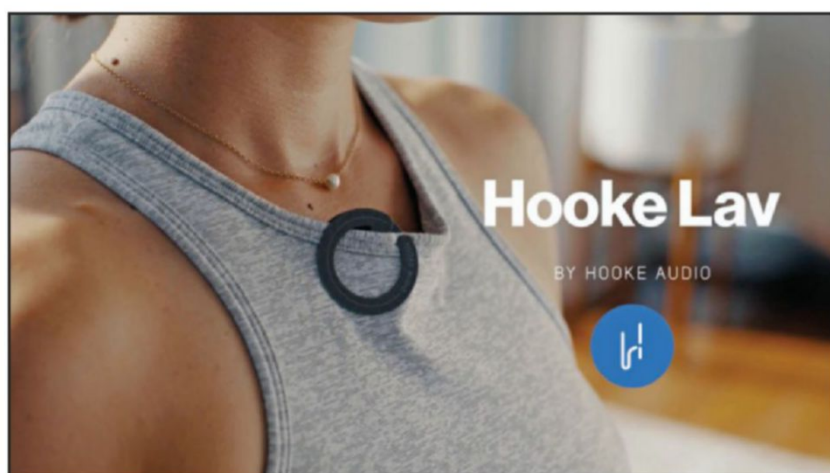
In fact, the concept of a wireless microphone for those use cases seems so appealing that it motivated another very successful crowdfunding campaign from Hooke Audio, the same company that developed the Hooke Verse Bluetooth binaural headset—wireless earbuds with microphones—and attempted without success to compete with Sennheiser and its AMBEO headset in that space.

Now, Hooke Audio has returned with the Hooke Lav, a “lapel” friendly design that looks nice on-camera and promises to leverage updated Bluetooth 5 technology to actually work. With the Hooke Lav project, the company is promising “the first wireless, wearable, dual-channel, Bluetooth lavalier microphone designed to capture clear, professional audio (for video).” Like the Sennheiser Memory Mic, the Hooke Lav is able to record and/or stream to any compatible Bluetooth device—allowing social distancing. The Hooke Lav uses a microelectromechanical systems (MEMS) microphone array and promises up to 7 hours of record/transmission, or up to 24 hours, when using the internal storage of 8GB in island mode. It can also be connected by cable to a camera or smartphone, to record for as long as the device has storage and/or power.

Certainly an interesting concept, it is probably going to be delayed by the challenging manufacturing and supply chain conditions of the pandemic, particularly severe for Kickstarter companies with little market credibility.

The Gaming Perspective

Meanwhile, a thriving segment in 2020 was the use of microphones in gaming products (e.g., gaming headsets) and the appearance of a new generation of



dedicated gaming microphones, all obviously turned into multi-purpose solutions for online applications.

As Ben Yeung, the CEO at Fujikon Industrial, one of the largest factories of headphones and consumer audio products in China characterized, “gaming is a market where brands specify products with all the best features for half the price.” And many users were quick to notice that the latest flagship gaming headsets with boom microphones had better voice quality than many more expensive office communication headsets, and naturally started to use them during the pandemic.

As an example, ASUS, a large manufacturer of smartphones, computers, and electronics, created its Republic of Gamers (ROG) brand, which includes gaming headsets that can be paired with the company’s products, from laptops to gaming-optimized smartphones. Recently, ASUS partnered with Swedish digital audio pioneer Dirac to optimize immersive sound experiences and sound quality, creating a more balanced, clear sound, so dialogue remains crisp and voices intelligible—which is key for multiplayer collaborative gameplay. The Dirac-optimized ROG Phone 3 series of gaming headsets

The Hooke Lav wireless microphone was a Kickstarter sensation in 2020, reviving a concept that previously failed to excite the market.



When Harman launched the new AKG Lyra multi-mode USB microphone, they designed it to be used in the studio for music recording but also for podcasts and consumer recordings, YouTube videos, and more. In 2020, the AKG Lyra turned into an instant hit, with its Adaptive Capsule Array that provides user-selectable capture modes, and it is packaged with robust AKG K371 headphones for \$249.

already includes more than 10 different models, all offering features that are way above standard consumer-level headphones. As an example, the recent, ROG Delta S USB-C headset is even equipped with a Hi-Res ESS Quad-DAC, large, high-quality drivers, and are the first headphones with a full MQA Renderer built-in.

In fact, these gaming headset companies are paying more attention to high-resolution audio (HRA) than many other high-end audio oriented brands. HRA is supported in gaming consoles and available content at par with image technologies (e.g., 4K and HDR),



Reacting to the exploding demand for microphones that can be used at home for webcasting, meetings, podcasting, voiceovers, or general home studio use, PreSonus announced the new Revelator (\$179.95), a bus-powered, USB-C microphone design with selectable polar patterns, preset configurations, and loopback audio.



Launched at NAMM 2020, the new Sontronics Podcast Pro was a timely success story for the British company. Designed specifically for podcast, gaming, voiceover, broadcast, and audiobook applications, with a tight supercardioid pattern, three layers of inner protection and a built-in pop-shield, the Podcast Pro screams professional, looks great on camera in shiny red and retails for \$149.

addressing a very large new potential market. That is also the reason why Tidal and Qobuz are among the streaming music services that are creating strategic partnerships with gaming brands.

And even other brands with no experience in the gaming segment (e.g., Audeze and Bose) decided to join the fray. In 2020, Bose introduced its first gaming headphones, cleverly repackaging the extremely popular Bose QuietComfort 35 II ANC model as a gaming headset. For that, Bose combined the already-proven features of the QuietComfort 35 II design with a detachable gaming module and attachable microphone boom, turning it into a 2-in-1 gaming and lifestyle headset. The QC35 II Gaming Headset will respond perfectly to the demands of hours of competitive gaming and music listening, with a comfortable fit, with just a few added controls on the earcups.

But the gaming accessory brands had already discovered some other lucrative opportunities in the segment—and that included desktop microphones. HyperX, the gaming division of Kingston Technology Co., launched its HyperX QuadCast microphone for streaming and audio recording in 2019—and naturally this sold very well in 2020. The HyperX QuadCast features a built-in anti-vibration shock mount and offers four polar pattern choices, gain control, a built-in pop filter, and a tap-to-mute feature with a cool built-in LED light—all for just \$139.99.

More recently, the company launched the HyperX SoloCast USB cardioid microphone, a more affordable option (\$59.99), already designed for streamers and content creators. Of course, the SoloCast benefited from the excellent reviews the QuadCast received, while helping the company to fend off the numerous USB microphone offers selling for \$29.99 on Amazon. The SoloCast's small size and rotatable stand with adjustable tilt positions gives it the ability to fit in small or tight places (e.g., under a monitor), which shows the level of attention that designers are paying toward streaming applications. And the SoloCast, like the QuadCast, is certified by Discord and TeamSpeak—the gaming communication platforms that are also being used simply for staying in touch by many non-gaming users.

Unexpected Consequences

According to Futuresource's research, 2020 had a positive impact on the gaming industry. Households in lockdown have gravitated to gaming as an immersive and escapist form of entertainment—an ideal antidote to a troubling situation. As a result, not only have existing gamers been able to spend more time gaming, others have rediscovered gaming, and some people have engaged with gaming for the first time.

With more than 40% of the worldwide gaming accessories retail market value, gaming headsets (typically fitted with microphones, built-in or detachable) account for the lion's share of the market, with Futuresource forecasting 26% year-on-year volume growth in 2020. Looking to the brands, the top four gaming headset players—Turtle Beach, Logitech G, Hyper X, and Razer—dominated in terms of worldwide volumes in 2020, capturing nearly two in every three dollars of overall consumer spend.

The gaming software market was worth an estimated \$154 billion US in 2020, and gaming hardware represented \$15 billion US. Not surprisingly, even traditional audio brands are also entering the gaming headset market, and Futuresource predicts the rising demand for wireless connectivity, surround sound, and spatial audio rendering will make traditional gaming headphones a thing of the past.

In contrast, the office-based headset market, which was steadily increasing in recent years, has suffered major disruptions, with an accelerated transition from traditional RJ9 and DECT wireless headsets to USB-connected headsets. "Working from home also means that consumer behavior is a far greater decider on which headsets people prefer to use," notes Futuresource.

While the pandemic has provided a huge opportunity for the office headset market, the demand generated by the boom in home working in 2020 has led to significant supply shortages. This has resulted in increased sales of all headset types, with the greatest increase coming outside the traditional office headset market, towards consumer and gaming headsets for use, at least partially, for work, adds the market research firm. "Where end users themselves are purchasing headsets, they are more likely to choose a product that they can use both for work and for leisure. However, where a business is equipping its workforce to allow them to work from home, they are far more likely to purchase a dedicated office headset." 



Relaunch, Repackage, Inspire. A winning strategy manufacturers are using to increase sales during the pandemic. Røde Microphones launched a new range of Røde Vlogger Kits featuring a high-quality Røde VideoMic for smartphones, bundled with a tripod, phone grip, LED light, and accessories. All to make life simpler for content creators and fosters sales through online sales channels.



The Bose QuietComfort 35 II Gaming Headset was a brilliant adaptation of an existing product at a time when production hurdles made it difficult to bring completely new designs to market, while the gaming category was showing signs of resilience, with increasing sales as people spend more time at home.

Resources

ASUS | www.asus.com

Austrian Audio | www.austrian.audio

Bose | www.bose.com

DPA Microphones A/S | www.dpamicrophones.com

Ear Trumpet Labs | www.eartrumpetlabs.com

Earthworks Audio | www.earthworksaudio.com

Futuresource Consulting | www.futuresource-consulting.com

Hooke Audio | www.hookeaudio.com

HyperX Gaming | www.hyperxgaming.com

Lewitt GmbH | www.lewitt-audio.com

Mikme GmbH | www.mikme.com

miniDSP | www.minidsp.com

Presonus | www.presonus.com

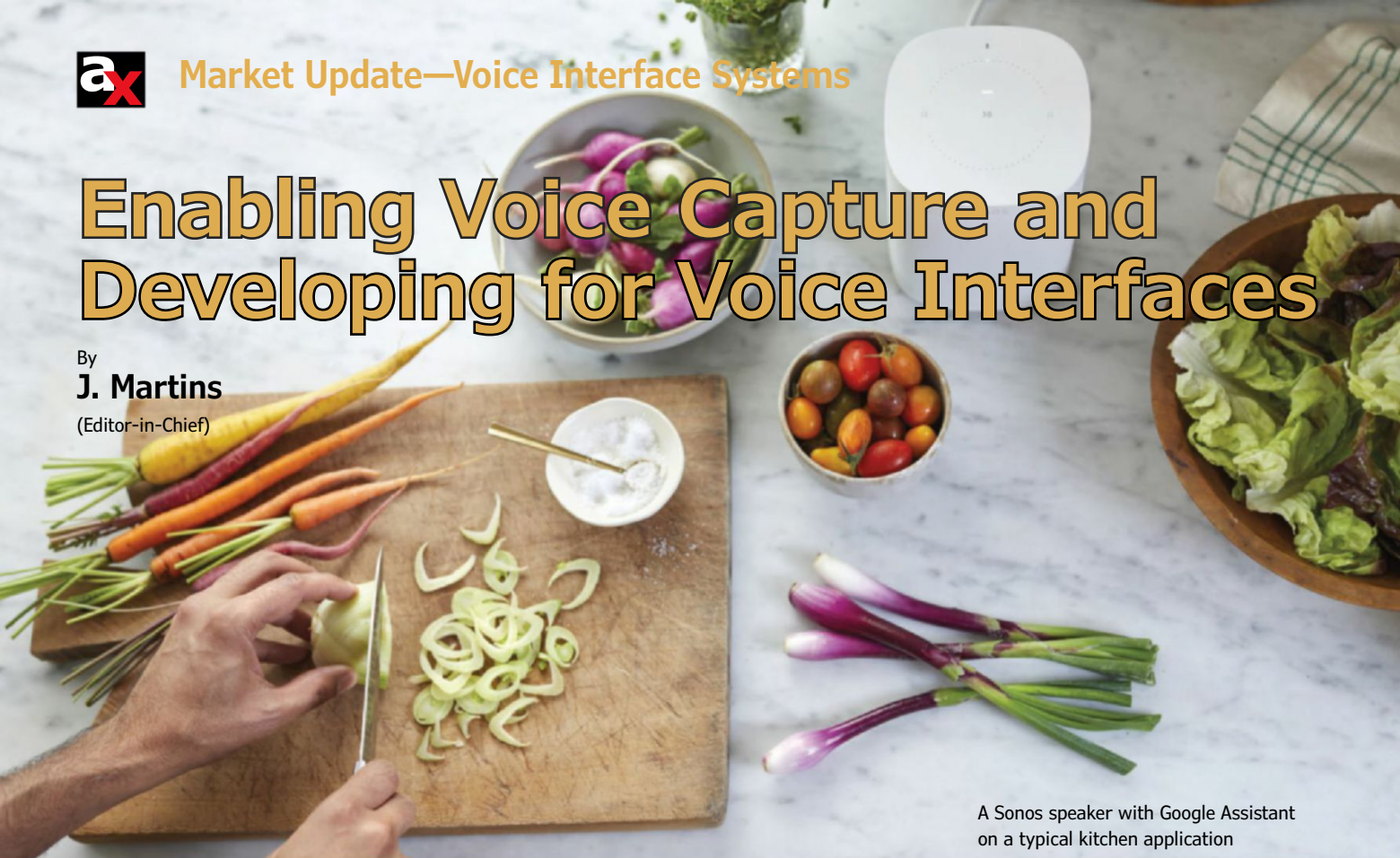
Røde Microphones | www.rodemicrophones.com

Sennheiser | www.sennheiser.com

Shure | www.shure.com

Enabling Voice Capture and Developing for Voice Interfaces

By
J. Martins
(Editor-in-Chief)



A Sonos speaker with Google Assistant
on a typical kitchen application

Because microphones are a critical front end for innovative applications in voice recognition and transcription—technologies that continue to show the same expansion trends in 2021—in this Market Update we explore what’s new in voice capture and voice interface development.

2020 has been an unpredictable year and most of the market data available suffers from being obsolete, partial, or relying on trend projections that can change at any moment. One thing that raises broad consensus is the fact that voice recognition and voice interfaces continue to expand and have become more essential than ever—even if the technology is still in its infancy.

What this will effectively mean is difficult to anticipate, given that voice technologies are still restricted in many important language markets.

For now, its main impact is being felt in consumer technology available in English and Chinese speaking territories. But even with strong language restrictions, voice engines, voice assistants, voice control, and voice interfaces are already considered one of the most significant interdisciplinary achievements, combining cutting-edge developments in computer science, machine learning, and artificial intelligence. But its implementation is also one of the most significant challenges ever for audio technologies and acoustics science.

Primarily, voice technologies can be applied and directly benefit basic applications in communications or even dictation—areas that are now quickly changing precisely because of the intense research in voice capture and recognition. A new study from Juniper Research

found that the global number of Voice over LTE (VoLTE) users will approach 5 billion by 2024; rising from 2 billion in 2019. And the market research firm identified that, along with smartphones, smart speakers will be key to enable IP-based voice services, forecasting more than 220 million smart speakers will be used to make calls to landlines or mobile phone numbers by 2024. In addition, the research anticipates that VoLTE networks will be increasingly leveraged to direct mobile calls to smart speakers. In turn, users will link incoming traffic to both their smartphone and smart speakers simultaneously. And that’s even before wearables connected to 5G networks become massively available.

This means that devices optimized for voice recognition and for voice assistants will be adopted for uses that are not necessarily revolutionary, but will definitely benefit from the massive rate of adoption of technologies that would be otherwise impossible to commoditize.

At the same time, we continue to learn what else exactly these voice capture and voice interface systems will enable us to do, other than ask Alexa for the weather.

Convenient and Sanitary

According to the recent Adobe Voice Survey 2020 study, which surveyed 1,000 voice technology users in

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Innovation, performance and reliability make Celestion the drivers of choice for a growing number of big-name PA brands. So when you need superior quality compression drivers or professional loudspeakers, check out about the speakers the pros use.



celestion.com

CELESTION



A Google Edge TPU machine-learning development kit with Google's purpose-built ASIC is designed to run inference at the edge. The type of solution used to develop applications for devices such as Smart Displays with Google Assistant.



The new Amazon Echo Show 10, equipped with cameras and advanced image processing, has the ability to follow a user when interacting with Alexa and is an example of a new generation of AI-powered devices designed for voice interfaces at home, but also for communications and security.



With its HomePod smart speaker, Apple also places Siri front and center for user interactions and takes it a step further in multi-device integration and intercommunication.

the US, voice technology adoption of voice assistants is clearly accelerating. And while voice interactions might be considered a safe, sanitary alternative to touch interfaces, they will be hard to implement for general user interactions in public spaces or offices, in contrast with its use at home, which is considered a welcome convenience by users familiar with the technology.

The Adobe survey still indicates that less than one-third of users already familiar with Amazon, Google, and Siri voice assistants (in the US), use it for shopping or any other critical and complex interactions. In contrast, users feel perfectly comfortable using voice to automate tasks, ask knowledge questions, and use digital assistants the same way they would ask someone for instructions, or make a call to order pizza. And the notion that the user is actually recognized and understood is vital in voice interactions.

Strangely, 86% of users say they would see voice technology being used for everyday tasks like opening a door (56%), choosing a floor on the elevator (55%) or using a vending machine (49%)—things where using voice would be hardly desirable or practical. And yet, 62% of the surveyed respondents in the Adobe study admit they feel awkward using voice commands in public, or interacting with a voice assistant when other people are present.

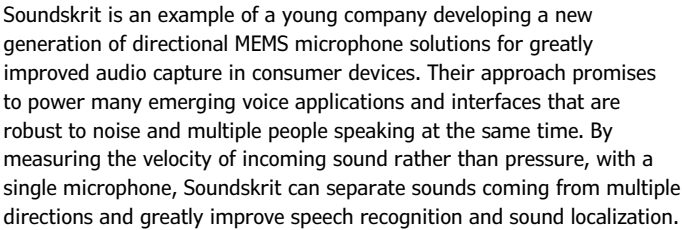
The Adobe survey also shows an interesting result regarding the top benefits of voice assistants identified by actual users. 63% mention "hands-free," 60% call it "convenient," close to half say it "saves time," it's "easy to use" and gives them "freedom to multitask, while only 35% like the idea of voice assistants being "always on." And, 31% of users consider voice interfaces to be a "sanitary" option.

Expanding Voice

Using voice for simple control tasks based on commands or keywords is currently one of the most reliable applications for voice recognition, and yet it's by far the least useful or even practical. Requesting to hear an album or a song, or to see a movie, dictating messages or chat interactions, requires complex interactions with databases and even artificial intelligence (AI) engines.

Saying "open" or "seventh floor" is much simpler to implement, and yet, neither command is safe nor practical with multiple users in predictably noisy environments. As anyone with experience using voice knows, in the simplest interactions, 89% accuracy is not enough.

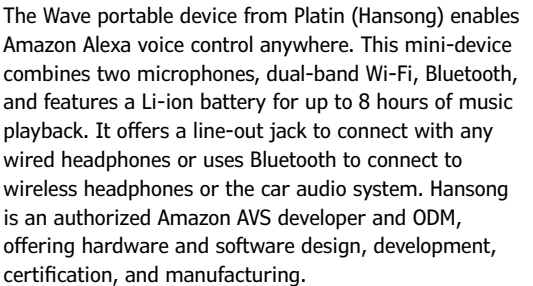
In general, most studies reveal that awareness of voice technologies is rising very quickly, and market research is optimistic in predicting that by 2025



Also, the current example of voice assistant implementation in smart speakers and smartphones still needs to be transposed to other more critical environments (e.g., inside a vehicle) and transposed to edge devices that don't depend upon connecting to the cloud.

Fluent.ai is an example of a company focused on the hardest challenges of voice recognition, such as multilingual speech understanding solutions, multi-user identifications, and ultra-low power edge implementations.

And yet, everyone across multiple industries is fully committed to advance the applicability and reliability of voice technologies, and that starts with the voice capture front-end, voice engines, and processors. In 2020, even under the circumstances, announcements in this front continued at a steady pace. In this article, we selected a few recent important announcements in voice technology development, a topic on which we will expand in our April 2021 edition, addressing on-device and wearable applications. 



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Fluent.ai | <https://fluent.ai>

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NXP Semiconductors

Solutions for Connected Devices with Voice Interface

Fully recovered from the attempted merger with Qualcomm, NXP Semiconductors was extremely active in 2020, and the company seems to be fully committed to be at the forefront of connectivity and voice interface technologies.

NXP announced that its Wi-Fi and Wi-Fi/Bluetooth combo solutions are now pre-integrated across its i.MX RT MCU platform, allowing developers to create the optimal pairing of processing with connectivity. And NXP combined its MCU and Wi-Fi/Bluetooth platform solutions within its MCUXpresso Software Development Kit (SDK) to simplify and accelerate the development of connected designs.

With this new level of integration, NXP also expands the connectivity capabilities of its EdgeVerse edge computing and security platform. By pre-integrating driver support within the MCUXpresso SDK, NXP helps developers accelerate compliance and streamline Wi-Fi or Wi-Fi/Bluetooth combo deployments.

These new platforms make it possible to mix and match the right MCU with the right Wi-Fi or Wi-Fi/Bluetooth combo devices, providing developers with the flexibility needed to meet the performance and power requirements of their connected designs, whatever the application.

And by using the MCUXpresso SDK, developers can easily combine wireless connectivity with AI/machine learning capabilities, display controllers, and graphics accelerators integrated into the i.MX RT family. The pre-integrated Wi-Fi/Bluetooth drivers have been verified and tested, and can use Amazon Web Services as a framework for connected products as well as device, gateway, phone and cloud connection.

Far-Field Offline Voice Control Solution

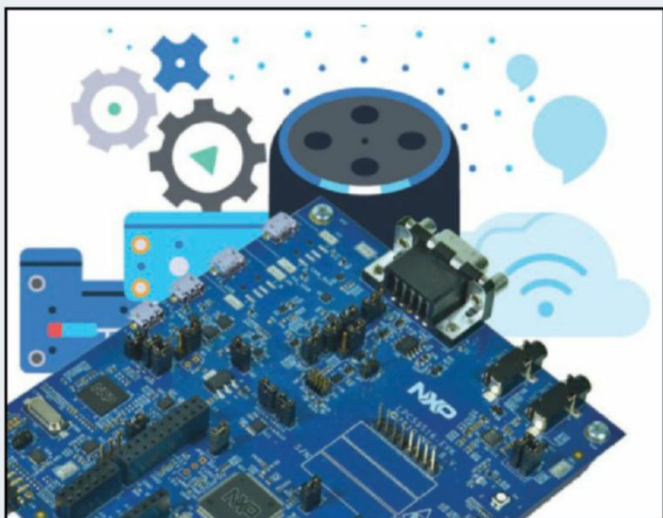
NXP also announced availability of its voice solution, SLN-LOCAL-IOT, a fully integrated development platform for offline voice control. The solution consists of a full hardware module design and associated software required to implement far-field voice control with a customizable wake word and local commands.

The new NXP far-field offline voice control solution is based on the i.MX RT106L crossover microcontroller (MCU), which addresses the market demand for embedded voice control in a broad range of smart home and many other application areas.

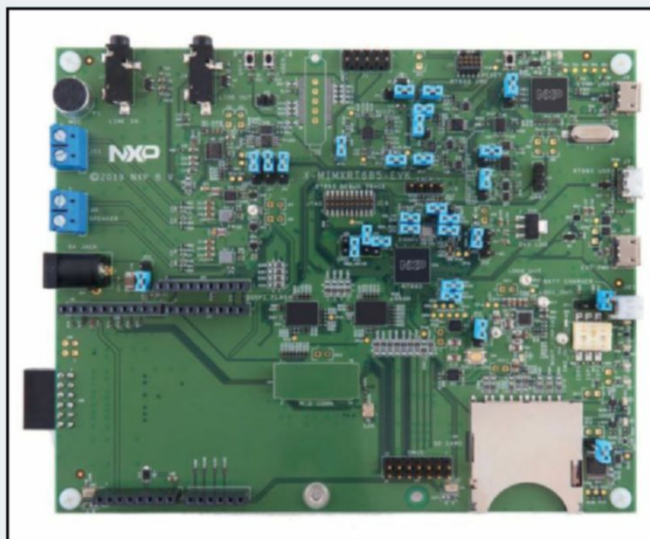
The availability of NXP's i.MX RT106L MCU-based voice control solution makes it possible for original equipment manufacturers (OEMs) to leverage a complete production ready solution to drastically reduce system cost and time to market.

Additionally, the local voice control solution eliminates the costs and complexities of cloud connectivity and can help protect user's privacy by enabling audio to be processed directly on the device. This is achieved with integrated, machine-learning-based, automatic speech recognition (ASR) technology, delivered in partnership with Sonos, which recently acquired Snips, a pioneer company in embedded voice recognition software for businesses, products, and services.

The i.MX RT106L is an edge-ready solution-specific member of the i.MX RT1060 family of crossover MCUs targeting offline embedded local voice control applications. It features NXP's advanced implementation of the Arm Cortex-M7 core, which operates at speeds up to 600MHz. The i.MX RT106L processor is licensed to run NXP's turnkey local voice control software solution,



The NXP LPCXpresso55S16 Development Board based on the Arm Cortex-M33 architecture includes Arduino UNO compatible shield connectors and has been validated to support Alexa Connect Kit (ACK)—helping to accelerate the development of voice-enabled applications.



The NXP i.MX RT600 evaluation kit is fully supported by the NXP MCUXpresso suite of tools and software and Cadence's Xplorer IDE for i.MX RT600, offering complementary DSP function libraries and audio codecs.

which includes the Voice Commands by Sonos technology, a machine-learning far-field audio front end, acoustic echo cancellation, barge-in, ambient noise reduction, beamforming, playback processing, and many additional features.



NXP far-field solution for embedded voice control

And NXP Semiconductors also launched the i.MX RT600 crossover microcontroller (MCU) family, an ideal solution for ultra-low power, secure-edge applications, including audio, voice and machine learning. Machine learning support is optimized using a Glow neural network compiler. The new MCU family features an Arm Cortex-M33 processor running up to 300MHz, optional Cadence Tensilica HiFi 4 audio/voice digital signal processor (DSP), and features NXP's advanced embedded security technology—EdgeLock 400A.

For home applications based on NXP's i.MX 8M family of chipsets, developers are now also able to leverage the full Dirac digital audio platform, which includes both Dirac digital audio solutions and the Dirac suite of tuning tools. This enables OEMs to customize the product's sound experience consistently across a range or brand catalog.

NXP also partnered with Alango Technologies, DSP Concepts, and Sensory to provide high-performance voice pre-processing and recognition software, plus professional audio libraries and tools. The integrated HiFi 4 DSP is supported with the NXP eIQ for Glow machine learning compiler, accessible in the latest MCUXpresso SDK.

mcuxpresso.nxp.com | www.nxp.com

XMOS

XMOS XVF3510 Voice Processor Updates for Integrated Solutions Development

XMOS announced new firmware variants, supporting USB and built-in applications of its XVF3510 voice processor, further opening the door to voice-enabled smart product development. The two variants—XVF3510-UA for USB accessory devices, and XVF3510-INT for built-in solutions—incorporate purpose-designed algorithms that ensure a clear voice stream is captured from across the room, even in noisy environments and with audio streaming from the device itself.

The British company says the new firmware variants are designed to address the rapidly evolving demands of the far-field voice market, enabling the development of high-performance voice-enabled smart products at a low eBOM. The new purpose-designed algorithms for voice processing introduced in the new XVF3510-UA and XVF3510-INT variants include advanced acoustic echo cancellation (AEC), enabling the detection of voice signals even through high-volume audio interference, combined with interference cancelling (IC), working intelligently to remove static point noise and ignoring audio from other devices. Developers will also benefit from advanced noise suppression, removing both stationary and non-stationary diffuse noise sources, plus automatic delay estimation control (ADEC) that adjusts signal latency to guarantee optimal AEC performance, and automatic gain control (AGC), tuning

output channel level for optimum results.

The new solutions also support real-time system control and parametrization, monitoring and adapting algorithm control, plus system configuration and GPIO. Requiring just two microphones, the XVF3510 is optimized for audio visual products, smart home devices and appliances, health and fitness solutions and conference calling.

www.xmos.com/products/voice/amazon



XMOS VocalFusion 2-Mic Dev Kit for Alexa Voice Services, based on the XMOS XVF3510 voice processor

DSP Concepts

Pushing Audio-Processing Further for Voice Interfaces

DSP Concepts and its Audio Weaver platform are leading the way in audio processing for voice applications and optimize front-end solutions for voice technologies. Audio Weaver already powers more than 50 million devices worldwide, and the work of the California company is generating particular interest in the automotive industry. Audio Weaver is already central to the audio experiences of companies like Tesla, Porsche, Bugatti, and Volkswagen. DSP Concepts recently confirmed it has secured another \$3 million strategic investment from Subaru, adding to the \$28.4 million capital already raised to date with other investors, including Porsche, Sony, MediaTek, and ARM.

The past year was a strong moment for the company's technology with the launch of AWE Core OS, a new variant of the audio-processing engine powering its Audio Weaver platform, now optimized for multi-core development, tailored for auto manufacturers. This development enables brands to dramatically reduce costs and components typically required to build new audio features. And DSP Concepts built the multicore solution specifically for automotive brands to address complexities and costs associated with creating new audio experiences in that space.

AWE Core OS gives audio engineers the freedom to move audio processing to the main SOC without worrying about running out of CPU on a single core. There are myriad benefits of multi-core audio development, but the most important are the dramatic reductions in the costs and components typically required to design and manufacture new audio products. Using the Arm Cortex family of processors as an example, developers can benefit from AWE Core OS for real-time applications, such as the advanced audio processing for noise reduction and voice interfaces running simultaneously.

TalkTo Audio Front End Qualified by Amazon

Also in 2020, DSP Concepts announced that TalkTo, its suite of proprietary microphone processing algorithms that use adaptive learning technology to deliver a quiet-room experience in noisy and unpredictable environments, has been qualified by Amazon for Alexa built-in devices using Alexa Voice Service (AVS). This enables more hardware brands to easily build Alexa built-in devices that will perform well even in noisy situations, thanks to TalkTo's leading sound cancellation technology.

Amazon rigorously tests all software solutions before adding them to its hub of qualified development kits, and TalkTo is one of just four audio front-end reference designs that currently meets Amazon's standards. The other three are from Cirrus Logic, Intel and Microchip, while there's also Amazon's own Alexa Premium Far-Field Voice development and evaluation kit.

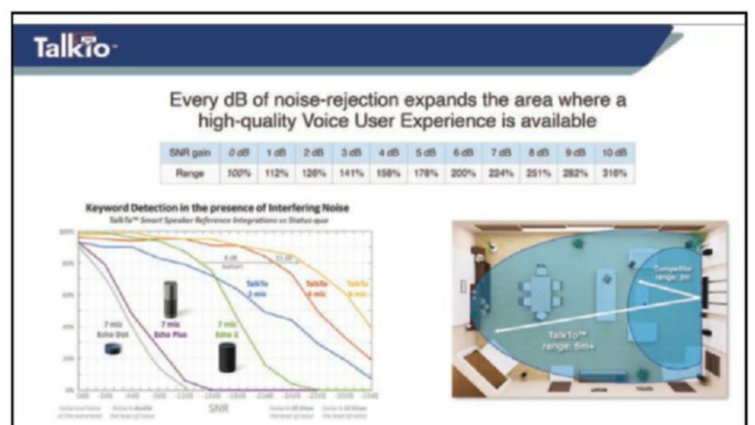
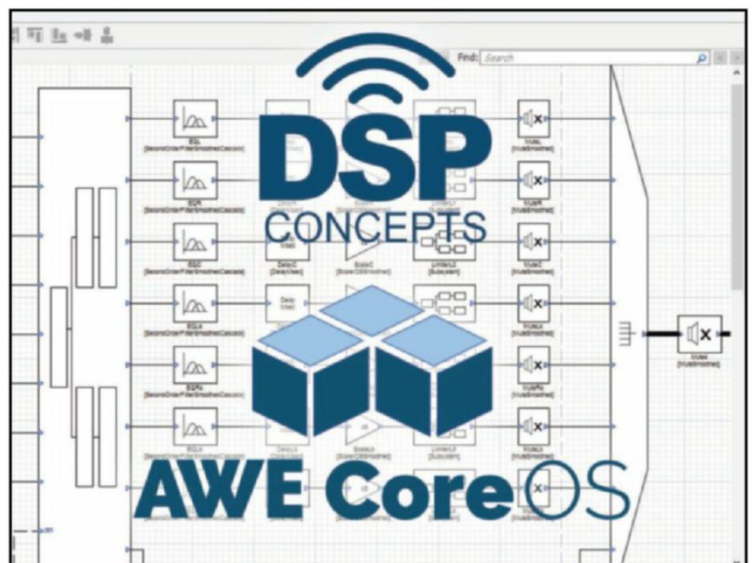
DSP Concepts' TalkTo audio front end is able to cut through everyday noise with Adaptive Interference Cancellation, as much as up to 30dB of unreferenced noise to activate Alexa in any environment.

And because of TalkTo's noise cancellation capabilities, product designers don't have to worry about mic placement with regard to picking up on a user's commands. DSP Concepts says that, with this solution, retail kiosks and food ordering systems will be even able to use voice inputs to pick up on user commands in noisy environments.

The turnkey audio front end software solution also allows increased design flexibility, supporting two, four, or six microphone arrays, as well as custom configurations with Acoustic Echo Cancellation (AEC) for up to seven channels. The solution can also be optimized for target applications including televisions, soundbars, set-top boxes, smart speakers, hearables, and wearables.

With DSP Concepts' TalkTo, developers can access a free trial to evaluate the software and prototype new devices before purchasing a production license.

<https://w.dspconcepts.com>





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nureva.com/microphonemist



STMicroelectronics

All-In-One Amazon-Qualified Far-Field Voice Reference Design

STMicroelectronics (ST) has recently released an Amazon-qualified reference design package for smart connected devices leveraging Amazon AVS voice integration, which can be used to create Alexa Built-in products using simple microcontrollers (MCUs). The company believes this level of integration will be essential to enable easier access to their voice interaction features and deliver more rewarding experiences.

The new reference design eases development of natural-language understanding and voice-based user interaction for smart home devices and smart appliances, featuring an improved audio front end created with DSP Concepts for far-field audio detection.

ST's Amazon-qualified solution jumpstarts the design of Alexa-enabled appliance controllers, with Alexa voice-user-interface software already integrated on high-performing MCUs of the energy-efficient STM32 MCU family. These Arm Cortex-M MCUs are ideal for low-cost, small, and simple connected devices that require state-of-the-art features such as far-field audio capture and natural-language understanding.

ST's AVS Integration for AWS IoT Core reference design contains a compact 36mm x 65mm main board that combines a high-performance STM32H743 MCU and Wi-Fi module. Unlike other devices commonly used for Alexa products, such as digital signal processors (DSPs) and flashless processors, STM32 MCUs integrate all the necessary system features including powerful audio front-end processing, local wake-word detection, communication interfaces, and memory, including RAM and Flash, in a single chip.

Created by DSP Concepts, and distributed by ST, the front end comes with a free Audio Weaver tool license to help users easily fine-tune their designs. This creates an audio front end with support for TalkTo far-field voice detection, even in noisy environments and



with closely spaced microphones.

Leveraging the features of the high-performance STM32 microcontrollers, users can even add enhancements such as a second wake word, extra local commands, and a graphical display, for an even more compelling user experience.

The development platform includes audio-capture software with Advanced Audio Front End (AFE) with noise reduction, echo cancellation and advanced beam-forming signal processing for far-field audio detection, as well as audio output software.

Further simplifying prototyping and development, the reference design hardware includes an audio daughter board as a separate module. This contains an ST FDA903D audio codec, user LEDs and buttons, and two MP23DB01HP MEMS microphones spaced at 36mm for size-constrained products (e.g., power switched plugs). The hardware's modularity lets users implement a custom daughter board if application-specific mic spacing, acoustics, and user-interface definitions are needed.

As ST explains, the bill of materials (BOM) for the reference design is engineered to be less than \$10 when purchasing the components in high volumes.

www.st.com | www.dspconcepts.com

DSP Group

100 Million Smart-Voice SoCs Already Shipped

Demonstrating the importance of voice as a user interface in applications ranging from smartphones, tablets, and laptops/PCs, to the smart home, wearables, and all types of connected devices, DSP Group announced it has shipped its 100 millionth SmartVoice system-on-chip (SoC), marking a major milestone in the journey of voice user interfaces (VUIs) from a novelty to a necessity. This milestone is testimony to the importance DSP Group's customers place on end-user VUI experience.

"SmartVoice was launched less than five years ago and has quickly emerged as the go-to platform for voice user interfaces across such a broad spectrum of customers and applications, covering home and office electronics, hearable and wearable products, as well as for security

and industrial use cases," says Ofer Elyakim, CEO of DSP Group. "As voice, AI, and sensor fusion continue to grow, I believe we are barely scratching their potential."

At the core of DSP Group's SmartVoice technology are a suite of low-power, multi-core SoCs that include dedicated artificial intelligence (AI) and machine learning (ML) accelerators. These are supported by the company's HDClear voice algorithms as well as partners' algorithms. Together they enable accurate and reliable Always-On voice trigger with minimal power consumption, as well as voice command and automatic speech recognition (ASR) for local and/or cloud-based processing.

www.dspgroup.com

New VoiceHub Portal for Wake Word and Voice UI Development

Sensory—the Silicon Valley, CA, company pioneering AI at the edge and the de facto standard for enabling a voice UI on apps and devices—unveiled VoiceHub, a new online portal that enables developers to quickly create wake word models and voice control command sets for prototyping and proof-of-concept purposes.

Sensory's VoiceHub empowers developers with free tools to immediately create custom wake words and voice command sets for their applications. Designed with flexibility in mind, the tool enables users to easily select languages and model sizes through drop down menus, while its intuitive interface allows to quickly build vocabularies in dozens of languages with no coding experience required.

These projects take just moments to put together and some models are trained and downloadable within an hour of submitting them. VoiceHub outputs wake word and voice command set models, compatible with a companion Android application for quick prototyping, or as code for specific target DSPs for more advanced proof-of-concept testing. The tools offer vast flexibility, allowing developers to create wake word models, either custom branded or based on today's most popular voice assistant platforms, and command set models targeting a desired memory footprint. This makes it great for all applications, ranging from ultra-low power, resource limited wearables to high-power, high-performance appliances on the edge.

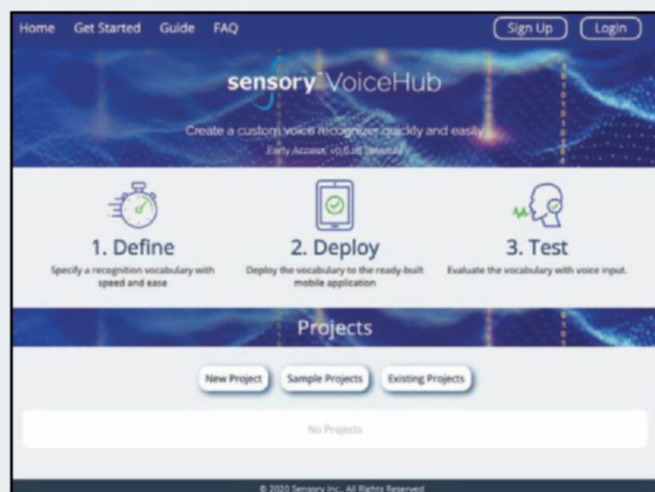
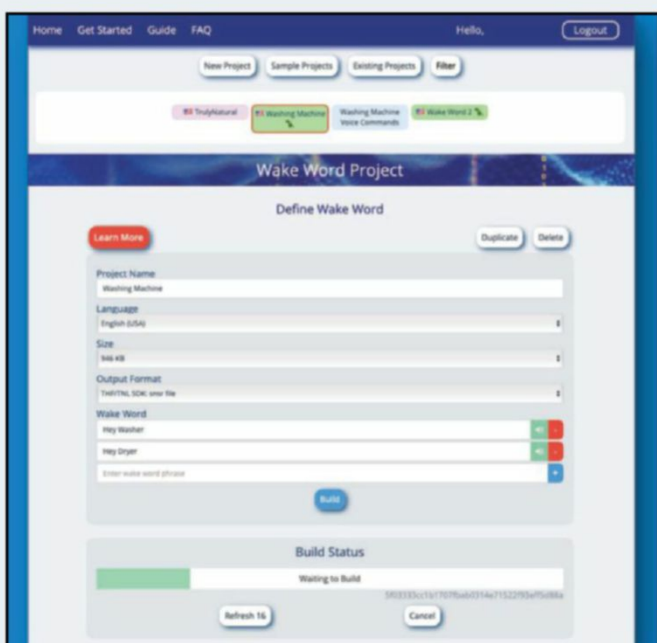
Based on Sensory's industry leading TrulyHandsfree technology—the same technology powering the voice user experience on more than 1 billion apps and devices—VoiceHub supports numerous languages, making it ideal for testing voice control across global product lines. Additionally, since VoiceHub trains voice models similarly to TrulyHandsfree, the wake word and voice control models created in VoiceHub are very accurate and in most cases suitable for mass production.

Testing conducted by Vocalize.ai—the company that Sensory acquired in February 2019 and maintains as an independent benchmarking and testing services provider for speech technologies and natural language interfaces—indicates that VoiceHub-generated wake words provide accuracy and performance equal to or better than that of industry benchmarks, such as Amazon's Alexa.

Following the announcement, Sensory is inviting interested parties to request an invitation to access VoiceHub for their own prototyping and proof of concept projects.

Sensory also anticipates a steady stream of VoiceHub updates and new features, including support for more languages, expanded DSP platform support, and the ability to quickly develop large vocabulary natural language models. For more information about VoiceHub or to request an invitation code, visit voicehub.sensory.com

www.sensory.com



Creating a new wake word project for a new application using VoiceHub

ArkX Labs

EveryWord Amazon AVS Far-Field Reference Design Kit

ArkX Laboratories, a joint venture between hardware design and development firm Surfaceink and consumer electronics manufacturing company Ark USA, announced its EveryWord Ultra Far-Field Reference Design Kit has been qualified by Amazon AVS.

ArkX Labs' EveryWord portfolio solves the biggest voice problems faced by OEMs, including capturing voice in the presence of a loud TV and other noisy and reverberative environments. Unlike conventional solutions that only hear in the X-Y plane, EveryWord hears, captures, and suppresses in all three X-Y-Z dimensions even with only two to four mics in one plane.

The advanced far-field development kit features Cirrus Logic SoundClear Technology and NXP i.MX 8 host processor technology and is a powerful solution for manufacturers dealing with noisy environments, different mic array geometry challenges, and the need to support three or five channels for surround use cases.

The EveryWord Development Kit comes with a full Amazon AVS voice stack pre-installed and exceeds all Alexa Qualification Tool and Reference Solution far-field test requirements in both two mic and four mic configurations. Using the company's proprietary 3D audio processing technology to improve its beamforming capabilities, it supports full-duplex far-field voice communication, allows concurrent voice service and voice communication, and does not require source ducking for reliable interaction.

The advanced far-field technology increases the accuracy and reliability of "Alexa" wake-word triggering and command recognition at far-field distances and in noisy environments; it comes preloaded with Amazon Pryon Lite Wake-Word-Engine and accepts Sensory Wake-Word engine. Additionally, it supports 8+ meters far-field operation with two, three, or four microphones and arbitrary mic array geometries, including circular, square, triangular, linear, or 3D.

The AEC is scalable to mono, stereo, three-channel, and five-channel (surround movie or music) use cases. The unit supports far-field voice call mode and in-call voice triggering and is also suitable for wall-powered devices or portable battery-operated products with low milliwatt consumption.

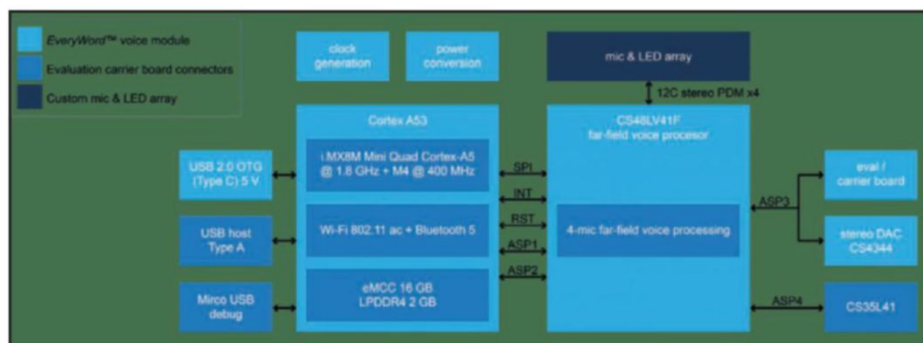
Other features include AVS compliant LED indicators and mic-mute button, Google Voice compliant mic spacing, an expansion bus for connecting GPIOs, I²S/TDM, SPI, I²C, and PDM mics and web-based firmware control console for simplified interactive operation, mic array selection, acoustic tuning, and diagnostics.

The platform is also suitable for use with all major voice assistants including Google, Siri, Cortana, Bixby, Baidu, Alibaba, and Tencent.

www.arkxlabs.com



ArkX Labs EveryWord Ultra Far-Field Reference Design Kit



Block diagram of the EveryWord far-field development kit, featuring Cirrus Logic SoundClear Technology and NXP i.MX 8 host processor.

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Why Thousands of Virtual Microphones Matter

The Impact of Acoustic Spatial Resolution in a 3D Space

By
Erica Yorga
(Nureva)

Thousands of virtual microphones. It's what Microphone Mist™ technology offers, and it sounds impressive. But what does it really mean? What impact do thousands of virtual microphones actually have on your daily conference calls?

Thousands of virtual microphones add up to more than just extra pickup points in the room. They represent a conceptual change to the status quo of physical microphones and beamformer systems. They alter our basic understanding of acoustic pickup in conference rooms and other meeting spaces.

Using an approach that focuses on discrete 3D locations and does not rely on broad coverage zones, we can deliver optimized analysis and solutions that result in audio conferencing systems that are more precise and capable. Put another way, if we move from using broad coverage zones that aim in the direction of a desired sound source and instead analyze sound at many thousands of spatial points (locations) in a room, we have the opportunity for a reset. A virtual microphone is any one of the thousands of predetermined locations that is a spatial point in three-dimensional space that the array can specifically focus on.

By applying a new approach that analyzes a room three dimensionally, audio characteristics can be determined for each one of thousands of locations in a room. That wealth of sound information can then be independently evaluated for desirability and quality, and automatic adjustments to the system can be made accordingly.

The Audio Conferencing Status Quo

There are two classical approaches to audio conferencing microphone pickup. One approach

uses one or more discrete microphones (**Figure 1**). Systems based on this approach may range from simple, such as a lapel mic attached to the person in the room giving a presentation or lecture, to complex (e.g., a collection of distributed goose neck microphones on a table or pendant-style microphones suspended from the ceiling).

This classical approach aims to provide specific coverage to a small target zone and/or broad coverage to a larger zone or zones defined by the polar pattern of the microphones. Although audio pickup performance can be good when a talker is close to the microphone, it degrades significantly when a talker moves away from the microphones or changes their orientation to them. Moving talkers can be especially hard to pick up, because when they stand to walk and talk, they will most likely leave the coverage zone of the tabletop microphones. The talkers are constrained to the limited coverage of the microphone system. This may not be suitable for meeting situations that are dynamic and that may require a talker to move to a different location, such as a presentation device, to illustrate a point.

If multiple sound sources are co-located within the coverage zone of any one microphone, the system cannot discriminate between them. That may not be a problem if all those sound sources are desired, but it will be a problem if one or more of the sound sources is a noise that is not desired, such as an HVAC system or the sound of somebody typing on a keyboard or shuffling papers. The microphone will pick up all the

sounds in that coverage zone and postprocessing will be required to deal with the unwanted noise.

The other common approach to audio conferencing uses beamforming arrays. These arrays, composed of multiple physical microphones configured to work as a beamforming array, use preconfigured coverage zones based on expected room use (**Figure 2**). The objective of the array is to amplify the sounds within the target zone with higher gain than is typical in a system that uses discrete mics and attenuate sounds that are outside the target zone so that reverberation and undesired noises are significantly reduced. These systems are called beamformers because each zone always begins at the location of the microphone array and extends outward, widening from there and typically extending to a room boundary.

A beamforming array may have just one zone (e.g., a beam-tracking array) or as many as a dozen or more zones configured to cover the room based on expected use (e.g., a multizone array). Because the beams can be narrowed, the system can reduce unwanted noise sources by targeting specific coverage zones and excluding others. However, even a narrowed beam does not provide enough spatial granularity to provide high-spatial resolution. That is, a beam deals with sound based on direction, not location. If an undesired noise source is within the beam, the beamforming system has the same limitations that a system based on discrete microphones has.

Undesirable noise sources still need to be dealt with by postprocessing using specialized noise filters and algorithms. If talkers are located outside the beam, such as at a display device, or are moving throughout the room, there is no way for them to know whether they are within a beamformer zone or not. With a discrete microphone system, the talkers can see the location of the physical mics. However, with a beamformer, the talkers cannot see where the beamformer is pointing or is configured to point so they may inadvertently leave the preconfigured coverage zones.

Both of these classical approaches are based on preconfigured broad coverage zones and within those zones identify the loudest sound sources in a room, which may or may not be the talker, and attempt to optimize the audio performance. In each of the classical approaches, the full-room acoustic coverage may not be adequate because acoustic spatial resolution and density is low and continuous acoustic monitoring of the full room in its entirety is usually not feasible. Also, each of the classical approaches is based on the expected use of the room. If actual use of the room deviates from the expected

use, the system typically has to be reconfigured. Obtaining detailed information about the acoustic properties and precise location of all sound sources in a room is difficult with broad coverage approaches.

In contrast, Nureva has developed a unique and innovative approach that covers the whole area by analyzing the three-dimensional space in its entirety with a high degree of spatial resolution. That allows the microphone system to obtain precise

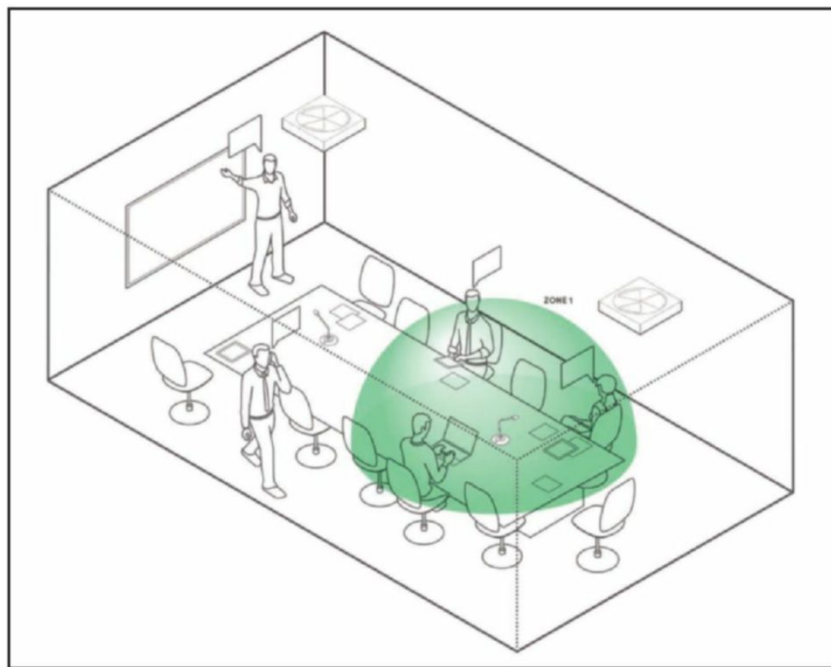


Figure 1: A tabletop microphone example

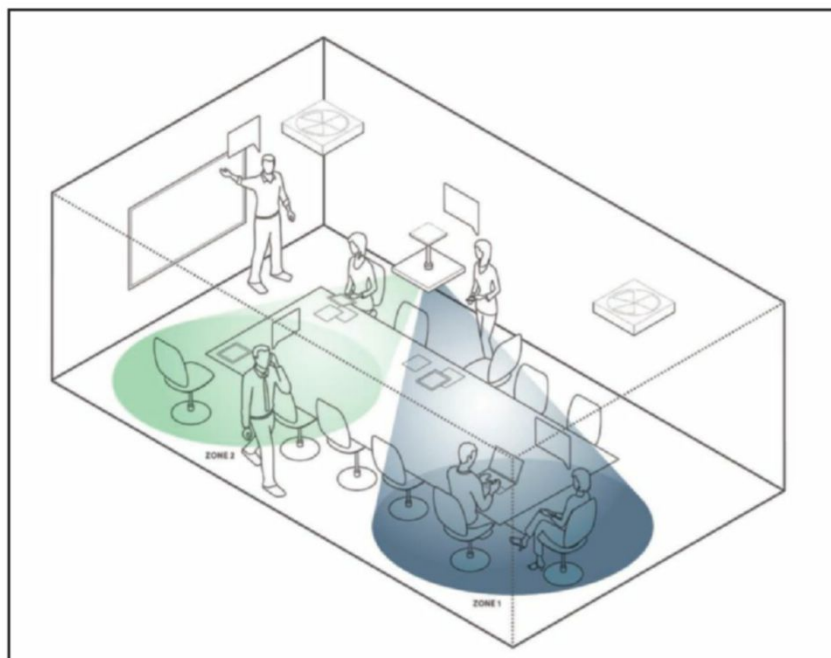


Figure 2: A beamforming array example

acoustical information at thousands of individual locations. Since the measured spatial granularity is very high, all acoustic sound sources regardless of location can be identified and managed concurrently. As a result, the system can provide a full-room acoustic perspective that optimizes the audio pickup performance to a level that is simply not feasible in lower spatial resolution systems.

A Novel Approach to Acoustic Spatial Resolution

Nureva realized that to improve the audio experience in meeting spaces, a new approach was needed. Our objective? Obtain better, more precise acoustical information by measuring sound characteristics in many discrete locations in the whole of an acoustic space—rather than simply optimizing an active sound source anticipated to come from one or more directions. We needed a new and innovative method to acoustically resolve a space to a finer degree than other audio conferencing systems.

To achieve this, we needed to look at a room as a three-dimensional acoustic space (**Figure 3**). Nureva has developed patented microphone array technology (called Microphone Mist technology) that can resolve an acoustic space in three dimensions, using thousands of evenly distributed virtual microphones. The result? A full-coverage

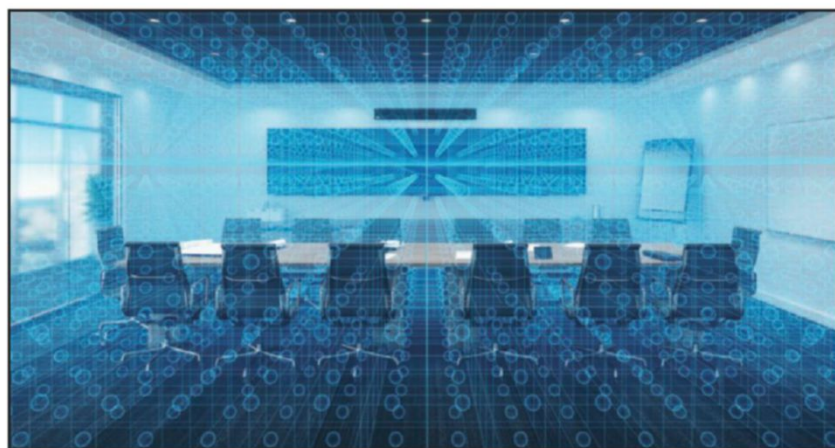


Figure 3: Microphone Mist technology

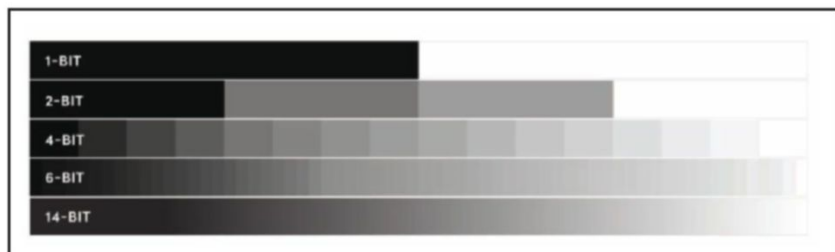


Figure 4: Color bit depth

grid that provides precise acoustic information at a higher resolution and density coverage than can be achieved using the classical approaches.

The concept of high spatial resolution is helpful in understanding what makes Nureva's approach to the acoustic space so unique. Though it's a new approach to audio conferencing, it is similar to high-resolution technologies in other fields.

Do We Need Higher Resolution?

When you buy a TV or camera, image resolution is one of the performance specs we know is important. A 4k picture is better than a picture rendered in 1080p. Higher resolution produces a better experience. It's the same with other formats and technologies—colors, music, images, and even telescopes. Higher resolution means more higher quality information and processing that allows a better experience.

For example, **Figure 4** captures the effect of bit depth on the color palette. When an image has a bit depth of 2, we get four color options. As we increase to 14 bits of resolution, we get 16,384 color options. As the bit depth increases, so does resolution of the color palette within the image, meaning that finer color details can be resolved. It becomes clear that the use of a higher bit-depth description of color is preferable for displaying and analyzing images.

In **Figure 5**, the low-resolution 4-dpi image of the guitar is blocky compared to a high-resolution 300-dpi image. Increasing the resolution means a more precise and detailed image can be displayed. Details that are obscured and blurry in the low-resolution image become very evident with a higher resolution image. We can perform better processing and analysis on the image data.

The same thing applies to music. In **Figure 6**, the table shows that a 4-bit recording has less resolution than one done in 24 bits. With a higher bit depth, we gain an improvement in signal to noise, higher dynamic range, less quantization error and so on.

The advantages of using greater bit depth and the resulting increase in data resolution are well understood and shared across digital platforms. These same benefits also apply to audio microphone systems that subdivide an acoustic space into finer and finer granular detail. If we resolve a space to smaller and smaller acoustic zones, we can describe and understand the sound sources and space better, resulting in optimized processing sound sources based on their own individual acoustic characteristics.

Resolution in the Acoustic Space

When it comes to audio conferencing, it is useful to divide a space into acoustic regions to

help provide sufficient coverage. The room can be divided into physical microphone coverage zones or virtual microphone locations. In the case of discrete microphones and beamformers, each zone starts from the center of the aperture of the mic system and is defined by the shape of the polar plot for each configured zone. In the case of Microphone Mist technology, each region is centered on the location of each of the virtual microphones in a 3D space. Each location can be assigned an individual spatial resolution value. The total resolution value indicates how many points of spatial granularity the microphone system is able to resolve in the 3D space.

Figure 7 shows a single discrete microphone and distributed discrete microphones in a space. If we assign a spatial zone value to each coverage zone, which in the case of a discrete mic starts at the center of the physical mic, we can see that a single mic has one individual spatial zone and a total of one spatial zone in the space (Figure 7a). In the case of multiple discrete microphones, three spatial zones are shown for a total spatial resolution value of 3 (Figure 7b). Although the coverage zones are large, the microphone systems are not able to discriminate individual sound sources within a single zone. This makes for a low total spatial resolution value. Just as with color and images, the more we can divide up the acoustic space, the more spatial resolution we have in measuring and describing it.

Figure 8 shows a generalized beamformer coverage pattern. The same type of quantification can be applied by assigning an individual spatial value to each zone. Figure 8a illustrates a three-



Figure 5: Image resolution

SIGNAL-TO-NOISE RATIO AND RESOLUTION OF BIT DEPTHS			
# BITS	SNR	POSSIBLE INTEGER VALUES (PER SAMPLE)	BASE-TEN SIGNED RANGE (PER SAMPLE)
4	24.08 DB	16	-8 TO +7
16	96.33 DB	65,536	-32,768 TO +32,767
24	144.49 DB	16,777,216	-8,388,608 TO +8,388,607

Figure 6: Music resolution

zone system that results in a spatial resolution value of 3. Even in the most complex implementations, coverage zones are usually limited to less than a few dozen and in this case six zones (Figure 8b), with a spatial resolution of 6. It’s clear that beamformers produce a small total spatial resolution value to describe the whole space.

Both systems are able to find the sound source in each coverage zone the same way a 4-dpi resolution image can display broad colors as fuzzy blobs. But discriminating sound sources within any one zone is problematic. And if a sound source is outside the coverage zone, it’s ignored entirely. The inability of either classical approach to divide the space into a high-density acoustic grid limits how the

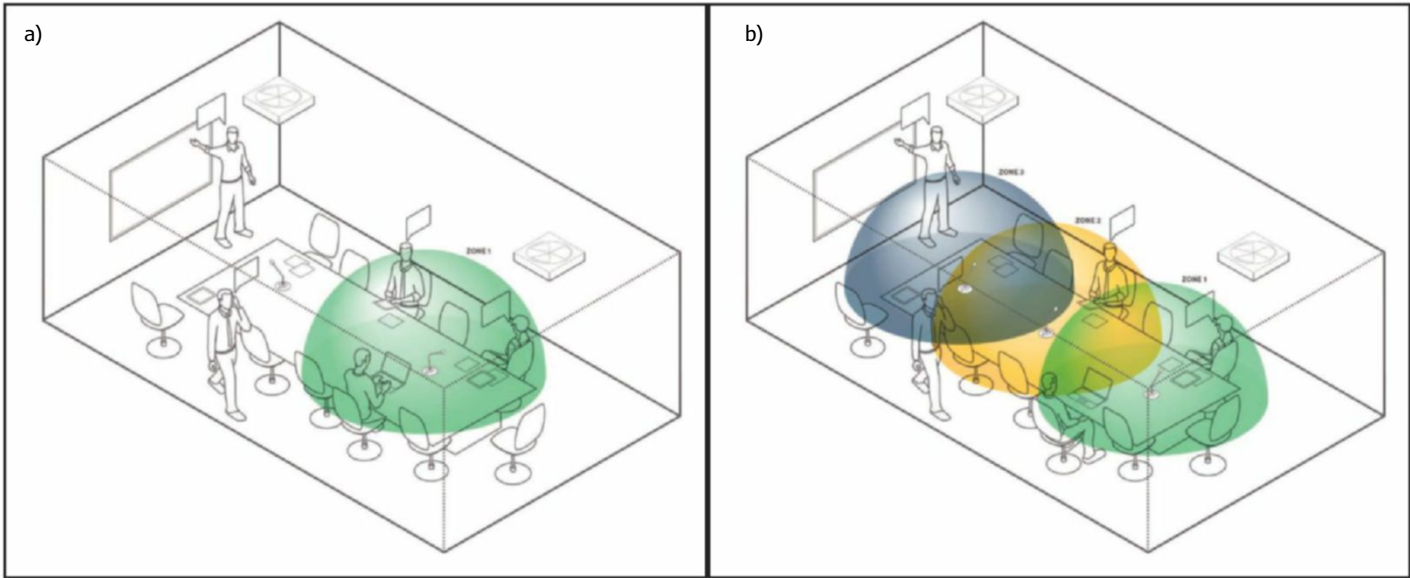


Figure 7: Discrete mic spatial resolution examples—(a) a single mic has one individual spatial zone and a total of one spatial zone in the space; (b) with multiple discrete microphones, three spatial zones are shown for a total spatial resolution value of 3.

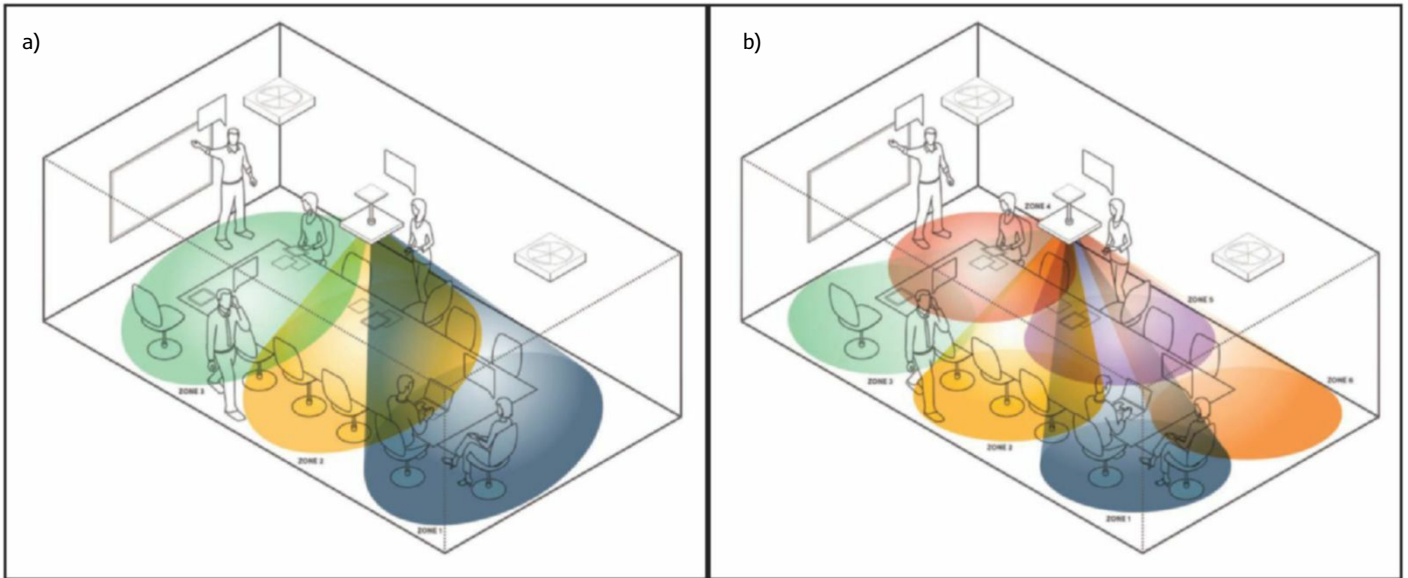


Figure 8: Beamformer spatial resolution examples—(a) a three-zone system that results in a spatial resolution value of 3; (b) six zones with a spatial resolution of 6.

microphone system can identify sound sources and their properties.

So, do we want a blurred acoustic picture with lower spatial resolution? Or can we enjoy the technology benefits of high-resolution information and data in an acoustic space?

Why Thousands of Virtual Microphones Matter

Dividing a space into smaller areas results in more precise focus. It's the only way to really understand what's going on in a space. The ideal scenario is to create thousands of evenly spaced locations—which is exactly what Microphone Mist technology does.

Figure 9 shows a system with thousands of individual virtual microphone locations and, in the case of Microphone Mist technology, results in a total

spatial resolution of 8,192. This is because, as in other approaches, each location is assigned a spatial resolution value. Microphone Mist technology creates thousands of concurrent virtual microphone zones. When you think about the benefits of resolution in colors and images, it is clear that greater resolution is critical for high precision and detailed collection of sound information. In the acoustic realm, Microphone Mist technology can divide a room into very finely spaced acoustic locations.

By dividing the acoustic space into a much higher resolution spatial 3D grid, our system can monitor and analyze each location based on its own characteristics. This means that at each virtual microphone location, our system measures, analyzes, targets, processes, and reports about that location to develop a comprehensive soundscape presentation of the entire space.

Because the system is continuously gathering acoustic information about each location, it can resolve individual sound sources in relative three-dimensional space within the whole coverage area. It's not limited to optimizing a single sound source and ignoring what else is going on in the rest of the room. When a person is speaking and typing, Microphone Mist technology has the spatial resolution to focus on the location close to the talker's mouth and deemphasize the location of the keyboard. The result is that the system can discriminate between the locations of desirable and undesirable sounds.

Talkers at the boundary of the room (e.g., those standing at a display device) do not have to worry if they are in the coverage zone or not. If talkers stand to walk, they can continue talking without considering how the microphone system is

About the Author

Erica Yorga is the Principal Intellectual Property manager at Nureva, a Canadian technology company. Her career spans a variety of leadership roles researching, developing, and commercializing a broad spectrum of technologies in markets such as audio/visual, oil and gas, telecommunications, military defense, and Government regulatory. The challenge of solving difficult problems by constantly learning and enabling teams through the application of unique approaches to technology and process is a passion she brings to every endeavor she undertakes.

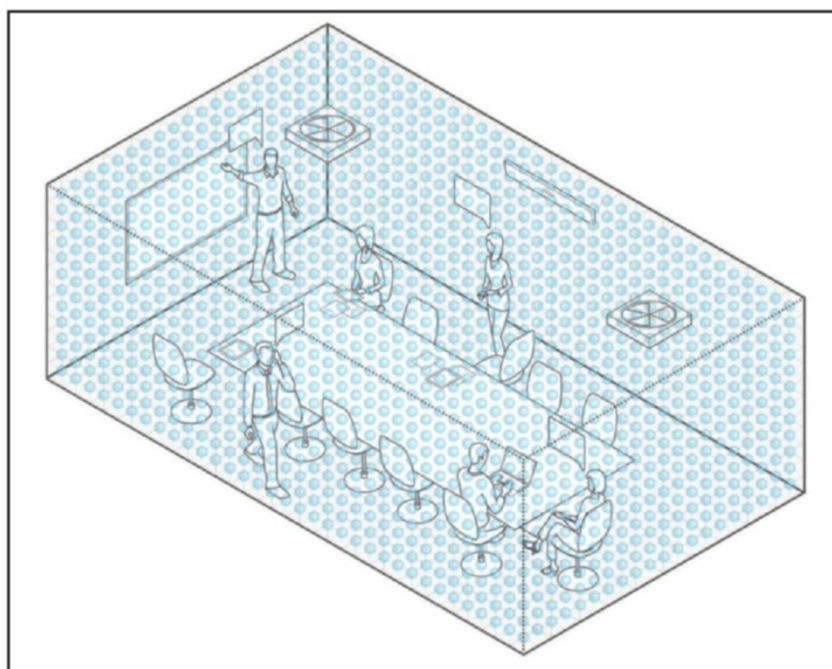
About Nureva

Nureva Inc. is a technology-rich, multiple award-winning private company that imagines and builds audio conferencing solutions that solve the frustrating and persistent problem of poor audio performance in meeting and learning spaces. At the core of every system is the company's patented Microphone Mist technology, which places thousands of virtual microphones throughout a space to pick up sound from anywhere in the room and deliver clear, reliable audio to remote participants. A passion for achieving simplicity through deep user understanding drives the company's product roadmap and the value it creates for its customers. For more information, visit Nureva's website and follow @NurevaInc.

configured. As talkers walk, they are transitioned to each individual virtual microphone in real-time through small seamless transitions because the room is covered with evenly distributed virtual microphones. This ensures consistent high-quality microphone pickup performance in the room regardless of where talkers walk in the room. Talkers can sit, bend, and in effect move naturally in the room without fear of dropping or fading out, which means natural speech, gestures and movements.

When it comes to other sources of unwanted noise such as HVAC, the system can treat them uniquely with our intelligent sound targeting technology. This technology targets and focuses on the specific location of a sound source in the room, even in complex acoustic environments with multiple sound sources. Each active sound source is referenced to a location in a three-dimensional space, and proprietary logic-based processing decisions can be made on how and when to focus the system on a new sound source.

The higher spatial resolution of our system results in more precise information available to make better decisions for optimum acoustic microphone pickup and postprocessing data analysis.



Ready for the Future

So yes, thousands of virtual microphones do matter—they change how we understand acoustic spaces. In uncertain times, where physical distancing and other COVID-19 requirements demand more of our spaces than before, that's more important than ever. 

Figure 9: High spatial resolution coverage using thousands of virtual microphone locations

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Design Considerations Imbedding a High-Output Buffer Inside an Op-Amp Loop

By
Michael Steffes

(United States)

In this Audio Scope article, Michael Steffes provides specific suggestions about how to successfully implement increasing the output current for an audio op-amp by imbedding a unity gain buffer inside the loop.

A common approach to increasing the output current for an audio op-amp is to imbed a unity gain buffer inside the loop where that buffer offers a much higher output current than the core op-amp. Here, both general considerations and specific suggestions for successful implementation will be shown using a recently updated BUF634A buffer applied as shown in **Figure 1** (Reference 1, Figures 9-7).

Open-loop buffers like these offer relatively poor linearity specifications on their own. However, their open-loop nonlinearity driving a heavy audio load will be corrected by the loop gain of the overall op-amp loop. This possible benefit comes with some important caveats and hazards in implementation.

Those include:

- The buffer bandwidth and slew rate should be much greater than the core amplifier for stable operation. The BUF634A offers a supply current adjustment to lower quiescent current. That feature should probably not be used, as reducing supply current drops both bandwidth and slew rate, impairing loop dynamics—and is a feature not supported in the public BUF634A simulation model. The older BUF634 model does support this feature and shows a rapid loss of loop gain (LG) phase margin if used in the design of Figure 1.
- Ideally, the available output voltage swing in the buffer stage should meet or exceed the available output swing from the op-amp output. The circuit of Figure 1 uses a rail-to-rail output OPA2810 on the same supplies with the BUF634A having 1.8V to 2.5V output headroom to each supply depending on output current. If the OPA2810 outputs drive beyond the buffer swing limits, the loop will open up and might not gracefully recover. Using higher supplies on the buffer, or more closely matching output swing range between the op-amp and the buffer, might be prudent. Controlling the input V_{pp} to keep the buffer stage from limiting would also work.

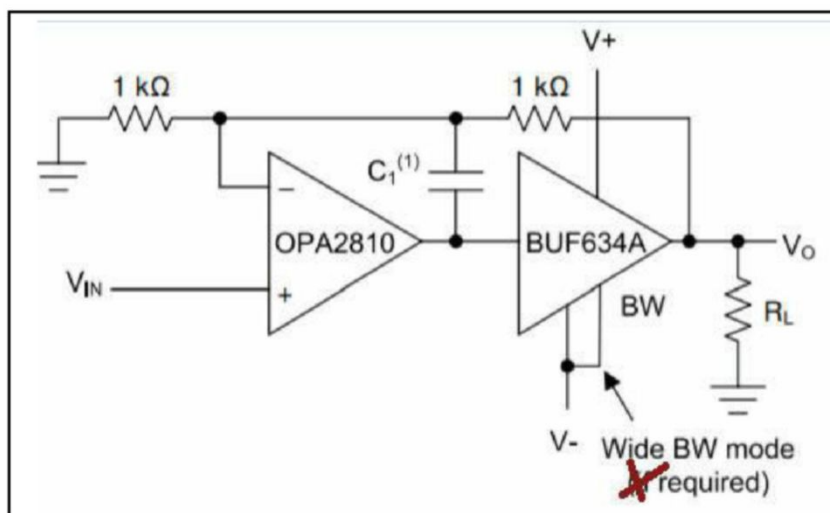


Figure 1: Typical imbedded current boost buffer design using the JFET input OPA2810 and BUF634A, providing a gain of +2V/V

One way to think about adding this buffer is to imagine you were the IC designer on the core op-amp and management asks for 10 times more output current at the 11th hour. Dropping

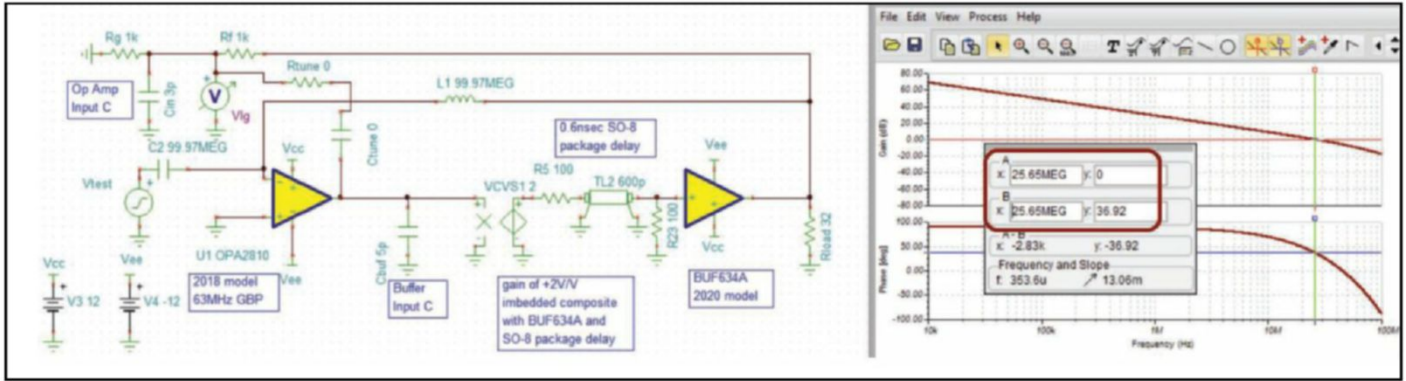


Figure 2: Loop gain (LG) simulation for the composite circuit of Figure 1

in the requisite larger output devices (along with their much higher parasitic capacitive elements) immediately shows more open-loop response phase shift and reduced phase margin (as with any imbedded power stage design like Figure 1). That normally leads to compensating the dominant pole back to a lower gain bandwidth product. There are features in this higher output current stage that usually don't matter since error terms like noise and DC offsets will be corrected by the main op-amp loop.

Phase Margin Analysis for the Imbedded Buffer Composite Amplifier

If the imbedded power stage (maybe a unity gain buffer like Figure 1, or it could be an amplifier stage as well) adds only a small phase shift at the unity gain crossover of the overall op-amp loop, a stable solution is likely. It helps in Figure 1 that the OPA2810 is set up for a noise gain of 2V/V pulling its LG 0dB crossover (xover) back in frequency. Where that is not the case, and the phase shift (or delay) in the power stage adds significant phase shift at xover, a more detailed analysis is merited. A low phase margin design will be producing a peaked closed-loop response and may slip over into oscillation across different loads, temperature, and production variation. Often, this happens well above the intended audio frequency of operation, but it

is still important to consider prior to production.

Focusing on the "delay" introduced inside the feedback loop by this buffer, it is important to implement a very tight layout and more generally consider the propagation delay added by the package. Public simulation models are for the die and do not normally include packaging issues. Bench measurements for the typical SO-8 packaging used for the BUF634A show approximately 0.6nsec package delay. That can be included in a LG simulation as shown in **Figure 2**, using a transmission line model prior to the buffer input.

This is implemented using a dependent voltage source at gain of 2 to recover the matching loss (100Ω Zo used here) in the transmission line model. The buffer input capacitance needs to be placed at the input of this delay block to include any interaction between that and the open-loop output impedance of the core op-amp. Leaving the feedback capacitor inside the loop at zero for now, Figure 2 shows an acceptable 37° LG phase margin for the circuit of Figure 1.

This should produce a closed loop peaking around 26MHz of ≈4.0dB (Reference 2, Figure 2). **Figure 3** shows this closed-loop response with the expected gain of 6.02dB gain at lower (audio) frequencies along with 4.5dB peaking at 32Mhz. While this peaking would be inconsequential in audio applications, it would add ringing to a broadband

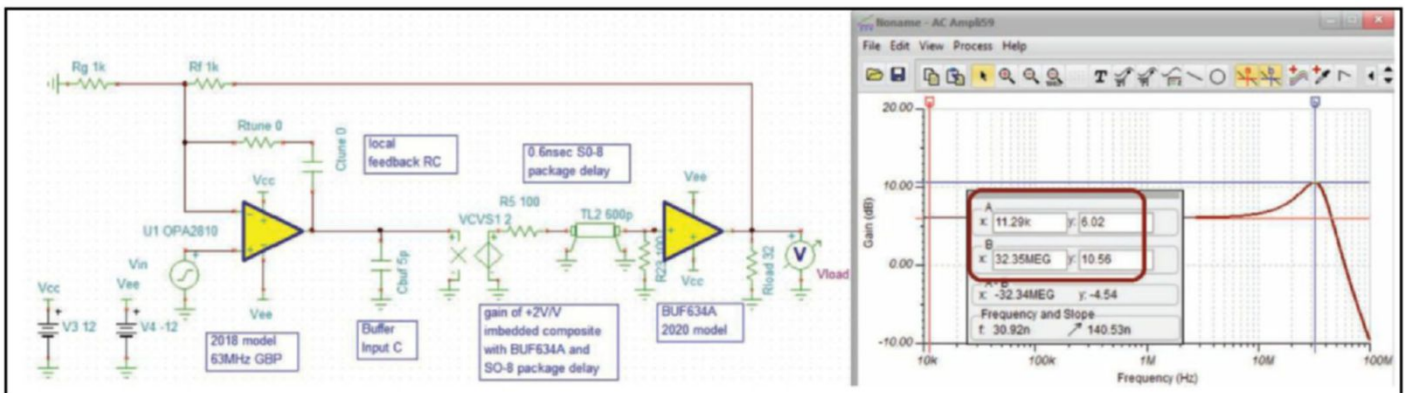


Figure 3: Closed-loop response peaking due to the 37° phase margin for this composite design

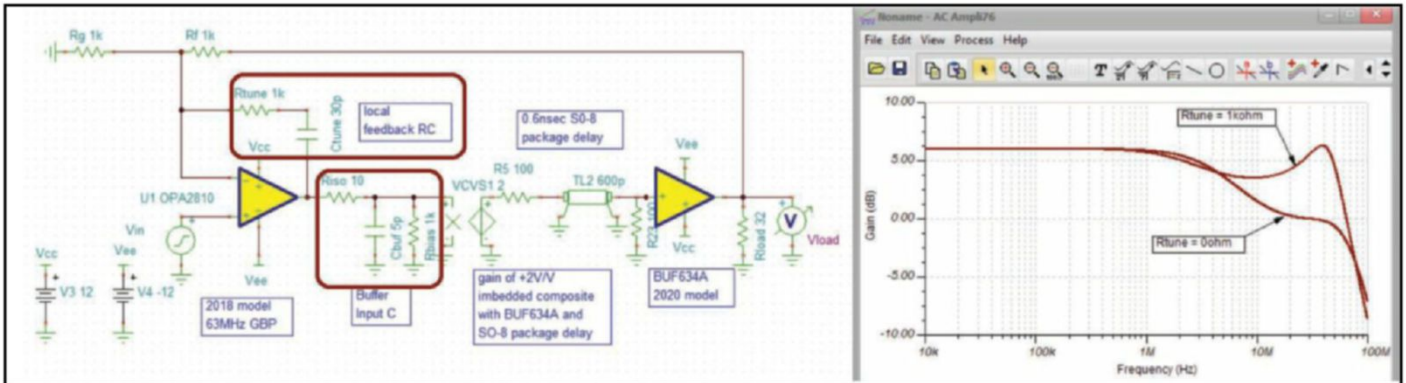


Figure 4. Composite amplifier with local RC feedback around the main op-amp

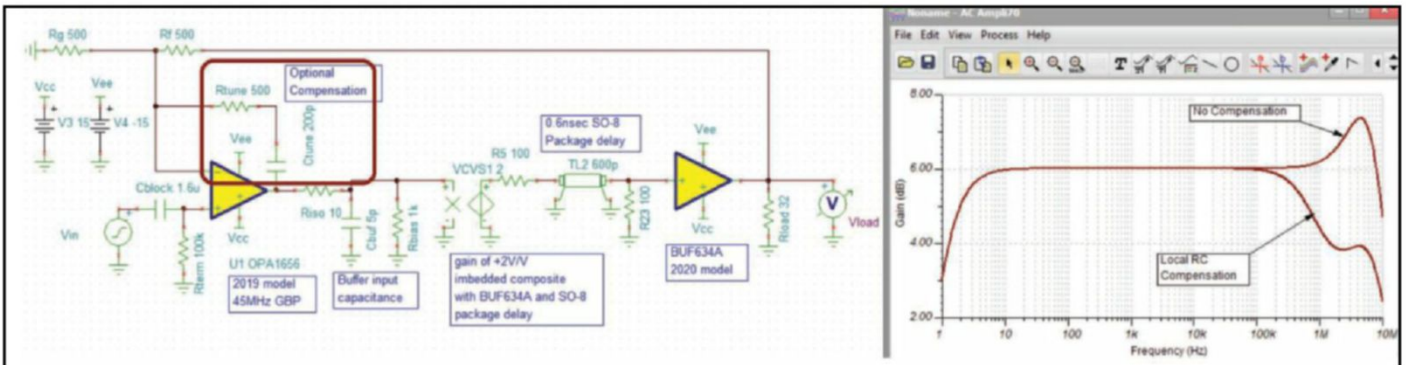


Figure 5. Alternate composite using the OPA1656 JFET input op-amp

step response. The 32Ω load is relatively benign for phase margin—adding capacitive load can degrade this quickly.

Normally, this 4.5dB peaking with 37° phase margin would be an acceptable response. In other designs with lower phase margin, some form of compensation might be needed. Figure 1 shows a simple feedback capacitor, which can be effective if the main amplifier is unity gain stable. This small capacitor adds a low-pass pole with the R_F value and acts to bypass the power stage at higher frequencies closing the op-amp loop as a unity noise gain feedback. More generally, adding a series R_{TUNE} option in that feedback is preferred to shape the noise gain (NG) to a higher resistive ratio value at higher frequencies in case decompensated op-amps are being applied. Setting that compensation R_{TUNE}

value equal to the feedback resistor, and targeting a zero frequency at 1/5 the original xover frequency (26Mhz here, target 5.2MHz, or $C_{comp} = 30pF$) gives the closed-loop response of **Figure 4** where an $R_{TUNE} = 0\Omega$ is also shown with just its low-pass pole. This ripple in the response is normal for this approach, but adding the RC compensation elements has also increased the phase margin to 43°. Setting $R_{TUNE} = 0\Omega$ would be acceptable here as the OPA2810 is unity gain stable.

Two additional resistors have been added to this simulation. A series output 10Ω R_{ISO} from the op-amp output to the buffer input (remember, the transmission line block is just for simulation—you will get that in SO-8 package delay and do not need to add it extrinsically). This R_{ISO} acts to isolate the small buffer input capacitance from any local loops

References

[1] “36V, 210MHz, 250mA Output, High Speed Buffer”, Texas Instruments Datasheet, revised September 2020, <https://www.ti.com/lit/ds/symlink/buf634a.pdf>

[2] M. Steffes, “Stability Issues for High Speed Amplifiers: Introductory Background and Improved Analysis,” Planet Analog, February 3, 2019,

<https://www.planetanalog.com/stability-issues-for-high-speed-amplifiers-introductory-background-and-improved-analysis-insight-5>

[3] “Ultra Low Noise, Low Distortion, FET Input, BurrBrown™, Audio Operational Amplifier,” Texas Instruments Datasheet, revised July 2019, <https://www.ti.com/lit/ds/symlink/opa1656.pdf>

in the op-amp output stage. Also, a $1\text{k}\Omega$ R_{BIAS} to ground provides a DC reference for the buffer input as the composite is powered up. This helps ensure a good DC bias point for the buffer as the op-amp loop closes on power up and a defined load for the op-amp output. It is also helpful to sequence the power to the buffer stage prior to the op-amp stage on power up.

Alternate Implementations Using Newer JFET Input Audio Op-Amps

The example power boost of Figure 1 operates as a non-inverting op-amp using a JFET input op-amp. This allows an AC coupled input to operate with a relatively higher terminating resistor allowing a lower blocking capacitor value. Using the concepts shown here, a more recent imbedded BUF634A composite can be adapted as shown in **Figure 5** (Reference 3, adjusting Figure 51) The OPA1656 offers phase lead in its Aol response giving a very comfortable 55° phase margin in the circuit of Figure 5 without the optional compensation. It is best to include places for the compensation RC in case the physical implementation needs them. Using the same approach as above (and placing the

compensation zero at $1/5$ of 8Mhz LG crossover) increases the phase margin to 62° and gives the broadband response shape of Figure 5. It is likely no compensation would be required for this implementation with the BUF634A operating in full power mode with a very tight layout. Again, if the load includes a capacitive element, this could quickly change and the optional compensation elements might prove very useful.

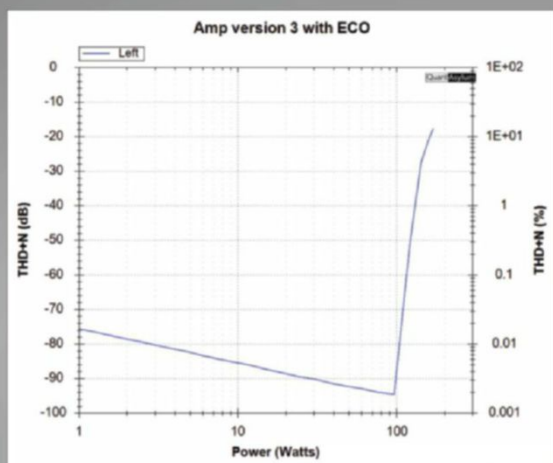
Where the application can drive the relatively low input impedance of an inverting op-amp implementation, a different compensation approach offering some useful benefits can be applied. We will explore that approach next month. 

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 more than 150 contributed articles and application notes since 1988.



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Widex Moment mRIC RD Hearing Aid

By
Brent Butterworth

Can a hearing aid use DSP to deliver superior performance plus the convenience and versatility of the latest true wireless Bluetooth earphones? Brent Butterworth tests the Widex Moment to find out.



Manufacturers are packing an incredible array of features into the latest true wireless stereo (TWS) earphones, including adaptive active noise cancelling, hear-through modes, biometric sensors, Amazon Alexa and Google Assistant access, multiple EQ tuning modes, and more. The millions of people who wear hearing aids must be a little envious, especially considering hearing aids fitted by an audiologist typically cost 10 or 20 times as much as name-brand TWS earphones.

Hearing aid companies have started to respond by turning their products into something more akin to TWS earphones. They're offering Bluetooth connections for music and podcast listening, multiple listening modes optimized for different environments, and even directional pickup modes that focus on sound coming from a specific direction.

One of the newest entries in the high-tech hearing aid category is the Widex Moment, which is available in three in-ear and three behind-the-ear designs. In addition to the features already described, the Moment uses a "Zero Delay Accelerator," accessed through its PureSound mode, which Widex says reduces the latency of the hearing aid to 0.5ms. This compares with around 2ms in the Moment's Universal mode.

According to Widex, this mode makes the Moment sound much more natural than other

hearing aids. The reason, I assume, is that it reduces comb-filter effects—cancellation and reinforcement patterns at different frequencies caused by the phase differences between environmental sounds that pass straight to the ear and those processed by the hearing aid. The company claims not only that the Moment "restores one's sense of perfectly natural hearing," but also that wearers streaming through Bluetooth will find the Moment to be "the best headphones they've ever used."

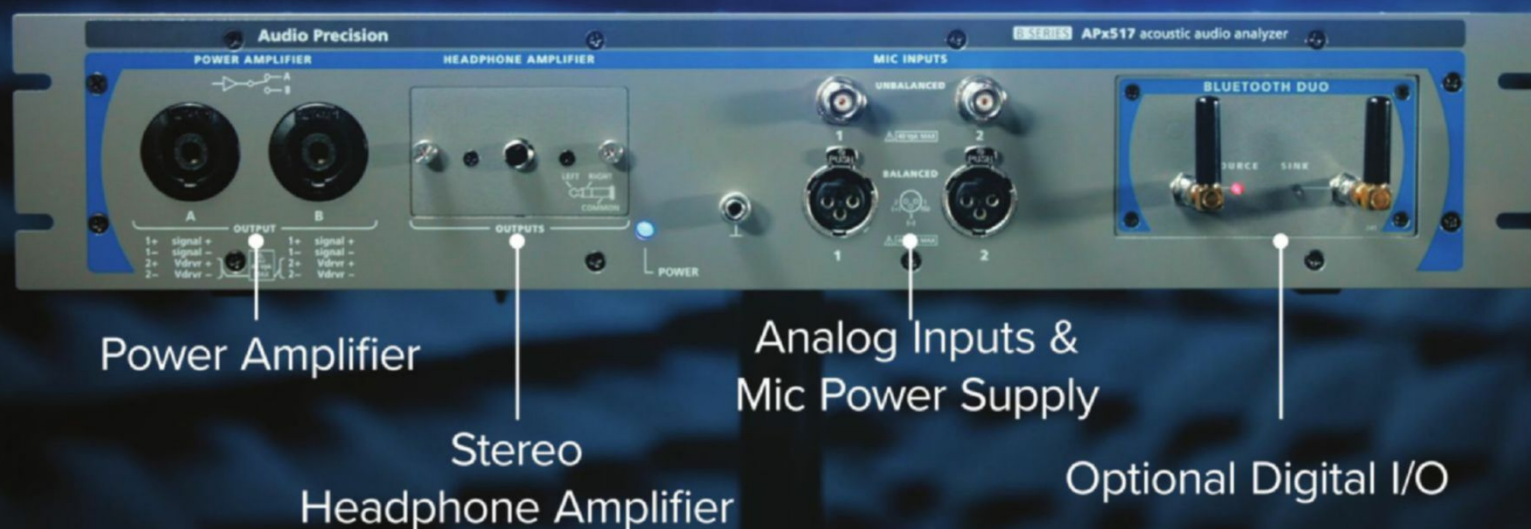
The Product

Each of the Moment mRIC RD hearing aids consists of a tiny, 1" long (25mm) electronics module, connected via a thin, plastic-covered wire to a single balanced armature driver. (Most behind-the-ear hearing aids have the driver inside the electronics module, with a tube carrying the audio from the driver into the ear.) An assortment of open and closed silicone ear tips is provided.

The Moment uses a rechargeable battery rated for 16 hours of runtime with a full charge, and four hours' runtime with a 30-minute charge. The charging module has slots into which each hearing aid drops. It connects with a USB-C to USB-A cable to a provided wall-wart charger. Removing the hearing aids from the charging module automatically powers them up.

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Almost all TWS earphones have a charging case that can recharge the earphones three to five times before the case itself must be recharged, but the Moment doesn't. Instead, it has a case with air-filled rubber pads to hold the hearing aids in place during transit, and vents that allow them to dry out. So if the wearer expects to be gone long enough to exhaust the Moment's battery, they'll have to bring the charging module, the charger and its cable, and the ventilated travel case.

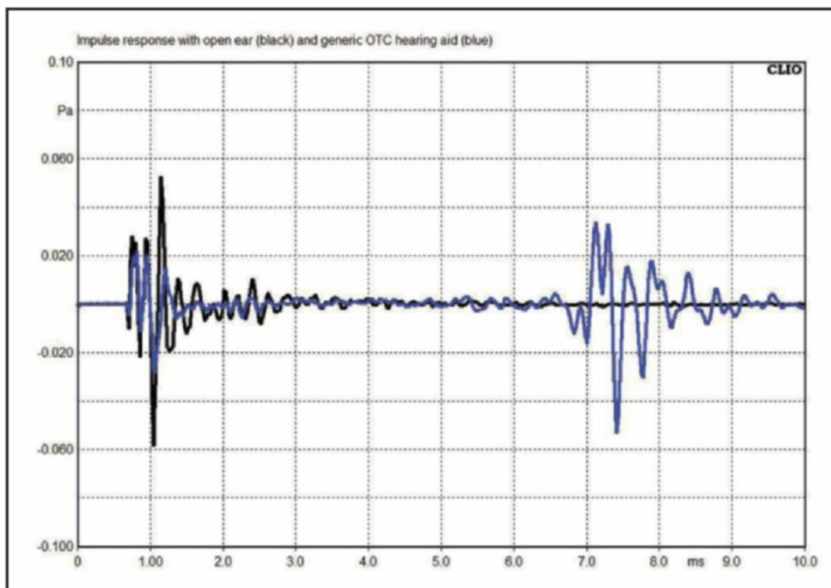


Figure 1: Impulse response showing latency of a generic over-the-counter hearing aid. The pulse furthest to the left is the original sound entering the ear simulator; the pulse to the right is the delayed sound that has been processed and amplified by the hearing aid.

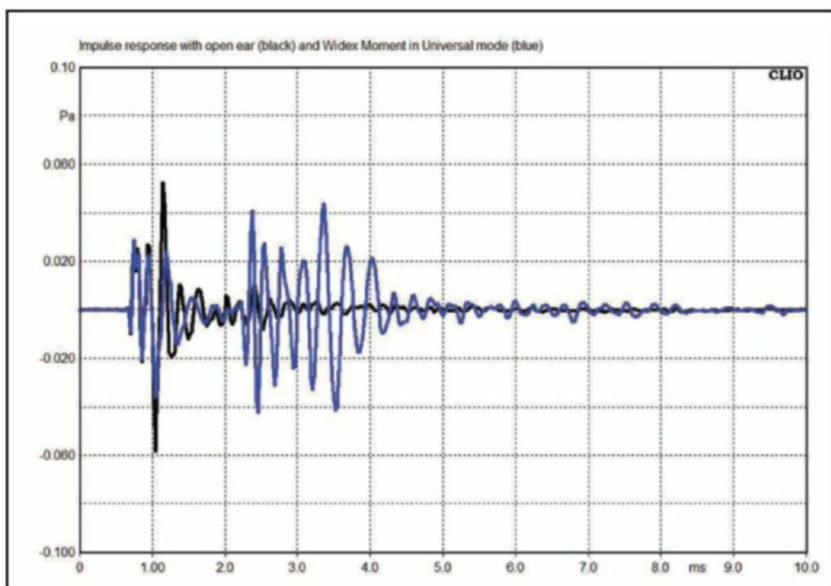


Figure 2: Impulse response showing latency with Widex Moment hearing aid in Universal mode

Each hearing aid has a single button on the back of the electronics module. A long press of this button turns them on and off. A brief touch can flip through the different listening modes, and also control volume if you're using two Moments.

You can use the Moment not just as a hearing aid, but also as a set of Bluetooth earphones, so you can listen to music and podcasts and take phone calls through the hearing aids, just as you would with TWS earphones. However, at present the Moment can stream only from iOS devices. According to Widex, the Moment is "... also prepared for future connections with select Android devices." There is a Widex Moment app for Androids, though, which allows the wearer to control the hearing aids, even if it doesn't allow the wearer to stream content through Bluetooth.

The Moment offers as many as 11 different basic listening modes, accessed through the app or through the button on the back of the electronics module. These modes are added by the audiologist as needed. My sample included the Universal, PureSound, and Music modes. Other options include modes optimized for quiet and loud settings, parties, music listening, transport, phone conversations, and more. The audiologist can add a Zen mode, which provides relaxing sounds, and there's also a SoundSense Learn mode, which uses AI to optimize the hearing aid's response for the user's daily routine.

Widex does not publish pricing on hearing aids, because its products are sold only through audiologists, and as part of a package that includes initial testing and fitting as well as return visits for fine-tuning. You'll have to ask your audiologist about the pricing, but as with other audiologist-provided hearing aids, expect to pay at least \$2,000 per ear for the product and the audiologist's services. Widex offers its behind-the-ear models in 13 colors (including Sporty Red and Shocking Pink!), and also offers optional interfaces that let you use the Moment to listen to TV, FM radio, and so forth.

The Fitting

Although I have only the mild hearing loss typical of a 58-year-old male (i.e., an easily detectable amount of loss, but not enough to benefit from hearing aids), I have recently been evaluating and measuring various over-the-counter personal sound amplification products (PSAPs), so I'm familiar with how hearing-assistance products work and was eager to see how a real audiologist-tuned hearing aid stacks up against the PSAPs I've tested.

Widex provided me with a sample of the mRIC RD, a behind-the-ear version of the Moment, along with access to an audiologist to set it up. Due to the

COVID-19 pandemic, we conducted this procedure remotely, using the Widex Remote Care Link. The Remote Care Link is a Bluetooth-connected device that hangs around the patient's neck, and connects through the Widex Remote Care smartphone app to allow the audiologist to conduct a hearing test that's much like what you'd experience in the audiologist's office. The only exceptions are the test tones come from the hearing aids themselves rather than headphones, and you don't take the test in a soundproof chamber as you would with an audiologist. A video conferencing link within the app lets the audiologist talk the patient through the process. The audiologist can then remotely program the hearing aids to best suit the patient's hearing deficiencies.

I'd had a professional hearing test about a year before that showed only mild loss typical for my age and gender, so I wasn't surprised when the correction curve the audiologist put in was too subtle for me to detect or even reliably measure. Later, a Widex technical representative used the same remote setup process while one of the Moment hearing aids sat in my GRAS Model 43AG ear/cheek simulator. She was able to program the Moment for a more typical hearing-loss profile that would make my measurements and evaluation easier.

Measurements

I performed the measurements using GRAS Sound & Vibration test equipment: the Model 43AG, an RA0402 high-resolution ear simulator/coupler, and a KB5000 anthropomorphic pinna. I also used an Audiomatica Clio 10 FW analyzer, as well as an M-Audio Mobile Pre USB interface with TrueAudio TrueRTA real-time analyzer software.

The first measurement I wanted to try was to confirm Widex's claims about processing latency—something I've honestly never thought about measuring in PSAPs, because most seal in the ear canal and block the majority of external sound above about 1kHz. I measured the latency by placing a Genelec HT205 powered monitor a few inches from the ear/cheek simulator, then playing a logarithmic chirp tone to measure the impulse response with no hearing aid, with the Moment in Universal and PureSound modes, and with a generic over-the-counter (OTC) hearing aid I had sitting around.

In each of the charts presented, the black trace represents the impulse response with no hearing aid, and the blue trace shows the impulse response with the hearing aid. Measuring the difference between the original pulse and the second pulse created by the hearing aid gives us the latency.

Figure 1 shows the latency with the generic

OTC hearing aid, which measures 6.34ms. **Figure 2** shows the latency with the Moment in Universal mode: 1.60ms, much less than the generic hearing aid. **Figure 3** shows the latency with the Moment in PureSound mode: 0.48ms. Clearly, Widex's latency claims are accurate.

Converting the Moment's impulse responses to 1/12th-octave-smoothed frequency response measurements was illuminating. You can see in **Figure 4** how the PureSound mode seems to

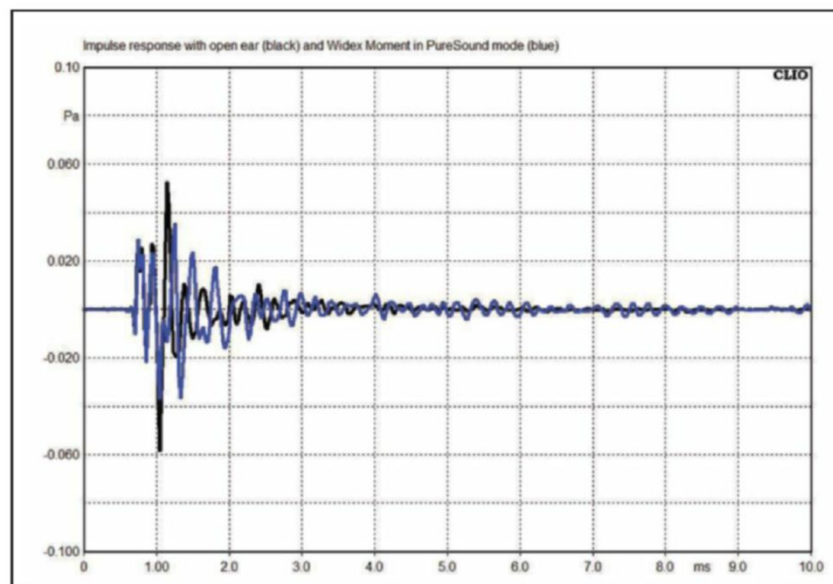


Figure 3: Impulse response showing latency with Widex Moment hearing aid in PureSound mode

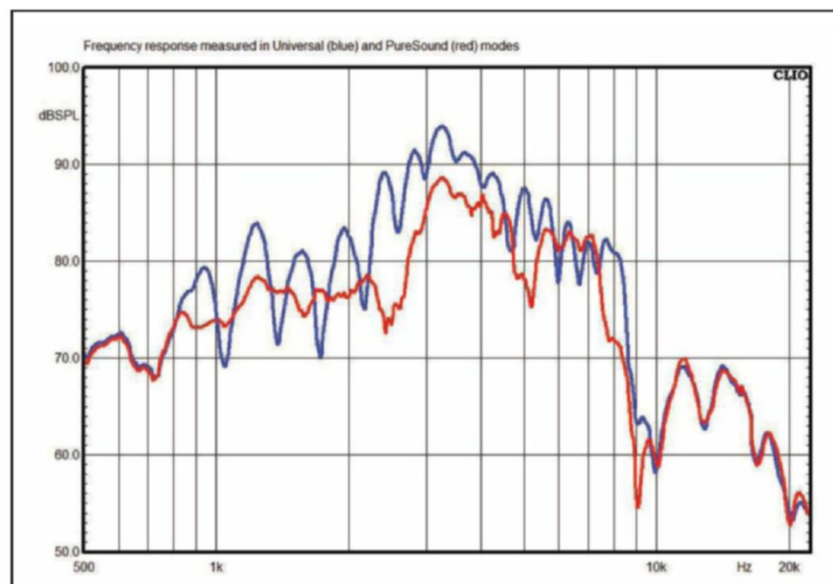


Figure 4: This graph shows the 1/12-octave-smoothed frequency response of Widex Moment in Universal (blue trace) and PureSound (red trace) modes, measured with logarithmic chirp tone. Comb filtering is easily seen in Universal mode but almost absent from the PureSound mode.

completely eliminate the comb filter effects that are easy to see in the Universal mode.

Next, I wanted to see the effects of some of the Moment's different listening modes. To perform this measurement, I used an array of six Dayton Audio B652 speakers mounted around my lab plus a

Sunfire True Subwoofer Jr., all playing uncorrelated pink noise at 60dBC. (This is the same setup I use to test active noise cancelling in headphones.) I connected the output of the ear/cheek simulator to the M-Audio USB interface and viewed the result in TrueRTA.

In the resulting chart, **Figure 5**, the 60dBC noise baseline (corrected for the response of the speakers, room and ear/cheek simulator) is shown in red, and the response curves of the hearing aid in the different modes show how much the hearing aid boosts or cuts the sound at different frequencies. (Keep in mind that this sample was tuned for the hearing loss of a typical hearing aid owner; a typical correction might be +10dB to +20dB between about 800Hz and 5kHz.) Note how similar the profiles of the Universal and PureSound modes are, and how the Music mode is more or less at parity with the others around 2kHz, but adds considerable boost above and below that frequency.

I tried several methods of measuring the efficacy of the Moment's directional pickup modes, finally settling on using a couple of Audioengine A1 speakers, one placed 1.5 meters to the front side of the ear/cheek simulator, the other placed toward the rear, so I could produce pink noise from each direction. I first ran response curves without the hearing aid, so I could subtract the ear/cheek simulator's transfer function and see only the response of the hearing aid itself.

Figure 6 shows the effects (smoothed to 1/6 octave) of the Universal mode, the Directional mode set to "front," and the Directional mode set to "rear." The upper set of traces shows the hearing aid's effects on sounds coming from the front, while the lower set shows the effect on sounds coming from the rear. The Directional mode set to "front" did boost sound from the front by 1dB to 3dB between 2.5kHz and 6.5kHz, while the same mode set to "rear" had a similar effect on sounds from the rear. (Incidentally, I kept trying to find a way to measure this because I could clearly hear a directional bias of about 3dB when listening to two smart speakers playing different Internet radio programs from in front of and behind me.)

Figure 7 shows the frequency response of the Moment (as configured by Widex for my tests) when used for audio playback from a Bluetooth-connected source, in Music and Speech EQ modes. For the stimulus, I used pink noise from my iPad, smoothed to 1/12 octave. For comparison, I show the response of the JVC HA-FW01 earphone, a passive model that has a fairly typical response for a dynamic-driver earphone. Because of the tuning of the hearing aid and the open-air tip I used, there's little bass,

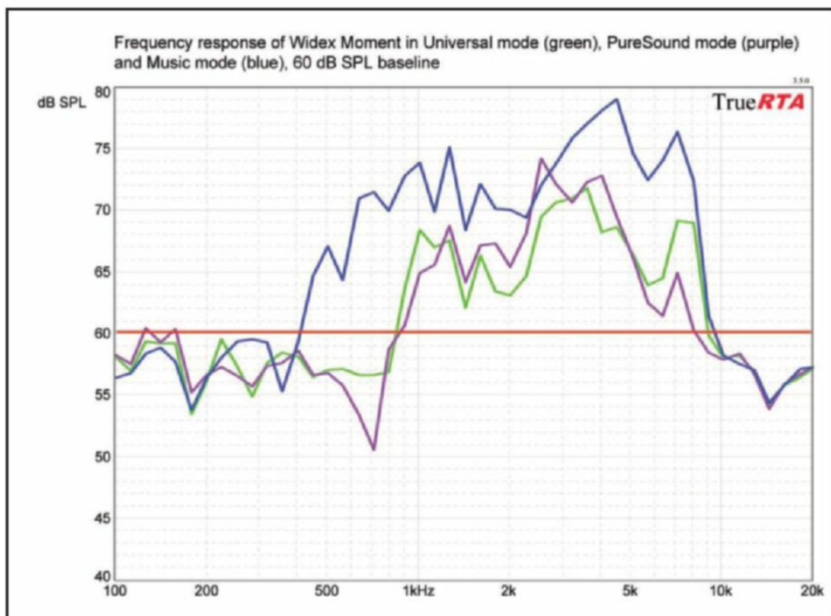


Figure 5: This graph shows the 1/6-octave-smoothed frequency response of Widex Moment in Universal (green), PureSound (purple) and Music (blue) modes, measured with 60dBC SPL pink noise.

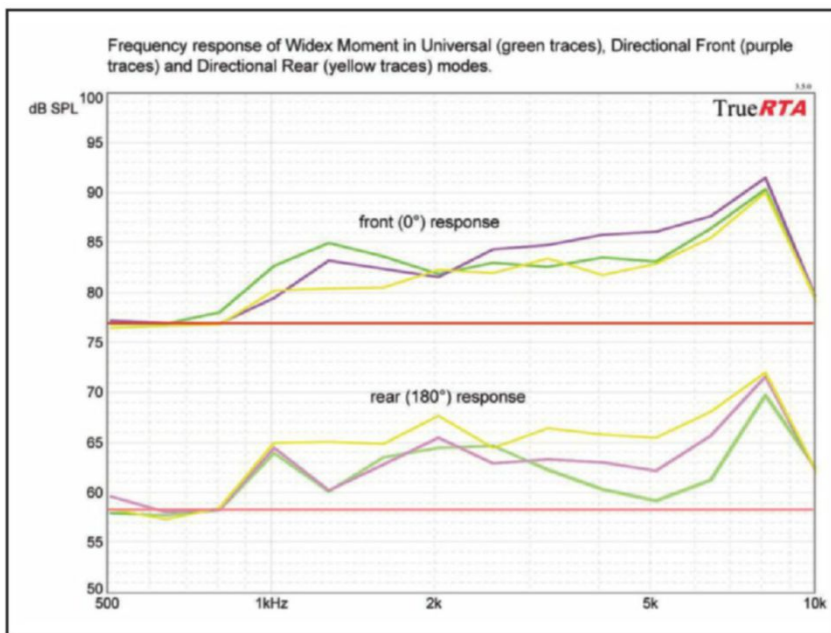


Figure 6: Directional effects of Widex Moment in Universal, Directional Front, and Directional Rear modes, 1/6-octave-smoothed, measured with 77dBC SPL pink noise. Upper set of traces shows the effects on sound coming from 0° (straight ahead), lower traces show shows the effects on sound coming from 180° (directly behind).

and it's obvious that the design/tuning intent of the hearing aid is much different from that of the earphones. I'll have more to say on this in my subjective notes.

Subjective Testing

Although due to the pandemic I'm not traveling or even interacting with people face-to-face all that often, I was able to use a pair of Moments while conducting typical errands (e.g., running to the grocery store and grabbing take-out from my favorite diner). For someone who's used hundreds of earphones, almost all of which include silicone tips that completely seal the ear canal, it was initially off-putting to have a couple of spindly balanced armature drivers on wires inserted into my ear canals, but I got used to it after a while and often forgot I was wearing them (**Photo 1**). One big, but hopefully temporary, downside of a behind-the-ear hearing aid such as the Moment is that it tangles easily in the ear loops of a face mask.

Feedback has traditionally been the bugbear of hearing aids, as the wearer's ears are momentarily tortured by an eardrum-splitting, high-frequency whine while donning and doffing the hearing aids.

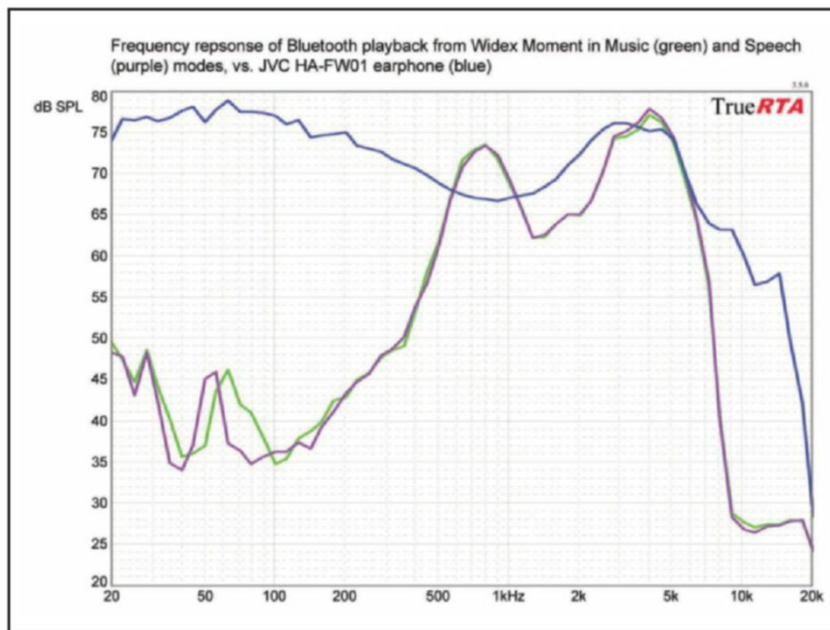


Figure 7: Frequency response of Widex Moment, configured for typical moderate hearing loss, when used for audio playback from a Bluetooth-connected source, in Music (green trace) and Speech (purple trace) EQ modes, using pink-noise stimulus and smoothed to 1/12 octave, compared with a JVC HA-FW01 wired earphone (blue trace).

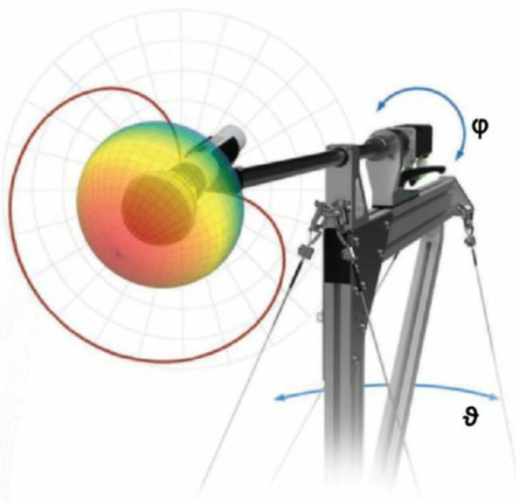
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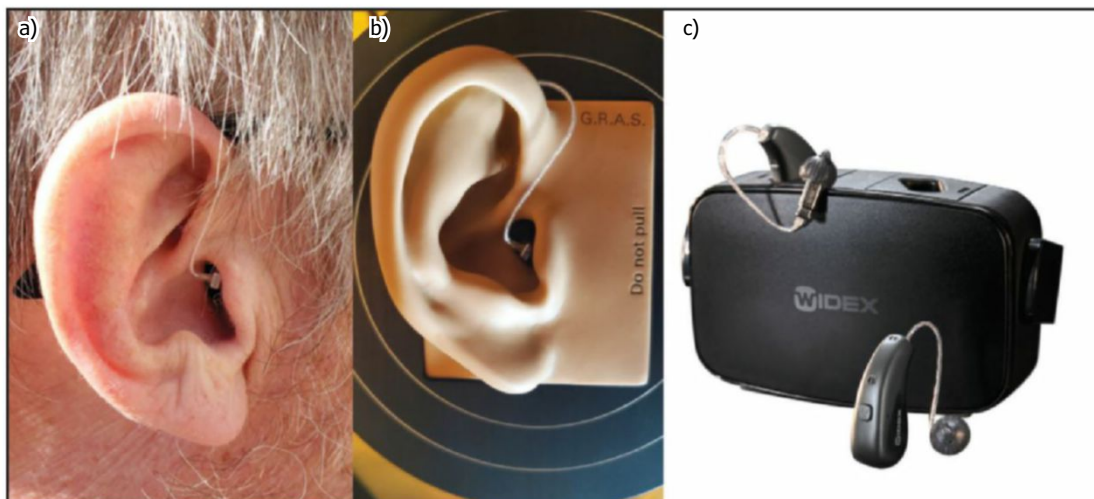


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Photo 1: The reviewer's real ear with the Widex Moment in place (a); the device-under-test in a GRAS KB5000 anthropomorphic pinna equipped with the RA0402 high-resolution ear simulator/coupler (b); and the Widex hearing aids with the charging module (c).



DSP has made feedback much less of a problem, sometimes by using compression to clamp down on problem frequencies, and I'm happy to say the Moments only squealed twice, and both times only fleetingly: once when I was putting on a face mask to go into a store, and once when I was putting one into the ear/cheek simulator.

Even with the "typical" amount of boost Widex had programmed into the units, I had to crank the level up to or near maximum (it's easily adjustable via the app) to clearly hear the differences among the modes. I focused on Universal, PureSound, and Music. All of the modes did a terrific job of boosting voices so they were easier to understand, and I was surprised that the sound didn't seem particularly unnatural; it was clear I wasn't experiencing my normal hearing, but I was able to get accustomed to the hearing aids effects after a few minutes of wearing them.

I wasn't sure what difference I was supposed to hear between the Universal and the PureSound modes, but I did hear a difference. With the Moments' level of boost set to maximum, the Universal mode produced a constant, low-level "whooshing" sound, as if I were sitting right next

to an HVAC vent. With the PureSound mode, I'd estimate that about 80% of this effect disappeared. Meanwhile, the PureSound mode didn't seem any less effective at enhancing my hearing. Surprisingly, I didn't note a major tonal balance shift when switching between the two modes; any shift I heard was fairly mild and not, to my ears, consequential.

Thanks to the COVID-19 pandemic, my opportunities to hear live music are practically nil until probably well into 2021, so to try the Music mode, I played some favorite albums through my high-end audio system, comprising Revel Performa3 F206 speakers and Parasound and Outlaw Audio electronics.

With the Universal and the PureSound modes, the sound of the instruments and the voices was thin; the bandwidth sounded about like AM radio, although the sound quality was otherwise much better. The Music mode seemed to fill in some of the lower midrange, which allowed instruments and voices to sound fuller.

Listening to recorded music through the hearing aids themselves, using a Bluetooth connection from an iPad, was less satisfying, although I'm hesitant to judge because while we all know what it's like to have problems understanding voices (due to poor telephone connections, poor acoustics, etc.), I don't know what it's like to experience recorded music as a hearing-impaired person.

With the standard open-air tip, and no seal in my ear canal, music from Spotify was fairly low in level and had limited bandwidth. It sounded something like a good 1", full-range driver playing at a distance of a meter or so. The midrange was clear and seemed reasonably natural, but there was little or no body to the sound.

Open-air tips are the norm with hearing aids used for mild to moderate hearing loss. The general

About the Author

Brent Butterworth has been a professional audio journalist since 1989, and has measured and evaluated thousands of audio products. His headphone measurements and reviews currently appear on soundstage.com and wirecutter.com. He has served as editor-in-chief of Home Theater and Home Entertainment magazines, contributing technical editor for *Sound & Vision* magazine, senior editor of *Video* magazine, and reviews editor of *Windows Sources* magazine. He has also worked as marketing director for Dolby Laboratories, and as a consultant on the design and tuning of speakers, subwoofers, and headphones.



concept is that hearing loss in low frequencies is rare, so it's best to let the wearer hear low frequencies naturally and boost only the midrange and high frequencies. The downside of this is that it's probably impossible to get full bass from a single balanced armature floating freely in the ear canal. I noticed that Widex's website shows the Moment with larger tips that seal the ear canal.

These tips are typically used for patients with severe hearing damage and/or damage in the low frequencies. I thought that they might sound fuller for music listening, but they didn't to any significant degree, and the sound seemed a little better with the open tips—more like a fairly smooth, haystack-shaped response curve centered at 2kHz or 3kHz, while the closed tip made the midrange sound less clear.

The bottom line is that a conventional hearing aid, with a balanced armature or a long, thin sound tube suspended in the ear canal, probably can't provide the kind of full-range music listening experience that even a good \$50 set of earphones delivers. Even when fitted with silicone tips that seal the ear canal, the hearing aid doesn't seem to have enough mass and bulk to prevent low frequencies from escaping. Of course, those \$50 earphones will likely not deliver a good experience for someone with significant hearing impairment unless a great deal of EQ (more than available through typical smartphone apps) is added at the source.

Music playback is an area where the hearing-aid industry—and the practice of reviewing hearing aids as multiple-use devices—is in its infancy. Whether hearing-aid designers can find a way to deliver the kind of music-listening experience that good earphones provide, and whether someone with healthy hearing can predict how a listener with hearing impairment perceives that experience, remains to be seen. There's a lot of work left for all of us here, but you gotta start somewhere.

Conclusion

Based on my subjective testing and my measurements, the Widex Moment seems quite effective at its basic task of making voices easier to understand, and the more natural, free-of-artifacts sound of the PureSound mode may be enough to get those with hearing loss to buy these.

In my experience, the PureSound mode sounded natural enough that the hearing aids didn't bother me, and that I sometimes forgot I was wearing them. The Music mode also had clearly beneficial effects—something that I, as a frequent denizen of jazz clubs, will likely find appealing when age degrades my hearing enough that I need help. Although I found it cumbersome to control the tiny units with my big fingers, it was easy and convenient to control them through the app.

As far as listening to recorded music over Bluetooth, all I can say is that this function works. To come to a firm conclusion on how effectively it works, I'll need more experience with other hearing aids that offer this feature, and I'll need to persuade a few people with significant hearing impairment to give me their impressions ... or to wait a decade or two until my hearing declines enough to make me a better test case. 



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The Science and Physics of High-End Mic-Preamp Design (Part 1)

Principal Mechanisms for Audio Designers: Engineering the Sound of Silence

By
Mike Meehan



This series of articles investigates the principal mechanisms of noise, distortion, and interference that microphone preamplifier audio designers must understand, before exploring various circuit topologies used to maximize performance and minimize susceptibility. Finally, the ideal solution is considered.

Show a layperson a mixing desk and likely-as-not, they're daunted by the number of knobs, (perhaps hundreds), buttons, indicators, faders, and meters. However, once broken down into its component parts, it is far easier to understand. Convention dictates that on the left-hand side of the desk are the channel strips. Immediately to their right (if applicable) are the sub-masters or groups (a sub-division of the number of inputs—so 4/8/16, typically). Finally, we have the stereo (or more) output section (although this may occupy the central section).

A "channel" strip comprises four main sections—the input section (at the top), the EQ section (in the middle), the routing and monitoring section next, and the channel fader. Ignoring anything else, it is both the input section and the fader (mixing bus) section that determine noise performance (with the proviso that the EQ and other sections will have been designed so as not to compromise heavily-won noise performance gains—no pun intended—in the input section).

As an aside, I was commissioned by the acclaimed classical music producer and former general manager at Abbey Road studios, Allen Stagg (RIP), to design a minimalist very-low-noise low distortion 16-channel desk for direct-to-master

of jazz and classical music. It had nothing but an input gain control, phantom powering, Pre-Fade Listen (PFL) monitoring, a peak indicator, and a fader. And that was it. No bells or whistles. Just extremely low noise and distortion input and mixing bus sections. And that is the approach I intend to take in these articles.

We look first at the physics and mathematics of the design of the input section. The EQ section (which can vary from the classic bass/treble Baxandall arrangement to a sophisticated four-band fully parametric type is ignored, as indeed are the other parts of the strip). Part Two puts the science to work in explaining the practicalities of the design of a mic preamp—looking at various different circuits, their pros-and-cons, and considering the various different tools and topologies available to the designer and the practical performance limitations too.

So, to paraphrase a *Sound of Music* song, let's do a do-re-mi and start at the very beginning because that's a very sound place to start.

Despite the increasing sophistication, fidelity, and resilience of all things digital, the very fact that sound essentially is an analog phenomena, requiring capture, amplification, subsequent conversion, and then reproduction while in this domain, means that excellence in analog design has never been of

greater importance. The first extreme of the audio signal processing chain, (after its initial acquisition by microphone) determines its achievable fidelity—loss of integrity at this point can never be undone.

These articles investigate the principal mechanisms of noise, distortion, and interference that any microphone preamp designer must fully understand. They then explore the various circuit topologies used to maximize performance and minimize susceptibility in respect of these effects, as well as reducing cost. Finally, they present what might be considered the ideal solution—one that will require efforts from both design/manufacturing departments, and those concerned with presenting robust and high-quality one-chip solutions to the audio engineering community.

Since the advent of the CD, and the BBC's Near-Instantaneous Companding and Multiplexing (NICAM) system, the importance and significance of high-quality audio has steadily increased, driven particularly by the development, implementation, and widespread professional and consumer acceptance, and by the falling costs of new technologies to acquire, compress, store, and transmit data, sound, and images in the digital domain. Primarily this is because of increased consumer awareness about sound quality via exposure to, and mass availability/increasing affordability of, digital formats such as CD, etc. and a burgeoning number of derivatives/new formats—the demand is there too as consumers invest in expensive home cinema 7.1 surround sound systems and similar such as high-end Dolby Atmos and DTS-X soundbars. Whatever the reason, this is most welcome.

Similarly, in formats where both audio and video are represented (e.g., cinema and DVD), no longer is audio just tolerated but not properly supported. Virtually year-on-year improvements in digital sampling/quantization—14-bit, 16-bit, then 24-bit, f_{SAMPLING} up to 192kHz, falling costs of the technology—once coupled to the increasing sophistication of coding algorithms in technologies like MPEG-4 structured audio, mean that digital representations of sound indistinguishable from the original are becoming more commonplace.

Despite the digital revolution, however, one aspect of sound that hasn't changed, not surprisingly, is the fact that it resides in an analog domain. It is deliciously ironic—in all of audio-engineering's 143-year history—taking as a reference point Thomas Edison's first reproduction of sound from a foil-covered cylinder at Menlo Park in 1877—that the key analog design skills necessary to build the amplifiers used to capture faithfully sub-millivolt signals residing in what sometimes are

quite hostile environments are more important than ever. Consequently, expertise in the field of high-precision, low-noise analog design has ramifications extending well beyond audio (RF design being a good case in point).

This series looks first to the various mechanisms to be considered when undertaking demanding analog design of this nature and explores various circuit topologies/devices used to reduce noise, distortion, and interference, while maximizing headroom. Finally I consider ways in which miniaturization/microelectromechanical systems (MEMS) could be used to match exactly preamp to microphone capsule, as well as eliminating one of the great hurdles in the quest for perfect sound signal capture.

Introduction

In audio mixing console and outboard audio effects' design, (whether analog or digital), there are principally three areas where design effort is particularly concentrated: low-noise microphone preamplification, equalization, and dynamic range control. Unsurprisingly these areas yield the greatest profits and the highest accolades for successful products—Neve, Massenburg, Orban, Blackmer, and Dolby, among many others, are synonymous with both sonic excellence and commercial success in these fields.

Consider the statistics in perspective. A state-of-the-art audio mixing console represents the very highest 1% of 1% of what in better-known domestic audio circles would be termed the highest-end audiophile (ultra-fi): microphone pre-amp design elevates this performance level to a higher plane by a large chunk.

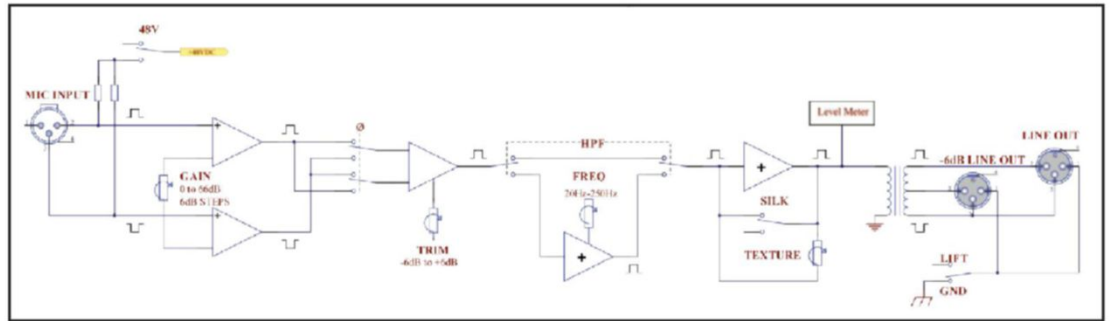
Low-noise low-distortion designs with high common mode rejection (CMR) and radio frequency interference (RFI) rejection extending into the low-AM (and above) region of the electromagnetic spectrum, represent the pinnacle of analog audio design: certain designs command premiums of \$5,000 per channel. The preamp challenge thus provides immediate commercial attraction—a fact reinforced by the seemingly exponential rise in outboard preamplifier products, from a handful of years back, over the decades, to more than perhaps 300 nowadays.

Microphone Basics

First, we must understand the instrument of sound capture and some of the important aspects of the sound to be captured.

Mics used in professional applications encounter an extremely wide dynamic range of sound pressure levels—30dBA SPL (ambient noise in a very quiet

Figure 1: Here is a block diagram representative of the Neve mic preamp.



room) to more than 130dBA (i.e., the threshold of feeling). A typical low-impedance (200Ω) microphone over this range of SPLs might vary its output from $\approx 20\mu\text{V}$ to 2V_{RMS} , while its self-generated output noise would be around 10dBA SPL in a state-of-the-art example. Thus, the single-most important attribute of any microphone preamplifier should be that it contributes no more noise to the output signal than does the resistive portion of the source impedance (**Figure 1**).

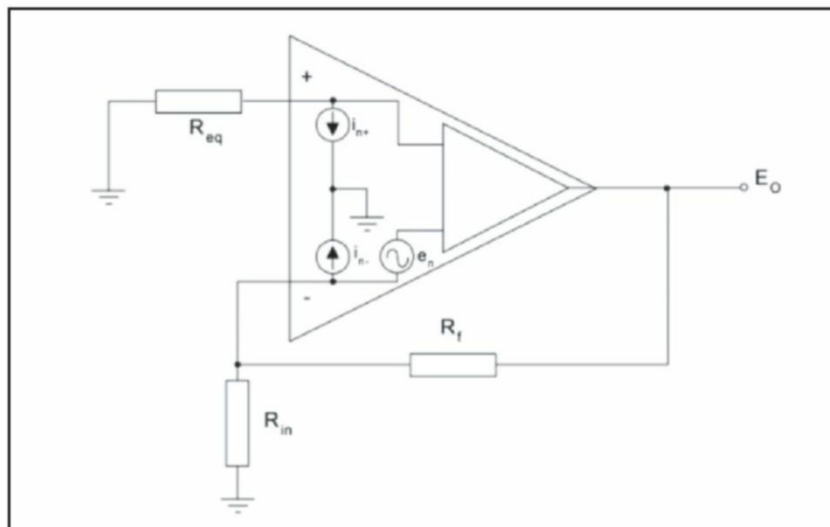


Figure 2: Noise model of an op-amp

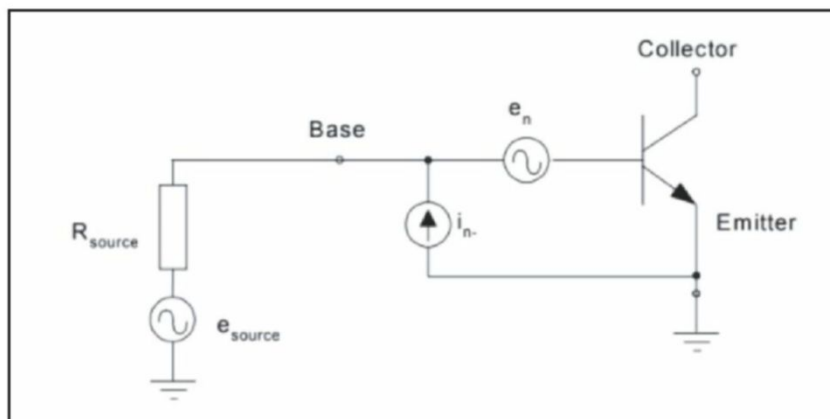


Figure 3: Noise model of a transistor

Given a theoretically perfect, noise-free amplifier, the lowest equivalent input noise (EIN) or “self-noise” of -129.6dBu (with $200\Omega R_{\text{SOURCE}}$) in a microphone input is equivalent to an 26dB SPL on a typical microphone: room noise—typically $\approx 33\text{dB}$ mark—is more likely, in many applications, to limit small signal capability. But real amps aren’t noise-free, their noise varies with gain.

In days past, it was often reasonable audio practice to allow a higher level of microphone-inherent and (amplifier) input noise—ambient room noise would tend to mask it. Such would cause a V_{NOISE} at the microphone output terminals around $>30\text{dB}$ than the intrinsic noise floor (due to R_{SOURCE}) of the microphone itself.

A typical design specification—maximum signal handling capability of -20dBu before undesired increase in distortion, and an input sensitivity down to 0.3mV for full output, yielding a total input signal range of 105dB. However, greater accuracy in digital quantization forces constant review. The increased resolution of digital audio cruelly exposes “front-end” deficiencies: Consumers today have hitherto unparalleled accuracy of reproduction. Subsequent critical evaluation of material allows all manner of sonic shortcomings—insufficient recording booth isolation, poor siting of air-conditioning, microphone stand rumble, geographical proximity of studio to subways/busy roads—to be identified by discerning listeners.

Equivalent Input Noise

Regardless of received frequency of interest, noise is the single most important design parameter and performance-limiting factor in any low-level amplifier. Given a theoretically perfect, noise-free amplifier, the lowest the EIN can be (for a 150Ω source) is -131dBu , but real amps aren’t noise-free—their noise varies with gain. Consequently, resistors—biasing and load, which inevitably form an integral part of amplifier design and application—are problematical: they contribute noise through thermal activity. Circuit models prove helpful in the analysis. Those of an operational amplifier

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and transistor are shown in **Figure 2** and **Figure 3**, respectively. Here, I model noise generators as a series voltage noise generator, e_n , and as two shunt circuit noise generators, i_n . To these, two other noise sources must be added—the thermal noise of the

source resistances seen by the amplifier, R_{SOURCE} , which comprises R_{EQ} and $R_{IN}R_F$. Finally, given values of e_n , i_n , and bandwidth, the circuit's total noise can be approximated as shown in **Equation 1** where:

E_T = total circuit noise

e_n is the amplifier noise voltage—V/√Hz

i_n is the amplifier noise current—A/√Hz (i_{n+} or i_{n-})

R_{EQ} and $R_{IN}R_F$ are the source resistances ($R_{EQ} = R_{IN}/R_F$)

K is Boltzmann's constant, 1.38×10^{-23}

T is the absolute temperature in Kelvin, typically 290K

BW is the noise bandwidth in Hertz

Thus, there are five potential sources of noise to be considered for minimization:

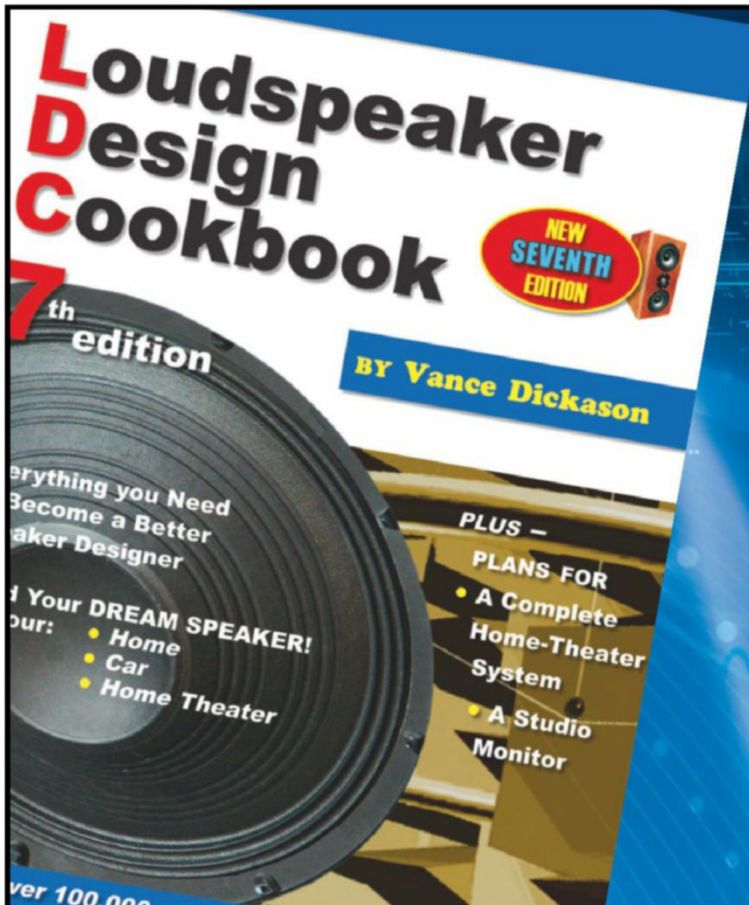
- The thermal (or Johnson, Poisson, or Nyquist) noise of the two source resistances seen by the inputs—this is an irreducible minimum, existing even in an ideal, lossless amplifier
- Both i_n generators
- The e_n generator

$$E_T \equiv \sqrt{\left(e_n^2 + \left[i_n \times R_{EQ}\right]^2 + \left[I_{n-} \times (R_{IN} // R_F)\right]^2 + 4KTR_{eq} + 4KT[R_{IN} // R_F]\right) BW}$$

Equation 1

About the Author

Mike Meechan studied electrical and electronic engineering in Glasgow and has both BS and MS degrees in the discipline. Until 1996, he worked as senior engineer at the BBC's World Service, first at the Bush House in London, then at BBC Monitoring, Caversham Park, Reading. 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. A Member of the Institute of Incorporated Engineers, Audio Engineering Society, and an Associate Member of PLASA, he has authored 80 articles/papers on pro-audio and broadcast-related matters, and designed, built, and published numerous acclaimed pro-audio products.



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For low values of source resistance, the effect of i_n is minimal: e_n will dominate as the source of amplifier noises. As R_{SOURCE} increases, the effect of i_n increases. Thus, at high values, $i_n + (R_{EQ})$ and $i_n \cdot (R_{IN} // R_{EQ})$ comprise the dominant components of amplifier input noise.

Where the source resistance thermal noise is known, Equation 1 simplifies to become:

$$E_t \cong \sqrt{(e_n^2 + [i_n \times R_s]^2) + e_s^2}$$

Equation 2

where E_t = total input referred noise, e_n is the amplifier noise voltage V/√Hz, i_n is the amplifier noise current in A/√Hz, and R_{SOURCE} is the source resistance.

As e_n and i_n are reduced, the total noise approaches the thermal noise of R_{SOURCE} . To match a given R_{SOURCE} to the characteristic noise resistance, R_{NOISE} , of the op-amp, R_{NOISE} must be calculated from the specified data for e_n and i_n as follows:

$$R_{NOISE} = \frac{e_n}{i_n}$$

Equation 3

where e_n is in V/√Hz and i_n is in A/√Hz. Below Z_{SOURCE} of around 10kΩ, bipolar types (because of lower e_n) score over field-effect transistor (FET) input types, and for the venerable bipolar NE5534, an industry stalwart for around 40 years, the values of e_n and i_n are 4.0nV/√Hz and 0.4pA/√Hz, respectively. Therefore: using Equation 4, with the previously specified data (when optimized and equal to R_{NOISE}), and values for e_n and i_n , we can calculate the noise figure (NF) precisely.

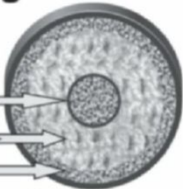
$$\begin{aligned} NF &= 10 \log_{10} \left\{ 1 + \frac{e_n^2 + (i_n R_s)^2}{4KTR_s} \right\} \\ &= 10 \log_{10} \left\{ 1 + \frac{(4.0 \times 10^{-9})^2 + (0.4 \times 10^{-12} \times 10 \times 10^3)^2}{4 \times 1.38 \times 10^{-23} \times 290 \times 10 \times 10^3} \right\} \\ &= 10 \log_{10} \{ 1 + 0.199 \} \\ &\approx 0.8 \text{ dB} \end{aligned}$$

Equation 4

Remember, however, that this is the NF achieved when the NE5534 is driven from a 10kΩ source equal to the optimum source impedance (OSI) of the op-amp. It is not the figure for an amplifier driven by a low-impedance microphone (**Figure 4**).

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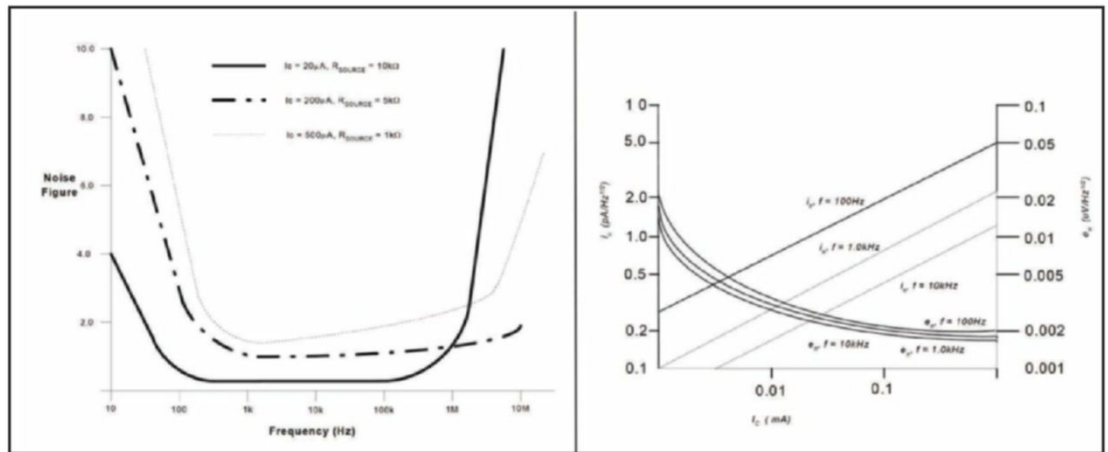
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Figure 4: Graphs showing NF vs. frequency for three choices of IC and R_{SOURCE} and equivalent RMS input noise voltage (e_n) and noise current (i_n) vs. IC for a BC560



Where we can control R_{SOURCE} , the amplifier's characteristic noise resistance— R_{NOISE} —may be used advantageously: best noise performance for a given IC is obtained when $R_{MICROPHONE}$ equals R_{NOISE} . Note that the degree of noise impairment is sometimes known as the amplifier's NF (or noise temperature), and often expressed as a ratio as follows: where $NF(dB) = \log(NF \text{ of the power ratio})$

$$NF = 10 \log_{10} \left\{ 1 + \frac{e_n^2 + (i_n R_s)^2}{4KTR_s} \right\}$$

$$NF = \frac{SIGNAL_{OUT} \times NOISE_{OUT}}{SIGNAL_{IN} \times NOISE_{IN}}$$

Equation 5

Summary of Requirements

By any standard of pro-audio design, distortion and noise characteristics alone required of the preamplifier are virtually beyond compare, certainly as far as analog audio is concerned. However, this is just part of the story. Now factor in that this performance must be achievable with input signals ranging in quite normal circumstances from -70dBu to +20dBu. (The most demanding applications widen this dynamic range by $\approx 30dB$).

In more recognizable domestic hi-fi terms, this compares at the low end to a moving-coil phono cartridge's output to a V_{OUT} nearly four times that from a CD player: and noise and distortion characteristics across this range must remain both constant—and constantly low!

However... this system is non-resident in the benign setting of a typical domestic living room. Designs must also discriminate 'wanted' from 'unwanted' signals in an environment where RFI via sources such as cell phone breakthrough and other sources is commonplace, crosstalk from adjacent pairs not unheard-of, and generally imperfect transmission media an everyday eventuality. (The use of long connecting cables – typically $\approx 100m$ to the FOH desk in a live concert environment—exacerbate this).

Thus, professional microphone outputs are balanced, driving a twisted pair of wires with signals in anti-phase. In the next article of this series, we explore the various circuits used as the front end of the mic preamp to discriminate wanted from unwanted (common mode) signals—the difficult-to-achieve audio elixir in a low-noise mic preamp design.

Resources

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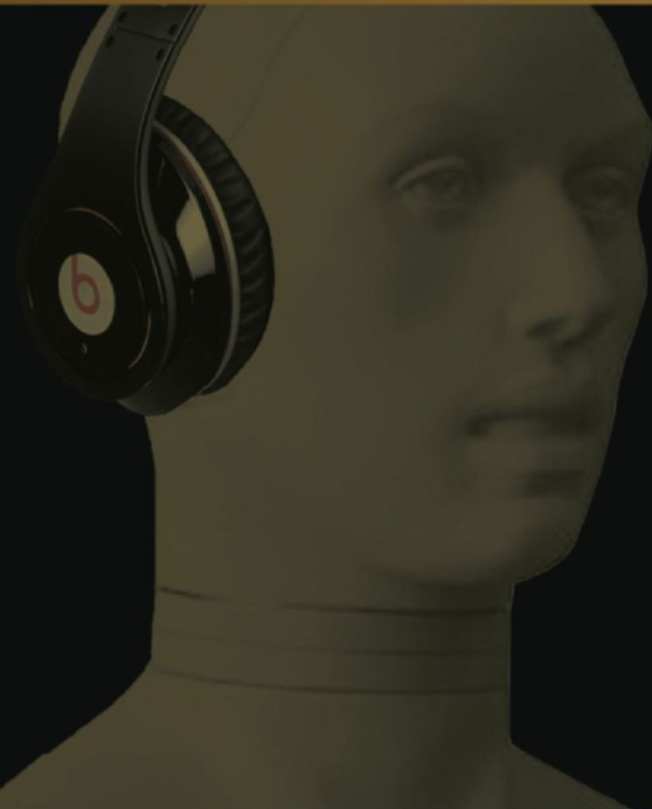
L. Tyler, "De-Integrating Integrated Circuit Preamps: Less Delivers More?" THAT Corp, www.thatcorp.com/datashts/Deintegrating_Integrated_Circuit_Preamps.pdf

Valuable resources also include: THAT Corp. content, particularly papers by Fred Floru, Principal Engineer, THAT Corp.; Gary W. Herbert, CTO, THAT Corp.; Les Tyler, President, THAT Corp; and Bill Whitlock, owner and chief engineer, Whitlock Consulting and former president of Jensen Transformers, Inc.; and books and papers by Ben Duncan and Doug Self.

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X-Altra Moving Coil/Moving Magnet RIAA EQ Preamplifier (Part 1)

Design Overview and MC Preamplifier



For this DIY project, Andrew Russell shares the X-Altra Phono EQ Preamp, which is an accurate, very low noise high-performance phono equalizer that will accept moving coil and moving magnet inputs.

By
Andrew C. Russell

The X-Altra phono preamplifier can be used with both Moving Coil (MC) and Moving Magnet (MM) pickup cartridges and separates the MC and MM input circuits to optimize overall noise performance. It features an ultra-low noise LSK389B JFET from Linear Systems in the MM front end with switchable gain and resistive loading. The MC front-end utilizes “current injection” input stage with Zetex ZTX851/951 bipolar devices for use with cartridges down to 3Ω coil resistance.

The MC preamplifier stage is best suited to cartridges with output currents in excess of 12μA. The design incorporates a 20Hz high-pass filter that can be switched to 45Hz to reduce “acoustic” rumble evident on some classical recordings. Surface-mount device (SMD) components are used extensively, and the result is a compact unit that fits into a Modushop “Galaxy” half-width housing. All DIP switch settings are accessible on the unit’s rear panel (**Photo 1**).

PCBs are available should builders wish to try out the design. The rear panel is made from drilled and milled double-sided, matt-black silk screened and gold-plated THP PCB, which replaces the aluminum rear plate supplied with the Modushop housing.

A complete unit will cost about \$300 to \$350 in construction costs, including the housing and PCBs (**Photo 2**).

Phono Preamplifier Design Considerations: Noise and Overload

A MM source poses a number of design challenges with respect to noise, where there are three major thermal noise sources to consider: the amplifier’s noise voltage, its noise current, and thermal noise of the real resistive part of the source impedance. The cartridge can be modelled as an inductor of between 400mH and 800mH in series with a (typically) 500Ω to 1.5kΩ resistor—the figures are dependent upon cartridge make and model, which is usually loaded with 47kΩ and between 100 pF and 200pF arising from cable capacitance. At low frequency, the input referred noise source impedance is simply the generator resistance shunting the 47kΩ load resistor and any high performance op-amp (e.g., an AD797 or a LT1115) would achieve around -75dB to ~76dB signal-to-noise ratio (SNR) ref 5mV. However, the cartridge impedance rises quickly with frequency so that by 20kHz, it is as high as 15kΩ to ~20kΩ.

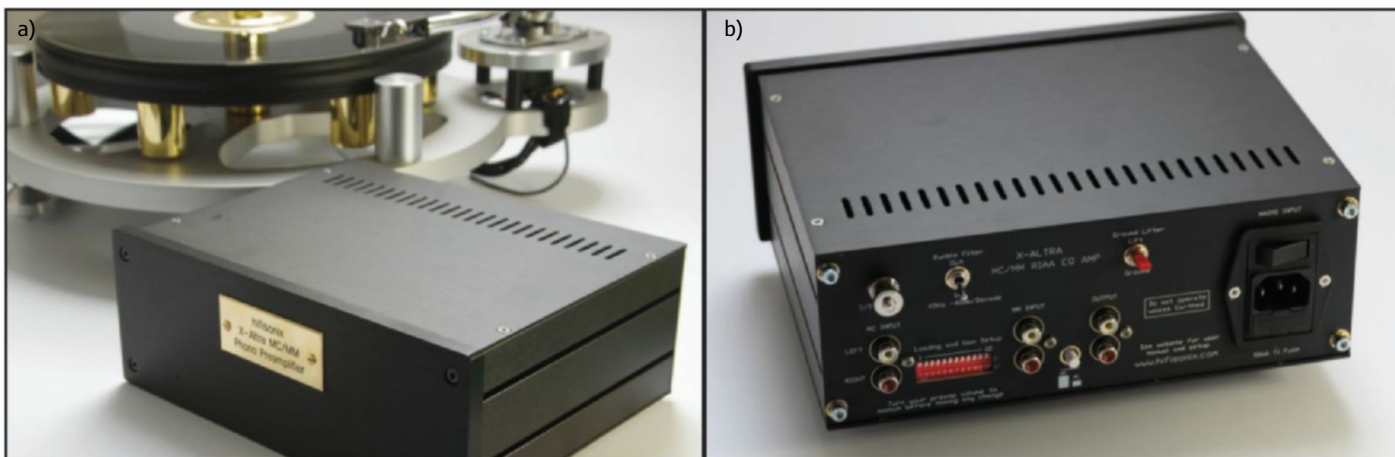


Photo 1: The X-Altra RIAA design incorporates a 20Hz high-pass filter that can be switched to 45Hz to reduce “acoustic” rumble evident on some classical recordings (a). SMD components are used extensively. All DIP switch settings are accessible on the unit’s rear panel (b).

This shunts the 47kΩ input load resistor, so at 20kHz the amplifier sees a source impedance around 10kΩ to ~15kΩ, rather than just the DC resistance of the pickup coils, which is the case at low frequency. The amplifier input noise current flowing through this source resistance gives rise to a noise term that adds RMS style to amplifier input noise voltage and the thermal noise voltage of the generator resistance itself.

In a MM bipolar input amplifier, noise current flowing through the input source impedance quickly limits achievable noise performance. In a typical cartridge with a DC resistance of 1350Ω and $L = 500\text{mH}$ (which I will refer to as the standard cartridge for the rest of this article), the total integrated noise over the audio band is approximately 3.1μV RMS or about 20x more than an op-amp such as an AD797 fed from a low-source impedance. One of the reasons the 40-year-old bipolar input NE5534A (the teal trace shown in **Figure 1**) remains so successful as a general-purpose MM phono amplifier building block is that its input noise current is low at 0.4pA/√Hz, which is about one quarter that of other low noise op-amps (e.g., the AD797 or the LT1115).

Junction field-effect transistors (JFET) op-amps have much lower noise currents, but input referred noise voltage is generally higher, although there are a few outstanding devices (e.g., the AD745) that are better than the NE5534A but at a significant cost premium. However, using a discrete JFET input design, even lower noise voltages can be had at the expense of a more complicated circuit.

Using a NE5534A, about the best practical noise floor achievable reference 5mV at 1kHz with an all-active EQ design and a shorted input is -77.5dB. Stuart Yaniger has produced a useful spreadsheet that will calculate the best SNR one can expect given the cartridge L , R and the amplifier R and C input loading parameters (see Resources). With our standard MM transducer -69dB ref 3mV or -72dB

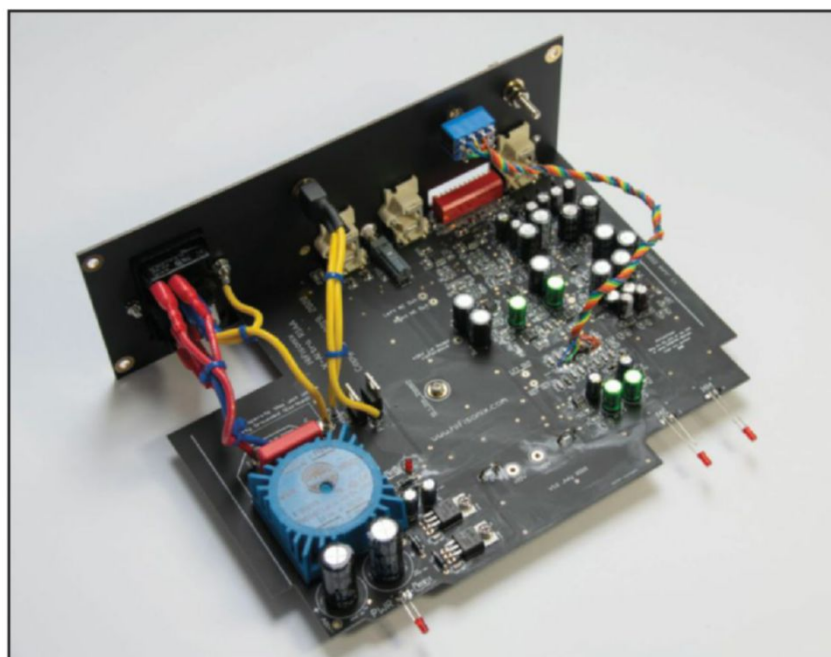


Photo 2: A complete unit will cost about \$300 to \$350 in construction costs, including the housing and PCBs.

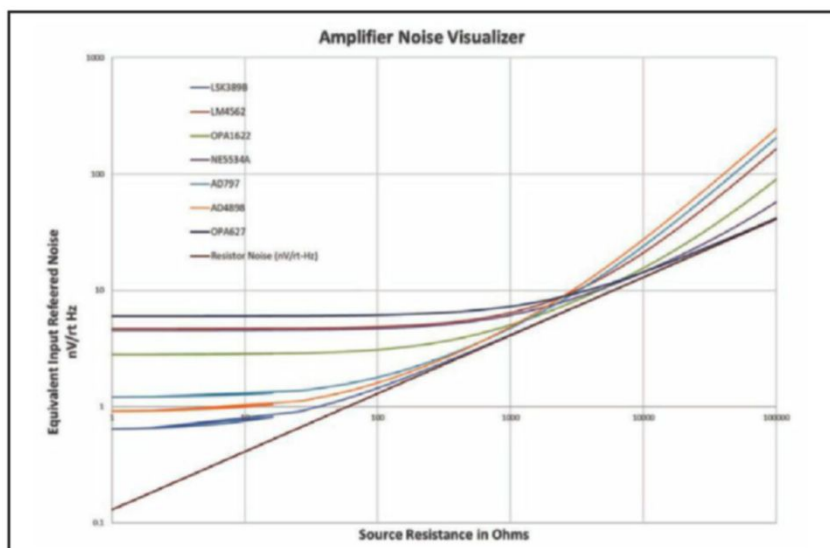


Figure 1: Selected device noise performance vs. source resistance

ref 5mV (both figures unweighted) is about all that is achievable. Using an LSK389B JFET front end with both devices wired in parallel (**Figure 2**) and a 110Ω gain setting resistor in the feedback network, a figure of -75.3dB ref 5mV is attainable in an all-active design, which is >3dB better than the NE5534A and >10dB than an OPA1602 in a similar configuration. Note: Vinyl surface noise will degrade these figures by approximately 15dB to 20dB even on a good pressing.

MC transducers feature much lower coil resistances, for the most part obviating concerns about input current noise. However, many highly

regarded MC cartridges feature coil resistances of below 10Ω and outputs in the 100μV to 200μV range, and here the focus must be on minimizing the amplifier input noise voltage to get decent performance.

Despite a lot of effort by some very skilled participants on “Richard Lee’s Ultra low Noise MC Head Amp” thread over at diyAudio, no one was able to convincingly beat the noise performance of Lee’s “Duraglit Special” given the two transistors and handful of passives used in the circuit. Lee’s circuit was a development of Marshall Leach’s design dating back to 1978, with input referred noise in the region of 200pV/√Hz to 300pV/√Hz readily achievable, making it the go-to topology for low-source impedance generators. (Richard Lee’s original head amp was built in a “Duraglit” polish tin, which was a household soft metal polish used on brass and pewter sold in the UK and hence became known as “Richard Lee’s Duraglit special.” Duraglit is no longer available generally.)

On both MC and MM inputs, the other area that presents a design challenge is that of overload margin. A number of papers have addressed this subject over the years, but probably the most widely cited reference is Tomlinson Holman’s 1977 “Dynamic Range Requirements of Phonographic Preamplifiers,” which leveraged studies from across the industry and academia. In most commercial phono preamps, gain switching in the phono stage is not provided and the only adjustment to cater for higher level output cartridges and recordings is the volume control, which is placed after the phono stage in the signal chain.

The designer, therefore, must cater to higher output transducers, “hot” recordings and the inevitable clicks and pops with their wide spectral content by ensuring overload margins of at least 25cm/sec re the 5cm/sec mid-band reference output. An extract from the Holman paper (**Figure 3**—with the green and red trace overlays added by the author) shows peak measured signals across a range of recordings with one of them achieving >70cm/sec at approximately 4kHz—a remarkable figure. Peak output signals appear to be clustered in the 2kHz to 10kHz region and 6cm/sec to 40cm/sec output range, with a few outliers higher than this as mentioned earlier.

Of interest here is many of these peak events are in the mid-band or just beyond, exposing the amplifier to high input signals at high (approximately 35dB) gain. At 50Hz and below, the equalizer gain is 20dB higher than that at 1kHz so warp and off-center records can also cause low-frequency overload problems, not shown in Figure 3 but discussed in the Holman paper. With all-active feedback and some gain

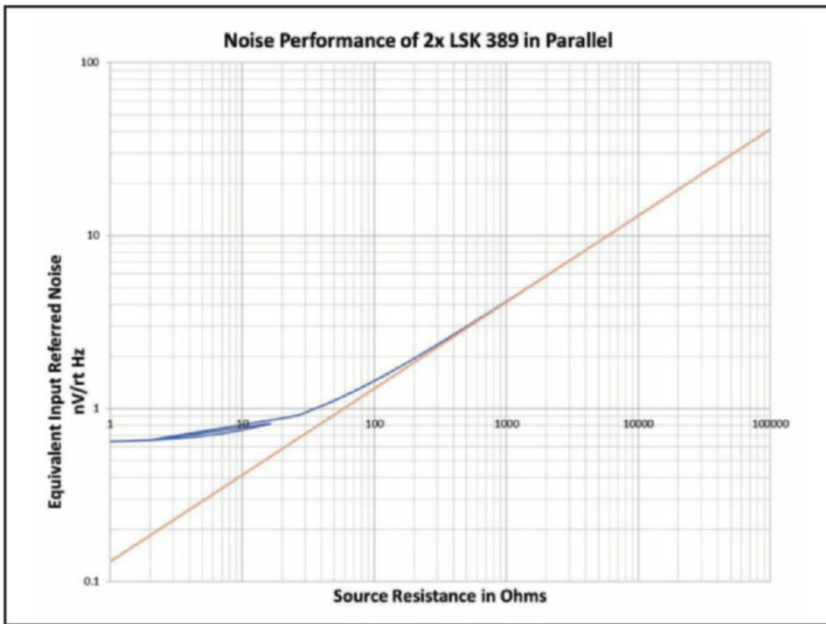


Figure 2: LSK 389 noise performance

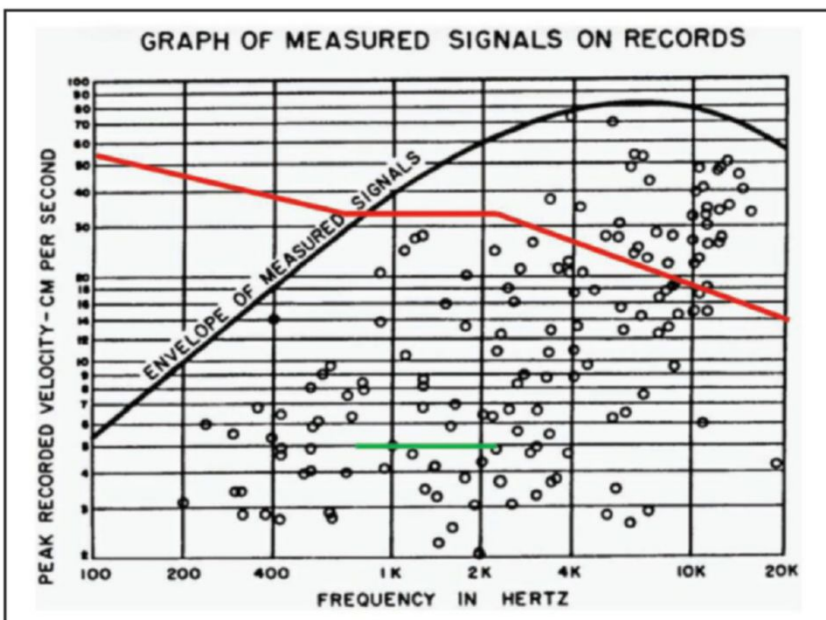


Figure 3: Peak vinyl outputs (Source: Holman, et al)

switching to cater for different cartridge sensitivities, it is relatively easy to achieve >30dB overload margins on ±15V supply rails ref 5mV input levels and similarly for MC inputs ref 500µV, over 20Hz to 20kHz.

System Block Diagram

For this entire document, descriptions will refer to the left channel, the right channel being identical, other than the system power supply unit (PSU), which is shared by both channels. The key component numbers and values shown in **Figure 4** correspond to the circuit diagram component references to be discussed in depth later.

MC Front-End Amplifier

Table 1 shows there is considerable current available from a typical MC cartridge with which to develop the output voltage across a suitably configured load resistor. Since the input signal, V_{cart} , is small, and a fraction of the transistors V_{be} , the distortion is low even when operating without feedback.

For a common base circuit operating in current injection mode, the idealized gain is given by:

$$V_{out} = I_{in} \times R_{load}$$

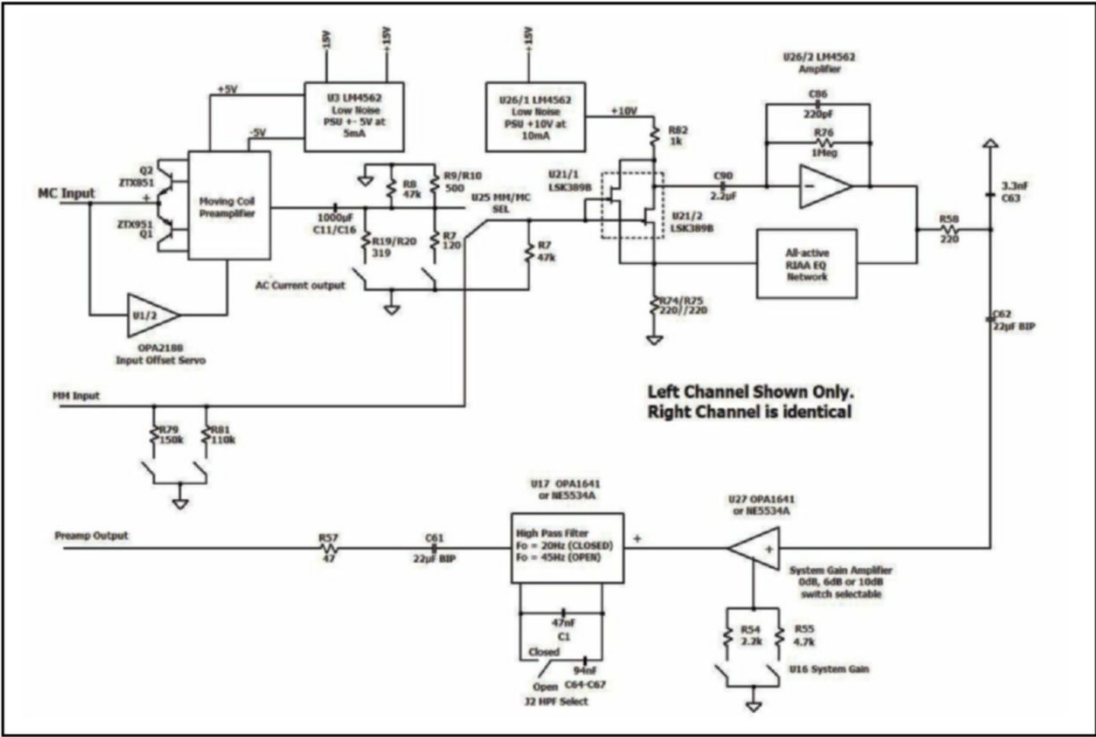


Figure 4: X-Alta block diagram

Cartridge Manufacturer and Model Number	Cartridge DC Resistance	Cartridge Output	Preamp Input Current, Including Amp R_{in} Error	Reswitch Setting	Required Voltage Gain for 5mV O/P	Actual Output Voltage
Goldring Ethos	4	0.0005V	69µA@1kHz	85	10	5mV
Ortofon Kontrapunkt "b" MC Cartridge	5	0.00047V	57µA@1kHz	107	11	5mV
Ortofon MC Vivo Blue	6	0.0005V	54.1µA@1kHz	114	10	5mV
Goldring Elite	8	0.0005V	44.4µA@1kHz	146	10	5mV
Ortofon MC Anna Diamond	3	0.0002 V	32µA@1kHz	229	25	5mV
Ortofon MC Anna	6	0.0002V	21.6µA@1kHz	435	25	5mV
Ortofon MC A Mono	7	0.0002V	19.5µA@1kHz	494	25	4.8mV
Ortofon MC Windfeld Ti	7	0.0002V	19.5µA@1kHz	494	25	4.8mV
Ortofon MC 10/MkII	3	0.0001V	16µA@1kHz	494	50	4mV
Hana EL and SL	30	0.0005V	15µA@1kHz	494	10	3.7mV
Denon DL103	30	0.00039V	11.7µA@1kHz	494	13	2.9mV

Table 1: Selected MC cartridges and associated gain requirements

However, the input impedance of the amplifier, R_{in} , is not zero, and related to the internal device emitter resistance r_e' :

$$r_e' = .026/I_c$$

where I_c is the collector current of Q1 and Q2

In this design, an I_c of 3.5mA was chosen as it provides the best compromise between lowest noise and reasonable power consumption resulting in a r_e' of 7.5 Ω . Since Q1 and Q2's r_e' appears in parallel to AC, the error term is 3.25 Ω and not negligible in the case of low generator coil resistances. Setting V_{out} to 5mV so the MC gain is up front to maximize noise performance, the required load resistor R_{Ltot} is given by:

$$R_{Ltot} = (.005/V_{cart}) \times (R_{cart} + R_{in}) \quad (1)$$

The 1k Ω resistors R9 and R10 loading the collectors from the power supply are in parallel with the 47k Ω output load resistor for AC signals providing a pre-load of 494 Ω , so we need to subtract this from the calculated R_{Ltot} to arrive at the (nearest) U2 switch load resistor setting:

$$1/R_{switch} = 1/R_{Ltot} - 1/494 \quad (2)$$

Table 1 lists some MC cartridges, their output

voltages, coil resistances, and resultant output currents along with the desired gain for 5mV output. If a cartridge is not listed in Table 1, apply the formulas in (1) and (2) to find the required R_{switch} to select the closest value with the DIP switches. Note that the gain can be further optimized by changing the system gain amplifier settings (discussed in Part 2 of the article), which would, for example, be required for the Denon DL103 and the Hana EL and SL cartridges to raise the overall preamplifier output. The maximum gain achievable is when the MC gain setting resistors (DIP Switch U2) are in the OFF position and the output gain setting load 494 Ω .

Referring to **Figure 5a** left hand channel, the input signal feeds into Q1 and Q2 (Zetex ZX851 and ZX951), which are configured as a complementary common base pair. These devices are currently the lowest noise small signal packaged devices available with r_b' independently measured and verified by a number of contributors on the Internet at 1.33 Ω and 1.4 Ω , respectively. In this design, that translates to an input referred 1kHz spot noise for the amplifier of 210pV/ $\sqrt{\text{Hz}}$ simulated or about one-quarter that of an ultra-low noise op-amp like the AD797, and a total integrated RMS noise over the audio band of 30nV RMS.

The common base configuration shown contributes no noise referred to the input signal other than r_b' . Because low output MC generator

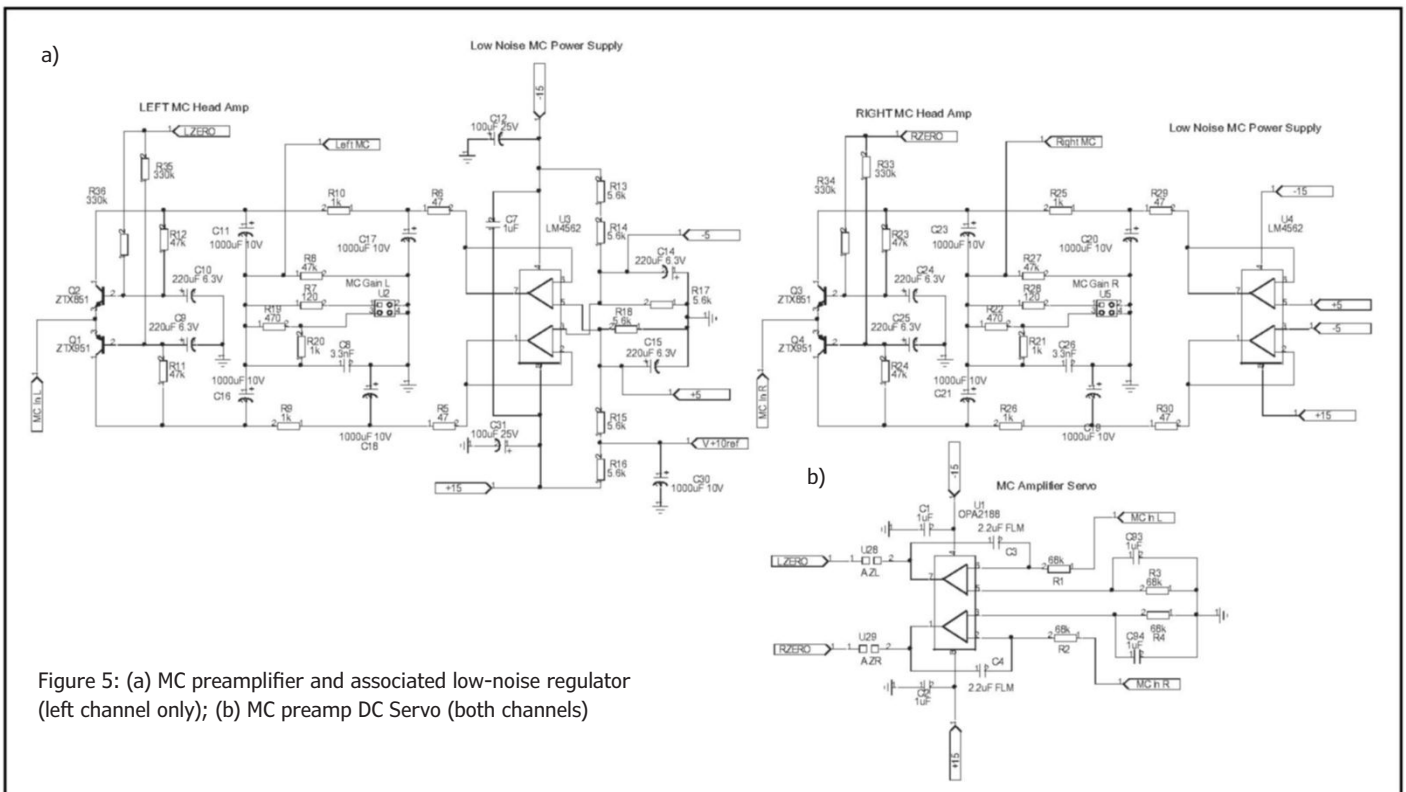


Figure 5: (a) MC preamplifier and associated low-noise regulator (left channel only); (b) MC preamp DC Servo (both channels)

resistances are generally not more than 10Ω , the noise current contributes little to noise performance, unlike the MM amplifier stage. R11, R12 set up the transistor collector current to around 3.5mA and C9 and C10 provide filtering and short the bases together at AC.

R7, R8, R19, and R20 combine the output signals from the collector load resistors R9 and R10 via C11 and C16 and provide loading adjustment via a two-way DIP switch U2 (MC Gain L) allowing the load resistor to be set to 66Ω , 83Ω , 194Ω , or 494Ω (the parallel combination of R9, R10, and R8). The output is then routed to the source selector switch U25.

The power supply rejection ratio (PSRR) of the MC head amplifier is zero. Rather than a discrete capacitance multiplier or a high capacitance RC filter, this approach leverages the outstanding PSRR of the LM4562 (U3), which at 200kHz and a 600Ω load exceeds -80dB while at low frequency, the figure approaches -110dB.

The U3 PSU reference voltages (also used by the right-hand channel regulator) are derived through the divider network comprising R13 through R18 and associated noise filtering provided by C14 and C15. The output filter comprises C17 and C18 with

resistors R5 and R6 further attenuating noise from the regulator. The capacitors have been oversized so that their equivalent series resistance (ESR) is low—anything more than a few ohms will contribute excessive thermal noise to the supply rails.

Simulations indicate a 1kHz spot noise of approximately $50\text{pV}/\sqrt{\text{Hz}}$ per rail or about 7nV RMS wide band noise. For the **Figure 6** plot, the input reference voltage $220\mu\text{F}$ filter capacitor ESR was set to $600\text{m}\Omega$ and the $1000\mu\text{F}$ output decoupling capacitor to $140\text{m}\Omega$ —both reflective of readily available components at reasonable cost.

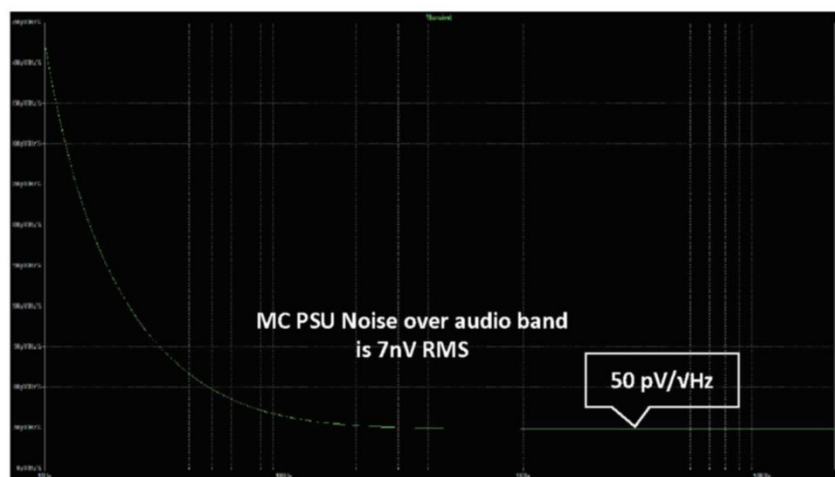


Figure 6: LM4562 power supply's simulated output noise spectral decay density

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Figure 7 shows the measured noise floor of the full MC plus MM preamplifier signal chain. The QA 401 noise floor is at around -155dBV on these plots. At 50Hz, the red trace the gain is 76dB, at 1kHz 56dB and at 20kHz, 36dB. The measured input referred noise on the red trace at 1kHz is approximately

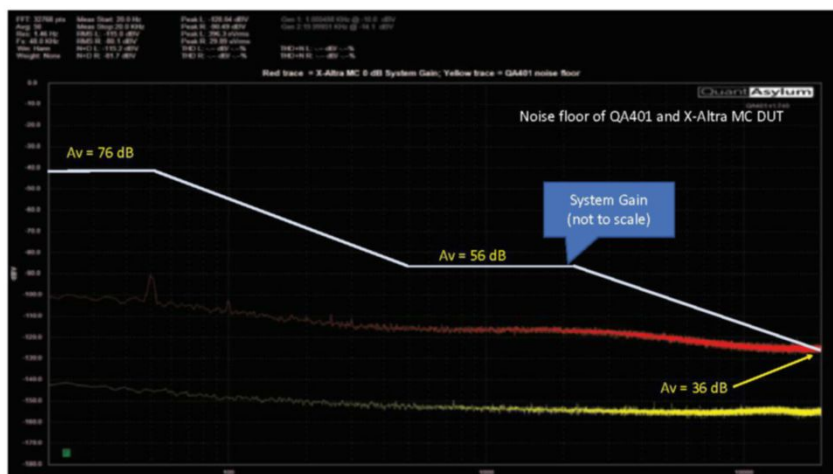


Figure 7: MC preamplifier self-noise

About the Author

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



220pV/√Hz vs the simulated figure of 210pV/√Hz.

The input offset voltage as measured at the junction of Q1 and Q2 emitters can be as high as 30mV due to normal V_{be} parameter spreads between the two transistors. In the battery-powered version, the power source is floating and no offset current flows through the cartridge coil; with a ground referenced supply that is not the case. An OPA2188 based servo (U1, Figure 5b) adjusts the bias on the bases of Q1 and Q2 via R35 and R36 (330kΩ each), forcing the input voltage to 0μV. The connection to the servo control point is by means of two flash pads U28 (left channel) and U29 (right channel).

During test and setup, the offset voltage is measured between the signal input and the signal ground at the input connector, which is typically 5mV to 15mV. After this, the pads are flashed closed and the offset then measured again which should be zero. I used a DVM with a 200mV FSR and was unable to resolve any offset, which is specified at ±6μV typical for the OPA2188 (±25μV maximum). It is important that the measurements are taken with the 0V reference right at the input connector as even a 1mV drop across the 0V rail will cause a severe reading error. The output of the servo amp (pins 1 and 7 on U1) once engaged should not exceed about ±6V, allowing the servo amp to output an additional ±6 V control range.

Next Month

Part 2 of this article will discuss the MM stage design, the system gain amplifier, the filters, and the measurements.

Project Files

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

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