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ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Automotive Audio Solutions

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A Glimpse of the Future, Inside a Car

A year and a half later, the devastating global effects of the COVID-19 pandemic are still far from over. Fortunately, the successful take-up in vaccines and new treatments, innovations in products and services, and overall strong social and business resilience, afford us relatively high levels of optimism for a strong global economic resurgence.

Most macroeconomic and industry reports tell us that while demand for many products was drastically reduced, changes in consumer behavior have also accelerated, and major trending shifts created strong opportunities. As most countries and large economies start to gear up for what promises to be a very strong post-pandemic economic recovery, it becomes clear that the pandemic accelerated technologies and consumer trends by at least a few years.

In parallel with spectacular sales in product categories such as televisions, computers, and even microphones, triggered by stay-at-home consumers, adoption of new technologies accelerated in-home networking, streaming content subscriptions, video conferencing, and remote collaborations. Despite the disruptions in manufacturing, product categories such as smartphones and smart home devices remained resilient. And new working practices have created strong incentives for investments in strategic areas such as IP-based infrastructures, cloud and broadband services, and 5G networks.

The audio industry in most terms has proven resilient throughout the pandemic, and some product categories in home audio or communications have turned into promising areas of growth. According to Futuresource Consulting, the value of the Home Audio market in 2020 was the highest it has ever been, rising 4% to \$19.8 billion. Soundbars and wireless speakers, which are relatively new product categories, have also seen very strong growth.

As I previously noted, while lock downs have frozen many activity sectors, the pandemic has in fact contributed to accelerate research and development efforts. Even if some standardization efforts have been delayed, the underlying work on specifications has continued and that will be reflected as soon as organizations and consortia are again able to convene with a clearer vision of the road ahead. This will enable accelerating trends that weren't a priority until now, such as in smart home technologies, and most certainly, automotive.

Many of the same technologies that have seen a resurgence or spectacular adoption at home during the pandemic, such as immersive audio and streaming services, will now also transition to the car. First, through aftermarket products, but steadily into new in-vehicle infotainment systems.

According to ABI Research, aftermarket sales and revenues for automotive applications have dropped 28% in 2020 because of the pandemic, and although they are being threatened by almost ubiquitous OEM embedded systems, hardware shipments are still estimated to continue to grow until 2026. Because consumers will look for ways to bring the technologies they are using to the car—even before they buy a new one.

Regardless of its many unique challenges, automotive audio technologies are one of the most interesting markets at the moment, conveying all the technological and consumer behavior trends in a well-defined ecosystem—a small microcosm of what the future will bring. One where it is easier to implement complete solutions, instead of one-off features—as Harman promotes.

Even before electric or hydrogen-fueled cars evolve into autonomous or shared-mobility models, consumers will expect enhanced audio quality and innovative experiences. Virtual acoustics, advanced noise cancellation, sound zones, augmented reality, and advanced voice assistants all make sense inside the car. And elusive concepts that are just a nice-to-have elsewhere, are expected to become indispensable requirements in self-driving or driver-assistance systems.

Before the pandemic, innovation was taking place in the mobile industry, where the investments and computing power were available. Post pandemic, the car will be the most valuable extension of mobility and a natural laboratory for cutting-edge experiments.

J. Martins
Editor-in-Chief

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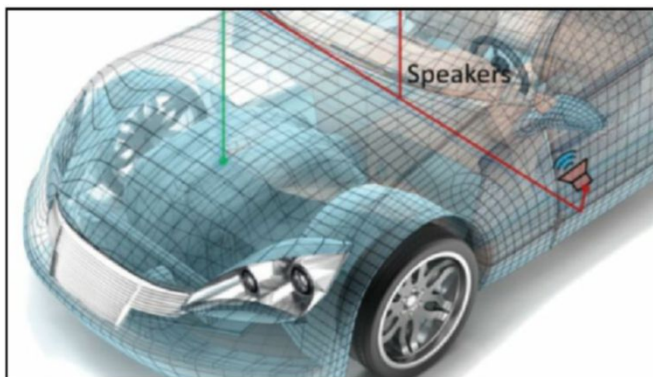
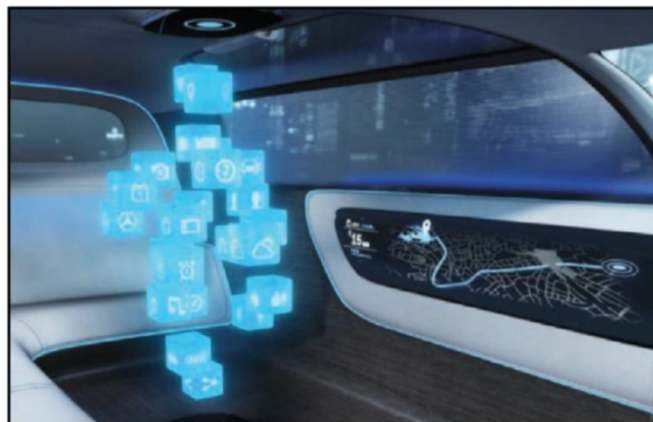


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By Oliver Masciarotte and Kent Peterson

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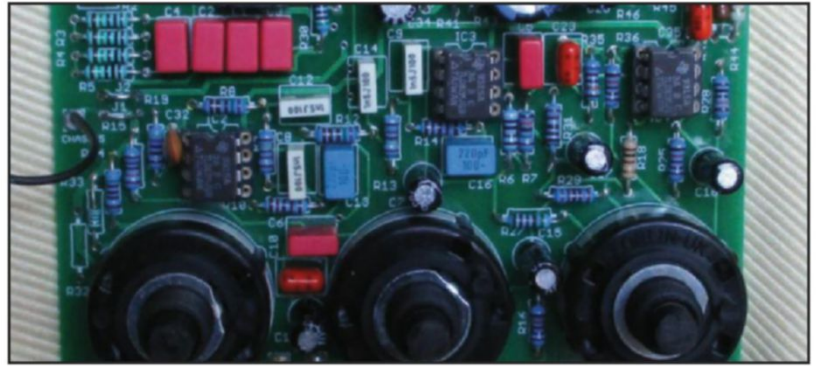
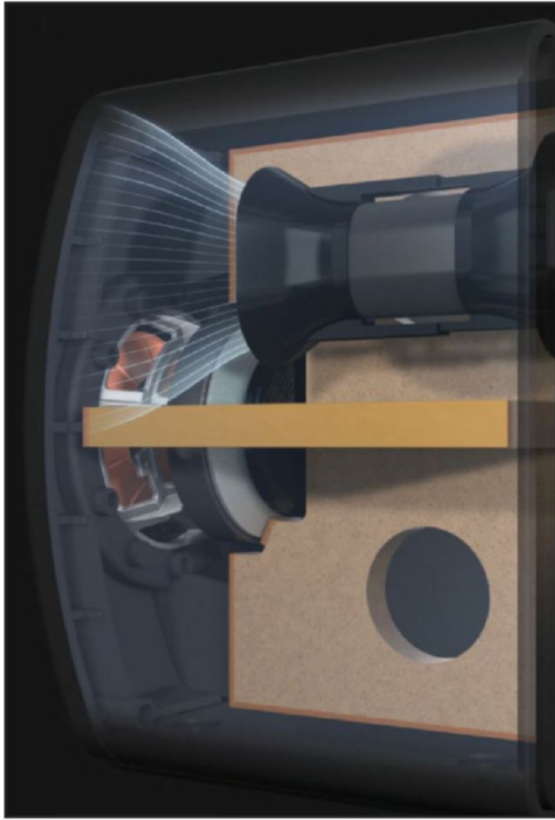
By Mike Meechan

In the previous articles, Mike Meechan considered the physics that play a fundamental part in microphone pre-amplifier design, explored its inhabitation in the real practical world to retain the fidelity of the mic signal along its path from transducer to output, and looked at what (theoretically) contributes to perfect active mic preamps. He concludes his article series with the golden rules of low-noise, mic preamp design.

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As consumers demand even more audio performance in ever-smaller packaging, Infineon's new MERUS™ family of Class-D amplifiers uses an innovative multilevel technology to deliver new levels of efficiency across the whole audio frequency spectrum while significantly relaxing the requirements for filtering. These improved efficiency levels open up new possibilities for the audio system designer.



● Automotive Audio Solutions

MARKET UPDATE

8 Tech for the Automotive Framework

By J. Martins

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By Richard Honeycutt



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Tech for the Automotive Framework

AUTOMOTIVE

By
J. Martins
(Editor-in-Chief)

Human Horizons is a premium smart all-electric vehicle brand from China that delivers an immersive in-car audio experience using Meridian's Re-Q and Meridian Digital Precision technologies.

In this brief Market Update we focus our attention on a few key technologies, platforms, and innovations that should be on every developer's radar when working on automotive audio projects. Audio solutions that are transversal to the vehicle's design and human interface.

As Dennis Goldenson, Research Director, Automotive and Emerging Technologies with SAR Insight and Consulting, correctly frames it in the company's recent Global Consumer Insights, "Audio is a significant component of the in-vehicle experience from head units and amplifiers to sound processing on the edge and speaker configurations. As a result, from a technology perspective, we can expect architectural transitions in the near future from the consolidation of the electronic control unit (ECU) and DSP to a centralized computing platform for seamless transition of infotainment features and controls within the connected vehicle framework."

And the connected vehicle framework is more than certain to encompass large dashboard, touchscreen displays, voice interfaces, and access to streaming services as the most visible components of the audio ecosystem, while all audio systems in itself—speakers, microphones, and processing units will remain mostly in the background. For the user experience though, immersive audio content is front and center in the vehicle experience, enabling efficient communications, creating distinct zones when required for multiple users, generating comfort and awareness as required, and most importantly, augmenting the whole driving experience.

This represents a growing business opportunity for companies active in the automotive segment, as well as for the front-facing consumer brands, which will remain an essential part of the marketing mix.

According to market research from Technavio, the automotive premium audio system market is poised to grow by \$13.86 billion USD from 2020 until 2024, progressing at an annual growth rate of more than 6%.

Even as the COVID-19 pandemic affected car production and sales globally, investment in the automotive audio segment remained firm and had a low impact on the automotive audio system market.

Harman, Continental, and Bosch continued to lead the way in connected infotainment systems, a segment that generated \$32 billion USD in revenues in 2019, according to ABI Research. Harman is the clear leader, with Continental in a distant second place, and Bosch a close third. Still, these three companies together accounted for only 30% of the total market revenue in 2019, revealing a highly competitive market that is bound to become even more fragmented as more key-technology players are focusing on automotive audio applications.

ABI Research's ranking of Tier One Connected Automotive Infotainment Suppliers identifies AISIN, Alpine, Aptiv, Bosch, Faurecia, Continental, Denso, Harman, Hyundai MOBIS, JVCケンウッド, LG Electronics, Melco, Panasonic Automotive, Pioneer, and Visteon as leading players in the space.

With navigation and multimedia offerings increasingly standardized, particularly with the introduction of Apple CarPlay and Android Auto, innovation in next-generation infotainment is now shifting toward offering platforms based on an integrated cockpit domain controller that can be scaled across different vehicle segments and advanced HMI features. While Harman leads the ranking because of its strong portfolio of connected infotainment solutions, Continental—a traditional player in the automotive industry—outperforms competitors in implementation due to higher OEM collaborations.

In this highly competitive and complex environment, audio technology companies and audio brands remain a key supplier of many of the key components that contribute to improved sound within the car. And that is the reason why we continue to see many marketing partnerships that explore consumers' brand perceptions at the core of many announcements in this space.

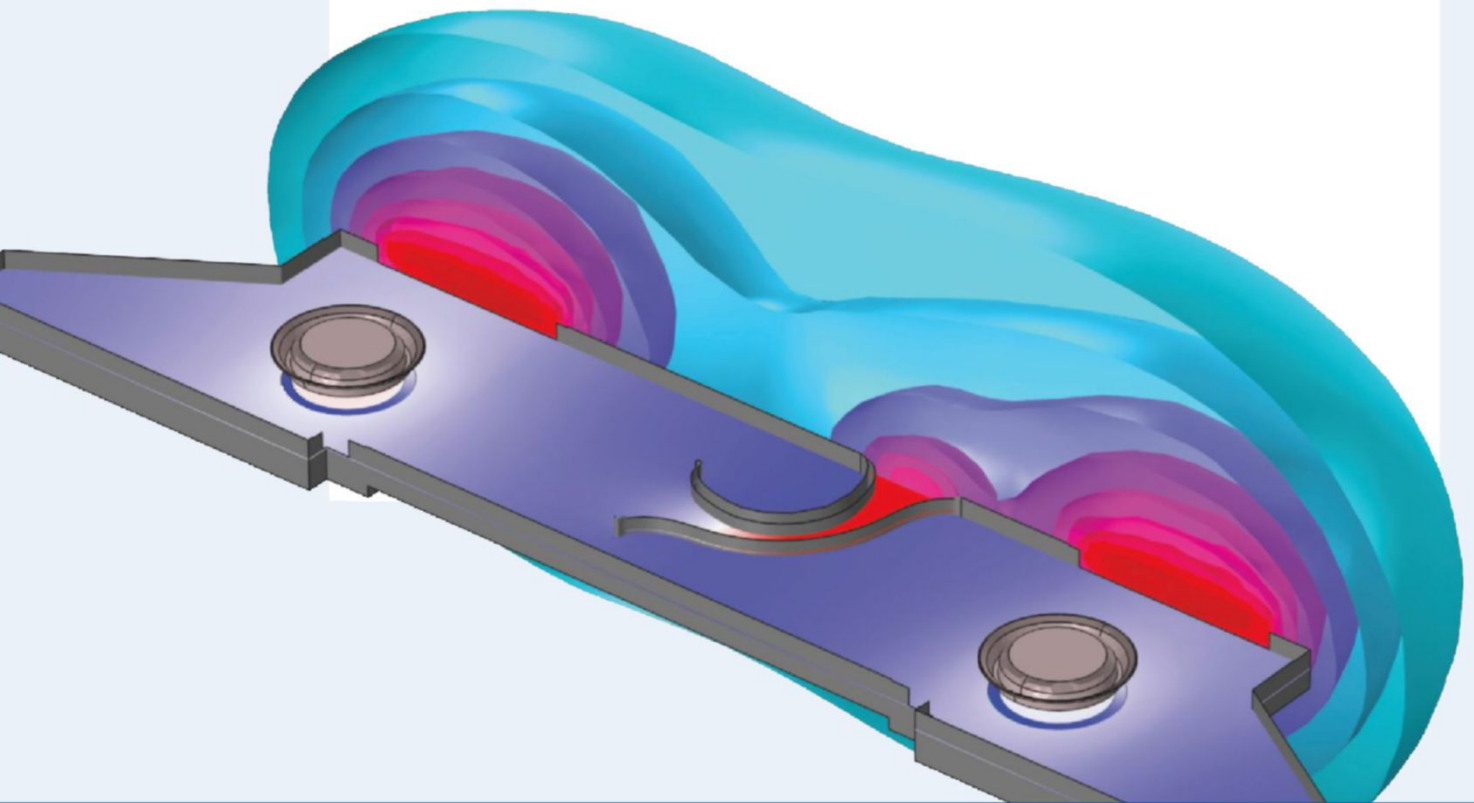
As an example that follows last year's partnership between Continental and Sennheiser—which this year was selected as a CES Innovation Awards 2021 Honoree at this year's online CES 2021—

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Panasonic Automotive Systems, Klipsch, and Dolby presented a proof-of-concept project that highlights the value of immersive sound and music introduced in a vehicle.

As a global player in this space, Panasonic is working with VOXX International's consumer audio brand, Klipsch, to bring new speaker technologies into Panasonic's in-vehicle premium-branded audio offerings, while Dolby is a key partner in promoting and certifying a complete Dolby Atmos content chain. Dolby Atmos content is expanding from movies and TV to music, and consequently from the home theater to mobile and the connected car. And Panasonic Automotive is one of the key Dolby partners to bring the Dolby Atmos experience to the road. The proof-of-concept result highlights the huge potential of an immersive soundscape for the automotive cabin, particularly with music.

In a similar approach, LG Electronics is working with Luxoft, a DXC technology company, in a joint venture aimed at enabling digital, consumer-grade experiences in automotive based on the webOS Auto 2.0 platform. The effort is lead by Alluto, a new company headquartered in Santa Clara, CA, that will drive the commercialization of production-ready digital cockpit, in-vehicle infotainment, passenger-seat entertainment (PSE) and ride-hailing systems based on the webOS Auto platform.

A clear alternative to Google's efforts with Android in the automotive space—which most car brands consider a dangerous and undesirable Trojan horse, and it is also very poorly perceived by consumers—webOS Auto 2.0 offers access to in-vehicle and connected services and brings greater control and convenience through enhanced head unit functionality. Most importantly, as a highly customizable and scalable platform for the complete digital vehicle, LG's webOS Auto enables OEMs to create their own user experiences.

The Sound

Advancing the transition from home to mobile and to the road, Swedish digital audio processing pioneer Dirac is a very active player in many of the most cutting-edge initiatives in automotive audio.

Dirac's patented impulse response correction technology is widely adopted by automotive sound systems that boast, maximum, 12 amplifier channels. Dirac's patented MIMO mixed phase correction technology, a solution for the high-end market, is regarded as the

industry's most advanced automotive technology, and can be used for the highest-end sound systems—that boast 10 to 20+ amplifier channels—to ensure perfect listening experiences in all seats.

Both solutions feature Dirac's digital audio platform and are trusted by world-renowned brands, such as BMW, Rolls Royce, Volvo, Harman, Polestar, and more.

Dirac recently teamed with Harman and Volvo in a flagship project that shows the full potential of digital signal processing (DSP) in car acoustics. The three companies worked in a unique acoustical design project for the 2021 Volvo 60 and 90 vehicle series, aiming to deliver accurate real-world acoustic replications. The project began with its "Concert Hall" mode, which simulates the aural experience of a live performance at the celebrated Swedish Gothenburg Concert Hall. As a follow-up, Dirac recreated the sound experience of Sweden's famed Nefertiti Jazz Club. The Dirac-enabled Bowers & Wilkins car audio system has now added a "Jazz Club" mode, recreating the acoustics of these real spaces within the car cabin.

Both modes were developed in collaboration with Volvo Cars, Dirac, and Harman—and combine Volvo's own acoustic engineering experience with Dirac's digital audio technology and a Bowers & Wilkins' speaker system delivered by Harman's Automotive division.

This serves to show the potential of Dirac's technologies. Faithfully simulating a jazz club or a concert hall in a vehicle cabin presents significant challenges—as each space has unique acoustical characteristics that alter how sound travels and therefore how it's perceived. The original acoustical characteristics of a car cabin are also significantly different from a hall or a club considering the cabin's size, shape, reflective surfaces, and more.

The company's approach to simulating a jazz club or concert hall begins by removing the unwanted acoustic characteristics of the car cabin to allow for the re-creation of new characteristics as realistically as possible. To do this, Dirac leverages its patented MIMO mixed phase correction technology, which co-optimizes all speakers' magnitude and phase response to optimally reproduce each input channel. In doing so, the technology makes it possible to remove the unwanted effects of a car cabin, such as reflections and modal patterns and improve impulse responses digitally—which was previously regarded as possible only in theory.

Once the unwanted acoustic characteristics of the car cabin



Expanding on their successful car audio partnership announcement at CES 2020, Sennheiser and Continental's speakerless immersive audio system for vehicles was selected as a CES Innovation Awards 2021 Honoree. The new audio system explores new ways to enhance in-car entertainment experiences by integrating Sennheiser's patented AMBEO 3D audio technology with Continental's speakerless Ac2ated Sound system.

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An audio system designed by Harman Automotive for Volvo in cooperation with key technology players like Dirac, is visible for the consumer as a recognizable brand such as Bowers & Wilkins.

are removed, new ones are then re-created with high degrees of accuracy. Dirac measured and collected the spaces' "acoustic fingerprint data" (e.g., reverberation patterns, reflections, frequency and spacing) and then copies this "fingerprint" into the car cabin—creating an algorithmic structure to accurately mimic those attributes in a vehicle interior.

The algorithm was handed off to Harman specialists, who—with Volvo's support—integrated it with the 19-speaker, 1400W Bowers & Wilkins stereo system available in the 2021 Volvo 60 and 90 series.

In a different project, Dirac and Chinese car company BYD Motors, one of the leading automotive companies in the hybrid and fully electric vehicle space, created a new in-car listening experience for BYD's new Han series of vehicles.

Dirac's cutting-edge processing technology optimizes the impulse and frequency response of all speakers within the vehicle's sound system to improve audio precision and performance, and minimize sound quality variations from seat to seat.

This collaboration follows projects with prominent manufacturers BMW, Bentley, and Rolls Royce, to introduce cutting-edge audio enhancements from Dirac and provide a more enjoyable driving experience in new vehicles. "Electric and hybrid vehicles



After securing a position in the high-end automotive market, Swedish audio processing pioneer Dirac announced the introduction of a new digital audio solution designed specifically for entry-level audio systems.

are inherently quieter than their internal combustion engine counterparts," states Lars Carlsson, Head of Business Development, Automotive at Dirac. "As a result, poor audio quality cannot be masked by the loud roar of the engine; all is exposed in the near-silent electric and hybrid vehicle cabin."

All the models from the mentioned brands include the Dirac Premium solution, featuring Dirac's frequency and time domain equalization, speaker and bass optimization, room correction, and other premium technologies for optimal listening at all seat positions. Mid- and entry-level BYD Han Series models will feature the Dirac Performance solution, which includes frequency domain equalization, speaker and bass optimization, room correction, and other features for achieving upgraded sound performance in a cost-efficient design.

Both Dirac Premium and Dirac Performance packages also come with the company's center-image optimization and filter design tool.

After securing a position in the high-end automotive market, Dirac recently announced the introduction of a new digital audio solution designed specifically for entry-level audio systems, expanding its digital audio solution to more automotive manufacturers. This enables automotive manufacturers to cost-effectively upgrade their entry systems' performance to deliver enhanced sound quality, reduced cabin colorations, generate a surround sound experience from stereo content, and introduce signature sound across models with high consistency.

"Dirac was born out of the automotive industry, and we've spent the last 15 years designing digital audio solutions for some of the industry's most well-regarded and discerning brands," explains Lars Carlsson. "However, with the introduction of our first-ever solution designed specifically for the entry market, we're now bringing our renowned audio innovations from the few to the many—and equipping entry market vehicles with optimized sound experiences."

Dirac's new digital audio solution for the entry market is built upon an all-in-one intelligent audio platform. It consists of a measurement system, tuning tools, sound optimization algorithms, and an IO to head-unit platform that—together—enable manufacturers to achieve perfected sound faster, more easily, and consistently across vehicle models.

The platform features the industry's first semi-automatic tuning system that optimizes performance and delivers signature sound in a way that is highly efficient, consistent, and predictable—eliminating the typical guesswork involved in the traditional manual tuning process, which significantly reduces overall tuning time.

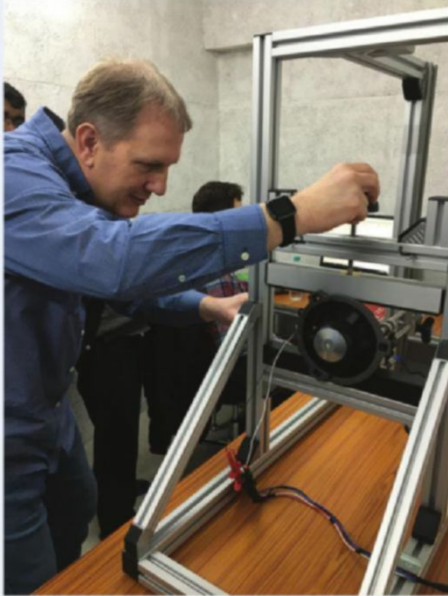
The Dirac platform also includes patented digital sound optimization and spatialization technologies that vary depending on the targeted markets. For the entry market, the available technologies are measurement-based magnitude response correction, patented virtual bass, surround sound, and virtual center—to tackle common acoustic challenges like negative car cabin impact, unbalanced soundstage, and lack of immersion.

The Space

Israel-based Silentium specializes in creating personal in-car sound environments, and advanced, broad-band Active Road Noise Cancellation (ARNC) solutions. When in 2020 the company introduced

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From sophisticated audio processing with the Dirac Premium solution, Dirac now also offers the ability to replicate the acoustic of real live music environments and transform the in-vehicle listening experience.

its advanced ARNC technology in a partnership with Jaguar Land Rover, Silentium finally received well-deserved attention, after many years in stealth mode.

Jaguar and Land Rover are the first carmakers to integrate Silentium's "Active Acoustics" software in three of their new vehicles, meaning the technology is now available for car buyers to experience.

ARNC removes 90% of unwanted noise across a wide range of frequencies—from 20Hz up to 1kHz—providing a quieter and more refined experience for occupants, and therefore preventing driver fatigue. In the new Jaguar F-PACE model, Silentium's implementation is capable of lowering unwanted noise peaks by 10dB and overall noise levels by 3dB to 4dB—the equivalent of turning down the in-car sound system by four "steps."



McIntosh Laboratory implemented its latest McIntosh MX1375 Reference Entertainment System, employing 23 specifically tuned speakers and a new Adaptive 3D Surround Processing system in the upcoming 2022 Jeep Wagoneer and Jeep Grand Wagoneer.

ARNC combines with Engine Noise Cancellation (ENC) on the new Jaguar F-PACE and new Range Rover Velar with the P400e plug-in hybrid (PHEV) powertrain to further enhance the sense of well-being. In addition, Silentium's Active Acoustics technology offers vehicle manufacturers a way to reduce their reliance on costly passive noise damping and insulation materials, and reduce vehicle weight—an increasingly important R&D factor as the industry enters a new era of electro-mobility.

Silentium's technology is similar to that found in a pair of high-end noise-cancelling headphones, but more advanced as it manipulates a larger amount of air in a vehicle. Up to six strategically positioned accelerometers monitor unwanted road noise and send a signal to an on-board control unit with Silentium's software, which plays an equivalent anti-noise signal through the vehicle's speaker system. The pressure waves from both the unwanted exterior noise and the manufactured anti-noise reach occupants' eardrums at exactly the same time and cancel each other out. And by monitoring sound in real time, unexpected noises from potholes or rough surfaces can be isolated and removed. In the Jaguar Land Rover models, the cancellation sound is played through the Meridian sound system.

But Silentium is also a pioneer in the sound zone concept for automotive applications and its ARNC solution perfectly complements the concept of not only creating sound zones that can be controlled by each passenger, but actually allow different activities inside the vehicle.

Using Silentium's Personal Sound Bubble (PSB) concept, any environment can be divided into several independent listening zones, meaning a personal audio experience is delivered to each area. In a two-zone test between the front and rear passengers, featuring four door woofer speakers and four ceiling tweeter speakers in the front and back, a contrast of up to 20dB could be created in sound broadcast levels. Not yet enough to allow each listener to enjoy his own choice of music, but enough to allow someone to listen to music loud enough to stay awake and alert, or have a phone conversation in the driver's seat, and not disturb the kids sleeping in the back. For now, Silentium's PSB is used to create two independent sound zones, one for the front driver and another for the rear passengers—using conventional car speakers placed in the doors and ceiling.

An even more ambitious concept in sound zones might soon be possible thanks to another Israeli company, which focuses on speakers with sound-beaming technology.

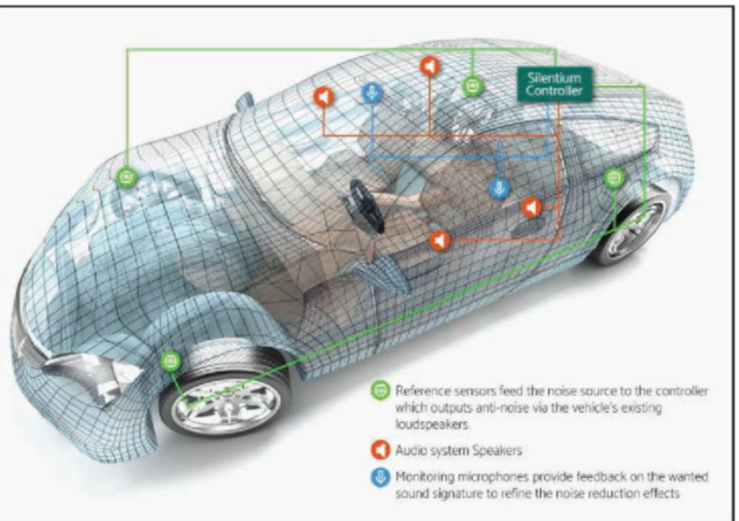
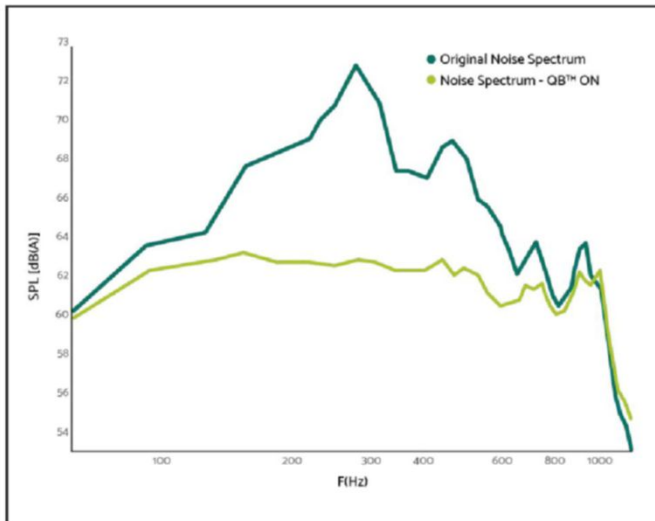
Noveto successfully demonstrated its sound-beaming technology for web conferencing and automotive zoning applications at CES 2018, and more recently partnered with electronics giant Foxconn to bring to market its own audio chipset. Now, the company is launching its own speaker design, and launched a Kickstarter campaign to bring to market a proof-of-concept—the SoundBeamer 1.0.

Noveto's patented technology approach uses an ultrasound 3D sensing module to locate a user standing in front of the speaker, and beam acoustic energy to create tiny sound bubbles precisely next to his ears. When the user moves his head, the sound bubbles will follow. For its Kickstarter speaker campaign, Noveto is promoting the concept as an alternative to using headphones when users need to isolate themselves, or while listening to audio and not disturb others. But the actual implementation combines ultrasound beams

A close-up, high-resolution photograph of a human ear, showing the intricate details of the ear canal and the surrounding skin. The image is in a dark, monochromatic blue tone, giving it a professional and scientific feel. The ear is positioned on the left side of the frame, with the rest of the head and neck visible in a soft, out-of-focus background.

THE **NEW** ONE FOR ALL

Welcome a new era for measurements around communication and audio quality: realistic, precise and flexible for all types of applications.



Silentium's Active Road Noise Cancellation (ARNC) technology employs proprietary algorithms and strategically located transducers to reduce road noise as perceived within the cabin of a vehicle by the occupants.

and audio modulation to generate audible sounds within a spatial location determined by ultrasonic frequency components—in other words, beaming the sound just to our ears.

The user's ears location determines the focal points for the ultrasound beams, enabling the generation of audible sound in the vicinity of the designated spatial location. And this is combined with a transducer system that includes a panel having one or more piezo-electric enabled foils and an arrangement of electric contacts coupled to the panel. This arrangement enables the efficient conversion of electrical signals to mechanical vibrations (acoustic waves) and/or vice versa.

The Noveto D1 Chipset holds its proprietary smart algorithm to control its sophisticated beamforming, face detection, tracking

capabilities and state-of-the-art DSP engines, enabling it to locate the position of the user's ears in space and beam acoustic energy around the user's head.

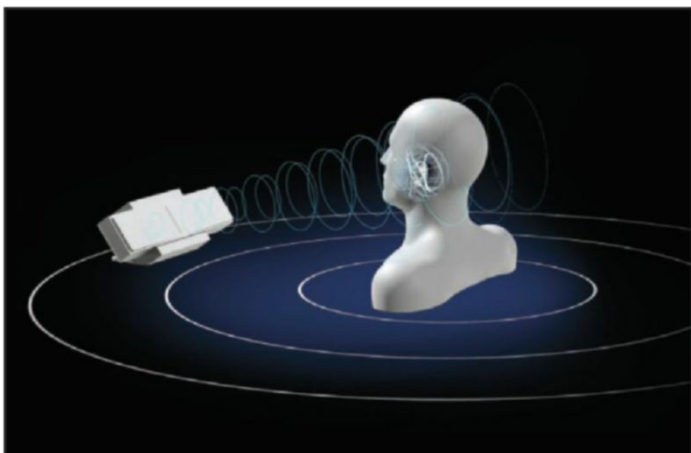
The combination of those patented technologies is what makes Noveto's technology so impressive and could certainly power automotive applications for vehicle cabin sound zones.

The company has been busy completing all the pieces of the solution, while its licensing efforts continue to take place.

Another company that is already on the radar of many developers working in advanced digital signal processing for automotive applications is DSP Group. The company is accelerating its development and rapid deployment of advanced audio technologies and recently announced the opening of an audio innovation lab in Straubing, in southern Germany, focused on advancing the state-of-the-art in audio signal processing and artificial intelligence (AI) for acoustic engineering.

The Straubing lab is already staffed with experts, each with more than 20 years of experience in the areas of voice and audio processing algorithm development and deployment, speaker nonlinear processing, AI-based audio processing, ANC, and rapid prototyping and tuning. More importantly, it adds domain-specific expertise in automotive audio, in-cabin and road ANC to DSP Group, since most of its engineers previously worked for Harman Becker Automotive Systems (Straubing Division), which was shut down in 2020, with work relocated to other Harman facilities in Hungary and Asia.

With the recent acquisition of SoundChip SA—a leading supplier of ANC technology, and production-line test systems for headphones and headsets, based in Switzerland, and with research facilities in the United Kingdom—DSP Group significantly reinforced its presence in Europe. DSP Group is headquartered in San Jose, CA, and Herzliya, Israel, and already has an operation in Germany, mainly to support some of its largest clients, which included Harman's Automotive and Lifestyle Audio divisions.



Noveto's proprietary transducer array is a multi-cell module that converts digital signals into non-audible acoustic waves, which beam audio to just outside the ears, creating a personal audio experience. The Noveto-developed proprietary DSP algorithms are the key to control the custom-designed transducer array, which generates the waves that are beamed to the user.

The establishment of the Straubing lab also comes at a time of rapid advances in voice integration and active approaches to noise cancellation in automobiles. These approaches are leveraging multiple microphones and DSP algorithms integrated into increasingly advanced automotive audio systems to cancel noise. This ensures a pleasant driving experience, while also enabling higher-performance voice user interfaces (VUIs) for calls and system control. With this new team and expertise, DSP Group will be able to quickly expand its automotive audio technology solutions, certainly a key player to consider in voice-assistant integration inside the cabin. 



In automotive applications, Noveto's sound-beaming technology is a step forward in the concept of individual sound zones.

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Silentium, Ltd. | www.silentium.com

Volvo | www.volvo.com



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Meridian Audio

Transforming Listening Experiences on the Move with New Technologies

British audio pioneer Meridian Audio has been creating innovative audio technologies and elegant, high-performance audio products since 1977. Meridian's ambition is to enable people to listen, enjoy, and fall in love with their music and movies wherever they are.

The company has enjoyed a great deal of success in the home audio sector with its easy-to-install high-resolution loudspeaker products and has established itself as one of the most highly regarded names in the high-end audio market.

By challenging convention and pushing boundaries with its program of continuous research, Meridian has developed proprietary technologies and delivered products that have both disrupted and shaped the audio industry. Meridian is recognized and appreciated by a sizable global base of consumers seeking a better audio experience wherever they listen.

Meridian and the Automotive Market

For more than 10 years, Meridian has built a strong presence within the automotive market. Working alongside market leading suppliers and collaborating with highly respected car brands to deliver innovation, audio leadership and immersive, captivating experiences for all vehicle occupants. Meridian has designed and delivered world-class audio systems for McLaren Automotive, Jaguar Land Rover, Human Horizons, Rivian, and Kia.

Meridian has a very close and successful long-term relationship with Jaguar Land Rover, which enables customers to enjoy the finest premium audio across their entire portfolio of vehicles. Together, Meridian and Jaguar Land Rover have brought innovations and new features to market, including 3D surround sound for the all-new Range Rover 2012 and the first high-end audio system in a battery electric vehicle with the award-winning Jaguar I-Pace in 2016.

More recently, Meridian has designed immersive in-car audio systems to integrate within Human Horizons' electric super SUV, HiPhi X; Rivian's ground-breaking range of fully electric adventure

vehicles, including the R1T and R1S; and a purpose-designed 14-speaker surround sound systems for the Kia, starting in the recently launched K8 sedan.

In order to successfully maximize the customer benefits of working with Meridian, the company has increased investment in its research programs and dedicated engineering resources to address the future needs and unique challenges of automotive audio. Moreover, Meridian has created a flexible network of collaborative relationships with leading engineering and manufacturing suppliers within the industry, enabling it to maintain its position of audio leadership and as a premium audio brand and technology supplier of choice.

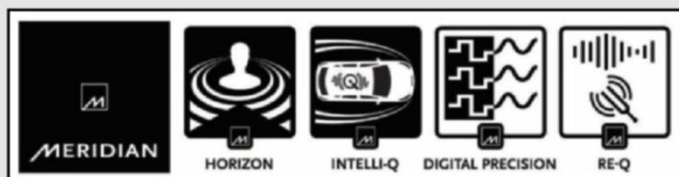
By working closely with its customers, Meridian engineers will lead the architecture design and the production of technical documentation for the system, which encompasses the design, specification and the review and approval of processes for hardware components to be sourced. The company's sound ambition is to provide premium audio experiences that integrate seamlessly with the design intent of the vehicle and deliver optimal and immersive audio to all occupants in the car.

Solving In-Car Audio Challenges

Through continuous research Meridian engineers have become familiar with the challenges that in-car environments present.



Meridian's systems are renowned for holding up against road noise better than the rest.



Meridian engineers have developed a number of new audio technologies that provide exceptional in-car entertainment experiences.



Jaguar Land Rover vehicles are equipped with a Meridian Surround Sound System, offering peerless audio performance and premium quality.

The team utilizes its in-depth knowledge of psychoacoustics to optimize the DSP audio systems featured in the vehicles to focus on what is most important for the human ear to provide a natural listening experience. The result is a definably clearer, more authentic sound.

Meridian is a recognized authority on digital signal processing (DSP) and this is used as the method of delivery of Meridian's proprietary performance-enhancing technologies. The company adopted DSP more than 25 years ago and has been perfecting it ever since. It provides the foundation to every Meridian speaker and is a key facet of its approach, substantially enriching the audio performance for car brands.

Meridian has improved and built upon its unique DSP technologies for automotive. Alongside Meridian RE-Q and Digital Precision, Meridian now offers the new Horizon and Intelli-Q DSP technologies.

Meridian Horizon is an upmixing technology that provides immersive multichannel audio from two-channel stereo content, configured for any loudspeaker layout. This provides the user with a truly enveloping and immersive listening experience.

Meridian Intelli-Q is Data-Driven Equalization that optimizes audio playback within the cabin based on data available from the vehicle such as speed, window state, occupancy and audio source. The automotive cabin is a highly variable acoustic environment, and this is particularly apparent in electric vehicles where there is no engine noise. Things such as the speed of the vehicle, the number of windows open, the number of occupants or the audio source can result in a highly variable audio experience. The application of Meridian Intelli-Q ensures all occupants in the vehicle enjoy optimal audio experience at all times.

Meridian Digital Precision maximizes the signal-to-noise ratio (SNR), ensuring the finest details of every song are retained. Distortion is reduced and the noise floor lowered, which optimize the signal output meaning the listener experiences every detail of the musical performance with minimal distortion.

Meridian RE-Q is a cabin correction technology that intelligently and discreetly optimizes the audio system to the precise



Meridian worked with Human Horizons to provide an audio system for its electric vehicles known as HiPhi.

acoustics of the vehicle cabin, preserving the natural rhythm of the performance. This unique Meridian technology enables bass notes to be reproduced cleanly, ensuring that the clarity and definition of each instrument and vocal are tailored to the interior of the cabin.

Meridian's In-Car Solutions

When developing these new automotive audio solutions, Meridian also puts great emphasis on enabling competitive advantages across key areas of customers' businesses. Focus is put on the design flexibility, energy efficiency, weight reduction, and customer experience.

In a Meridian designed in-car audio system, the team deploys its expertise in Digital DSP loudspeakers to unify and control all the elements that, in a conventional sound system, are separated: crossover, conversion, power amplifiers and the loudspeaker drivers themselves. This system approach retains absolute precision of the audio signal, resulting in a sound that is natural, three-dimensional, and remarkably easy to listen to. Meridian's transparent DSP performs the conductor's role within an orchestra. It ensures perfect harmonization among components by delivering specifically tailored sound signals. Such levels of accuracy, precision, and performance are impossible to achieve in the analog domain.

Working closely with customer's vehicle designs, Meridian is able to establish which combination of technologies will work best, as well as determine the speaker configuration within their cabin. With audio performance and design aesthetics in mind, speakers can be strategically positioned and tuned by Meridian engineers to create a completely natural soundstage.

Compared to rival offerings, Meridian's systems are renowned for their clear and well-defined bass, which creates an enjoyable and accurate sound stage even for back-seat passengers. Vital bass frequencies remain distinguishable regardless of environmental conditions, such as wind, tire or road noise even at motorway speeds.

Meridian's unique, elegantly engineered, holistic system approach, where all the elements of the system and space work seamlessly together, is meticulous in every detail and is renowned for offering the ultimate in audio performance while on the move.

Meridian's approach to in-car audio is transforming listening experiences on the move, creating differentiated, premium offerings for automotive brands that go beyond the ordinary.



Speakers are strategically positioned within the cabin and measured in their performance to create a luxurious yet completely natural Meridian soundstage.



The Changing Automotive Environment in 2021

By
Roger Shively

This update on the automotive audio industry discusses what is new in the user interface to the audio system, how cockpits are changing, and the approach to optimizing them for the end user's audio needs. It also looks at how audio systems are expanding to integrate Active Noise Control (ANC), Advanced Driver-Assistance Systems (ADAS), and Acoustic Vehicle Alert Systems (AVAS) for quiet car sound design and autonomous driver alerts. Finally, it examines new challenges for OEM audio system designs and how they are being met.

Without the physical interaction of conferences and trade shows in 2020 and early 2021, OEM carmakers, their Tier One suppliers, and independent consultants worked primarily in isolation on new ideas or focused more in-depth on ideas that were already in development. In-person meetings were no longer necessary, but communication within large global companies and internal collaboration improved. Software investments climbed. Everyone incorporated the necessary tools at home and many people experienced less distractions, as it were, on day-to-day hands-on activities. Tier One companies saw no loss in projects. 2020 produced more RFQs than before—planning for the immediate needs and the future goals.

As for the auto industry, insiders agree that the most inventive carmakers understand audio is not just about music and entertainment anymore. They have also focused on crafting signature engine sounds and creating auditory experiences that are truly unique.

The audio system has become increasingly important and its uses have expanded. There are also different goals than there were five years ago when it was a race to get as many speakers in a car and to get the power levels as high as possible.

Now, the quality of components and efforts to better integrate them are among the biggest improvements in the last 3 to 5 years. Carmakers are paying more attention to how they are designing their systems at all levels—no longer just writing off the entry-level sound systems.

Even what we thought 18 months ago has changed. And there are new viable OEM players (e.g., Tesla, Sono Motors, and Rivian) that might have been written off just a year ago.

With my background starting in the mid-1980s working with a Tier-One and interfacing with most of the OEMs, my own consultancy business JJR Acoustics is in a unique position as an independent contractor for OEM automotive audio. We work, and have regular conversations, with those in the industry about problem solving and future development for every aspect of automotive audio—whether it be for component or system design, a full-range of acoustic simulations, or all forms of in-vehicle or virtual tunings.

The state of our industry, the challenges it faces, and the more significant trends are a constant part of that discussion. Now, we are continuing to learn more about the technologies that started making waves in 2020—but with new twists.

Significant Audio System Trends

For carmakers, audio has become more of the differentiator between highly optimized mechanical experiences of power train and handling. First, there needs to be a good basis of audio quality and acoustic quality of the vehicle, and then features can be added to meet customer expectations.

Functionality is more important today: What you actually can do in the car, as well as security, parking assistance, and noise cancelling. The audio system now has many more uses.

And with the movement of high-end audio to meet the end customers' expectations of higher audio quality, audio signal processing companies (e.g., Dirac) are introducing new solutions that were once based in the mid-range and high-end market, but now target entry-level systems.

Harman International is focusing on the end users, keeping their potential moods in mind. Besides the more standard setting options for the end users (balance, fader, EQ), there are more systems that come with pre-defined sound modes or advanced audio adjustment technology that enable the end users to select a mode at the touch of a button to fit each individual's mood when driving. In Volkswagens, there are four Harman Kardon presets (PURE, CHILL-OUT, LIVE, ENERGY), and in Bentley's Continental GT, Bang & Olufsen's "One Touch User Experience" lets drivers choose music defined by their moods (Bright, Energetic, Warm, or Relaxed).

For all carmakers, Apple CarPlay (iOS) and Android Auto (open-sourced) are must haves. The combination of better audio and enhanced features is how many OEMs stay in the game.

There is a greater trend to open-source solutions in the cockpit domain controllers (formerly called head-units, or more simply radios). The traditional closed system is going, or gone, because of the loss of the traditional source media (discs or physical storage devices). The non-traditional sources are non-terrestrial radio sources (e.g., streaming: Bluetooth, 5G, and next-generation Wi-Fi).

Harman International is using its Harman ExP suite of technologies, whether it is Creator Studio, Drive-Live Concert, or gaming Intense Max, with all the existing connectivity portals to create a "third living space" to bond with friends and family and make driving a shared interactive experience.

Andy Goepfner, who leads audio development for Acura, put it very clearly: "You must create an equivalent or better experience garnered from a mobile device in the car because HMI and other external interfaces are so important to the listener. The question is always being asked: What can we do with audio to make the vehicle experience more



Polestar, a new Swedish automotive brand focused on electric vehicles, owned by Volvo Cars and by extension its parent company Geely, is betting strongly on its premium audio experience, working with high-quality components and the latest technology to design and integrate advanced audio.

luxurious or exciting? It's technology like Acura's ELS STUDIO 3D—that recreates the emotional experience of being in a recording studio—we will continue to build on."

When asked about in-vehicle multichannel networks based on a low latency, bi-directional, daisy-chained data bus, like Analog Devices Automotive Audio Bus (A²B) he agreed that, "These networks are simply, and powerfully, technology enablers. They will be the conduit to bring new entertainment features and functions into tomorrow's vehicle."

Immersive Experiences

OEMs, Tier Ones, and independent contractors agree, "The next big thing is Immersive." There will be more speakers in overheads and headrests



In 2021, Harman introduced its vision and industry support of Experiences per Mile (ExP) concepts—Gaming Intense Max, Creator Studio and Drive-Live Concert—representing a series of new in-car experiences for consumers and new opportunities for automakers.



As the industry leader in the automotive audio space, Harman provides a robust combination of core automotive technologies—including Digital Cockpit, Telematics, Car Audio, Cloud-based Software Services and ADAS.

for 3D. Every carmaker will be “immersive,” but real “immersive” is not well defined. Dolby Atmos is beginning to define it in a standard way. They are creating source material and a reproduction method, which drives architectures. Some OEMs are watching what Sony 360 Reality Audio is doing in automotive. For all immersive solutions, the focus needs to be on content. OEMs might remember the innovative first use of a DVD-A player in a car for Acura, but that format has practically disappeared. Streaming over-the-air (OTA) content or direct download is the key, but, it will start in the luxury automobile market.

The Lucid car manufacturer has already implemented Dolby Atmos. Immersive audio is also shipping in the Lincoln Aviator vehicle with Revel Ultima3D. As the source material becomes more

sophisticated, the application will be appreciated more by the listener.

The differentiation will be signal processing for each individual seat. With that, the challenge is to provide tunings without prototypes. Virtual tuning tools are required. Another effect of 2020 was that virtual tuning became necessary in a lot of cases, but some found it hard to tune a car.

Tools exist for remotely measuring a car audio system, visualizing the data, auralizing it, and tuning it with very close accuracy to the in-vehicle experience. For example, JJR Acoustics uses this method with its own hardware and the Illusonic software tool for automated data collection, tuning, and auralizing. Firmware can be updated remotely, and the tuning process continues as it would normally, except that it is virtual.

Some insiders agree that as the tools become better, then it might be best to virtually tune cars. We can take away the psychoacoustic biases of seeing the interior, smelling the leather, or hearing the door slam, all of which unconsciously influence our opinions. Some characterize virtual tuning as having the same value that mechanical or acoustical simulation does for learning a car. The same way engineers can virtually change mechanical parameters or acoustical damping, an engineer can change tuning parameters to understand the character of the car’s interior and system component interaction. Virtual techniques will need to expand with the use of immersive audio.

In the area of tuning immersive audio, or optimizing the in-cabin sound system for Android Auto in addition to Apple Carplay, some OEMs have built on DSP Concepts’ Audio Weaver platform.

Zoned Audio

With the use of headrest and overhead speakers for immersive audio, it is natural to think about Zoned Audio. Fully controlled and intimately private, Zoned Audio is still considered the “holy grail,” and it is considered on several manufacturers’ “hot track” for development. Originally, the goal was not to have headphones on the kids in the back seat, as fully autonomous vehicles could push that much further. Initially, the zoned approach used existing speaker locations. The first implementation of In-Car Communication (ICC) is a good example.

Bose has led the charge in many areas of automotive audio since it first introduced branded audio at GM in the early 1980s. Zoned Audio through its SeatCentric Experiences could be another early lead for it as well. Its Rear Seat Volume Attenuation focuses the performance of the vehicle sound system on the front row, while rear-seat passengers are



According to Harman’s consumer research, the ability to provide an immersive drive-in or virtual concert experience in the car has never been so valuable. With premium audio, 5G capabilities and cloud services, the car can become anyone’s favorite concert venue.

able to enjoy video content or music from brought-in devices without being disturbed by what is playing in front through the vehicle speakers. And, SeatCentric Call Placement allows passengers to continue enjoying their music while the driver takes a phone call through the UltraNearfield speakers built into the headrest. While the driver can clearly interact with the phone call, the music for the other passengers is never interrupted.

Engine and Vehicle Signature Sounds

With an internal combustion engine, the sound of the car itself as it idles and as it is driven creates an emotional attachment to the driver and is part of creating a comfortable experience. The current goal of automotive audio engineers is to help create emotional attachment. Internal sound synthesis of engine and road sounds have long been used to provide that comfort for combustion engine vehicles. But, with hybrid and electrical cars becoming quieter, the need is absolute, and audio systems will provide the solution. One example is the Ford Mustang Mach-E's use of engine sound synthesis, played through the audio system, which among other things, helps car owners hear when to shift.

Bose uses another approach through its application of Electric Vehicle Sound Enhancement technology, which enables car makers to create unique, unconventional powertrain sounds inside the cabin—delivered through the sound system's speakers. Electric Vehicle Sound Enhancement is part of the Bose Active Sound Management technologies, which was developed with the help of DSP Concepts. This innovation will debut later in 2021 in the GMC Hummer EV.

At least one OEM has created its own version of this with granular synthesis. Sound samples are split into very small pieces (1ms to 50ms) called grains. In playback, multiple grains may be layered on top of each other, and may play at different speeds, phases, volume, and frequency, among other parameters. At low speeds of playback, the result is a soundscape; at high speeds, the result is heard as a note or notes of a unique timbre.

These synthesized sounds, as well as the audio's full musical range, require low frequency to feel natural. For hybrid and electric vehicles, too much space is taken up by batteries, leaving no room for subwoofers. New subwoofer solutions for internal sound synthesis are needed. Subwoofers integrated into body structures, floors, and sills, are no longer possible in electrical vehicles because the battery uses all of the structural space. New ideas are needed, but concepts are slow to evolve. Some solutions follow the approach of Bluacs Technologies' Fresh

Air designs, which are ported outside the car cabin with the entire cabin treated as an extremely large enclosure. Unfortunately, the restrictive points of an opening to the outside are rust, dust, and exterior noise. A conventional 200mm subwoofer would need 20L to operate optimally. The Fresh Air Subwoofer design reduces the overall size by as much as 80%. The trade-off is the upper sub bandwidth, which will be approximately 80Hz and could be challenging if other woofers are difficult to package. There are Fresh Air Woofer designs as well to help solve this.

For many carmakers with the ultimate goal of only electrical cars by the year 2035, it will be easier to reproduce low frequency compared to overcoming the noise of a combustion engine, and they can concentrate on quality of bass instead of the quantity. Wolfram Jaehn, Senior Sound Engineer, AUDI AG, mentioned that "for the past in combustion cars,



British audio pioneer Meridian Audio was chosen by Rivian, the adventurous new electric vehicle maker and technology company, to provide the branded audio system for its electric vehicle platform.

About the Author

Roger Shively is a co-founder and principal of JJR Acoustics (Seattle, WA). He has more than 34 years of experience in engineering research and development, with significant experience in product realization and in launching new products at OEM manufacturers around the world. Before co-founding JJR Acoustics in 2011, Roger worked as Chief Engineer of Acoustic Systems as well as a functional manager for North American and Asian engineering product development teams in the Automotive Division of Harman International Industries Inc; a journey that began in 1986.

Roger received his degree in Acoustical Engineering from Purdue University in 1983, and finished post-graduate work in the field of finite element analysis. He is a member of the Audio Engineering Society (AES), Acoustical Society of America (ASA), and Society of Automotive Engineering. He has published numerous research papers and articles in the areas of transducers, automotive audio, psychoacoustics, and computer modeling. Roger also holds US and International Patents related to the design of advanced acoustic systems and applications particularly in the field of automotive audio. Roger is Co-Chair of the AES Automotive Audio Technical Committee.

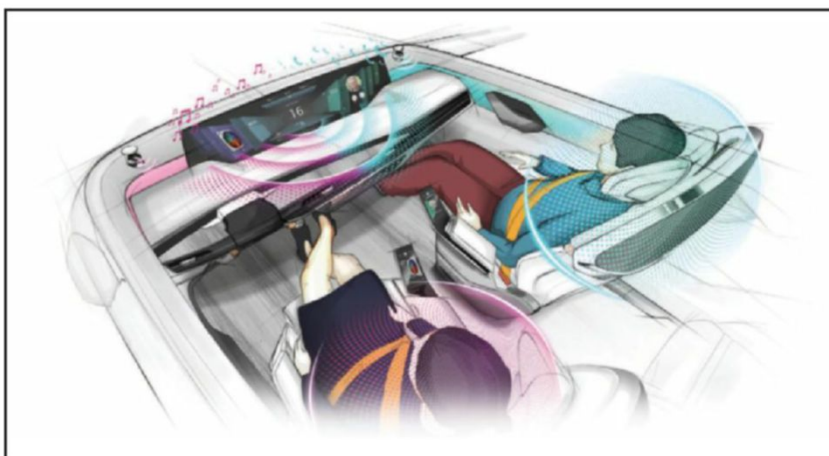


Bluacs Technologies' Fresh Air Speaker technology trades off bandpass function for low-frequency performance, and is applicable to woofer and subwoofer speaker designs in automotive audio. This allows the actual speaker box to be smaller, taking up less cabin space, and be significantly lighter.

we had to compensate for low frequency noise from the engine. This frequency spectrum is missing in electrical cars. Now higher noise such as the pitched electrical whine of engines or wind noise is more audible and we have to compensate for this."

Kevin Doak from Bose comments, "With quieter car designs, mainly driven by the increasing number of electric vehicles coming to market," Tier Ones like Bose, "are supporting OEMs, beyond simply delivering advanced sound systems for great music listening experiences."

Some remaining hybrid engines have languishing booming sounds due to the compact size of the engine design, which ANC can address. But the noise we thought was all engine noise turned out to be road noise: "rough driving surfaces, tire treads, and chassis vibrations. These are factors that produce noise in any type of vehicle, but they are especially noticeable in quiet EVs that do not have the conventional rumble of an engine to mask those annoyances." Bose recently introduced QuietComfort Road Noise Control



Faurecia, the French global automotive supplier and one of largest automotive parts manufacturers in the world is increasingly cooperating with external partners and audio brands to deliver on the vision of its cockpit of the future, which is all about immersive and personalized sound.

as an active and adaptive technology, which is also part of the company's Active Sound Management technologies.

Acoustic Vehicle Alert Systems

Due to the absence of an internal combustion engine, electric vehicles can be much quieter than their gas-powered counterparts. However, this lack of noise can mean that they pose a danger to other road users, especially people who are blind or partially sighted.

In July 2019, a new European Union (EU) rule went into force that all new models of electric cars sold in the EU must make artificial noise under certain conditions. Acoustic Vehicle Alert Systems (AVAS) will need to be installed in new models of hybrid and electric cars introduced from then onward, and all existing models by July 2021. With an AVAS installed, vehicles will need to make a sound while traveling under 12 mph, or while reversing. AVAS has become part of audio system and the audio engineering discipline.

In the US, the National Highway Traffic Safety Administration (NHTSA) requires that all hybrid and electric vehicles emit artificial noise by September 2020, although they'll have to emit the sounds up to the slightly faster speed of 18.6 mph.

The minimum noise standards also define a maximum overall level at a 2m distance of 75dB per driving condition. Next to that, the normal Pass-by Noise standards for maximum noise remain in place, which have slowly been reduced from 82dB in 1972 to 70dB in 2020, with a goal of 68dB in 2024. These contradictory regulations will, no doubt, eventually correct themselves, but they contribute to a complicated existence for carmakers.

Different manufacturers are able to decide exactly what their AVAS will sound like, but the legislations say that the sound should be similar to (and not louder than) a traditional combustion engine. It should also give pedestrians an idea of what the vehicle is doing by, for example, synchronizing with a vehicle's speed.

OEMs and Tier-Ones, with the help of independent contractors, have deployed many AVAS solutions into the automotive market specifically designed to meet each customer's requirements. Some of these deliverables included support for transforming audio bus messages to control signals from wave player models and consultation on appropriate speaker choices and placements in the vehicle.

One of the bigger problems with pedestrian alert systems is that the sound they emit is not heard inside of the car, but it is required to be. So, it is important to make sure the car is not too quiet and

to certify that it still meets the regulations through homologation. Simply increasing the volume of the AVAS unit to pass the standard can lead to annoyed drivers and passengers.

What was once unique in the Nissan Leaf has become more regulated and became a modest commodity, but Tier Ones that backed away from it are now returning due to the growing variety of needs and the potential for added value in external sounds and alerts.

Most EU OEMs are pursuing AVAS so strongly that it has become the hottest thing for their suppliers. The trend is getting more and more external speakers for recreating the engine noise. It had been one speaker box—now it is two in the front and two in the rear. Bigger boxes for entertainment sound, door closing sound, or talking to pedestrians.

Also, a start-up tone or external “Enterprise” sound, so called because of the sound of the Star Trek Enterprise: Whoosh. These small boxes are typically 0.5 liter in size, so that very little bass is possible. Some OEMs are now planning for 3.5L external boxes. And some models, such as the all-electric Porsche Taycan, are bringing that same sound into the interior. Others (e.g., Daimler) are using the external speakers



Porsche is leading in Electric Vehicle Sound Enhancement research, working with DSP Concepts on converting the unique Porsche combustion engine sound into the company's first electrically driven sports car, the Taycan. DSP Concepts' AudioWeaver was used to create a unique Porsche sound signature and streamline the audio development process.

for playing recorded engine noise while under 30kph (18.6mph).

Autonomous Future

Again, for OEMs, using a single component for multiple functions will continue to trend, and an external speaker box plan will exist for autonomous driving to give the pedestrian some clear

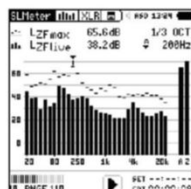
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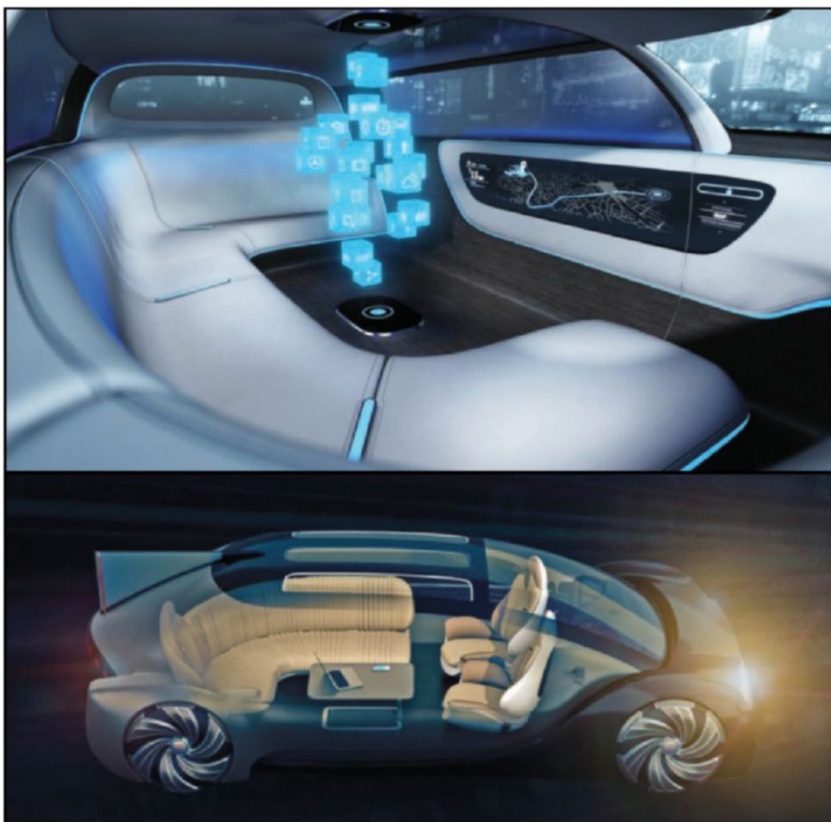


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Autonomous cars are a perfect playground to rethink the audio experience in vehicles, as the cabins are reconfigured and seating positions may change.
(Image concepts—Mercedes-Benz.)

understanding of what is happening, before any eye contact. Yet, autonomous driving—which we thought was closer to fruition, but social distancing might be the biggest thing slowing that down—brings additional questions. Autonomous cars would be the perfect playground to explore individualized space for audio.

In the US, Ford's AV, LLC, among others, are plugging away and research is ongoing. The approach is not only treating the development as a new product but as a service for humanity: Transportation as a Service (TaaS), but not just for delivering pizzas. This approach is common across carmakers.

Many Tier One companies are expecting the advancement of driverless vehicles and anticipate changes in the way vehicle cabins are configured. Seating positions might change; perhaps a traditional driver's seat will no longer be needed. These changes will impact the way people experience and consume entertainment content in the vehicle, as well as how they communicate with others.

"To support this evolution," says Kevin Doak, "Bose continues to research and invest in UltraNearfield speakers that are built into the headrest (similar to those found in Bose systems available today, such as the Personal Plus system in the Nissan Kicks). UltraNearfield

speakers, combined with advanced algorithms and digital signal processing, can help maintain the relationship between passengers and audio, even as seating locations change in future vehicles."

Other technologies originally developed for traditional human driving are in development, which can render sounds from different directions and help drivers quickly react and respond to various vehicle signals. The blind spot detector warning could come from the direction the driver needs to look in order to react. Rendering GPS navigation prompts from different places virtually, left or right, near or far, depending on direction and urgency would be a part of this future use. The system might also be able to effectively warn a driver whose vehicle is in semi-autonomous mode that there's an exit coming up and the driver will have to re-engage the vehicle to reach the exit ramp.

The Bose Aware Signal version of this system in autonomous and semi-autonomous vehicles includes the ability to dynamically apply audio tags to external obstacles and to render directional sounds that will reassure passengers that the car is aware of its environment (e.g., a bicycle coming up on the right or a stopped vehicle on the road ahead).

The big picture signals that the car might become a mobile private living room. Music will remain the most important part. But, telemetry information will be equally important. The use of artificial intelligence, not just to tailor the environment for the users, but to individually modify their sound experience, will be just as valuable. If we can move around inside a car, where we listen would no longer be a fixed position. The sound would need to change with movement in the cabin—meaning we would need a tracking system to make changes in real time.

Harman International looks to the next great evolution to be in the reimagination of the experiences that take place in the car. It calls this "the shift from RPM to EPM—moving away from measuring the impact of mobility as revolutions per minute to experiences per mile."

Whether or not there will be an autonomous revolution in the near future, there will still be the bespoke tailoring of sound to the individual listener. This is the highest value for the listener. We are already providing audio tunings to balance the sound quality in a car, and now we are working on tunings for each particular occupant/listener.

In the end, with electric cars, gas stations will no longer be needed nor will electrical pump stations. But there will still be a need to stop and rest, so maybe there will be well-being stations. And with various levels of autonomous driving, the well-being will begin in the car.

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

Features

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

Applications

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

THMD

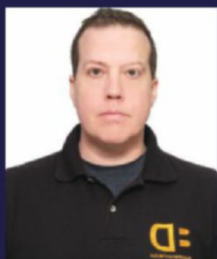
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Vehicle Electrification and Active Noise Cancellation

The Driving Forces Behind Innovation in Future In-Vehicle Infotainment (IVI) Systems

This article examines noise sources in automobiles and the various noise cancellation technologies to better understand the future of sound management in car audio systems.

By
Krunal Maniar
(Silicon Labs)

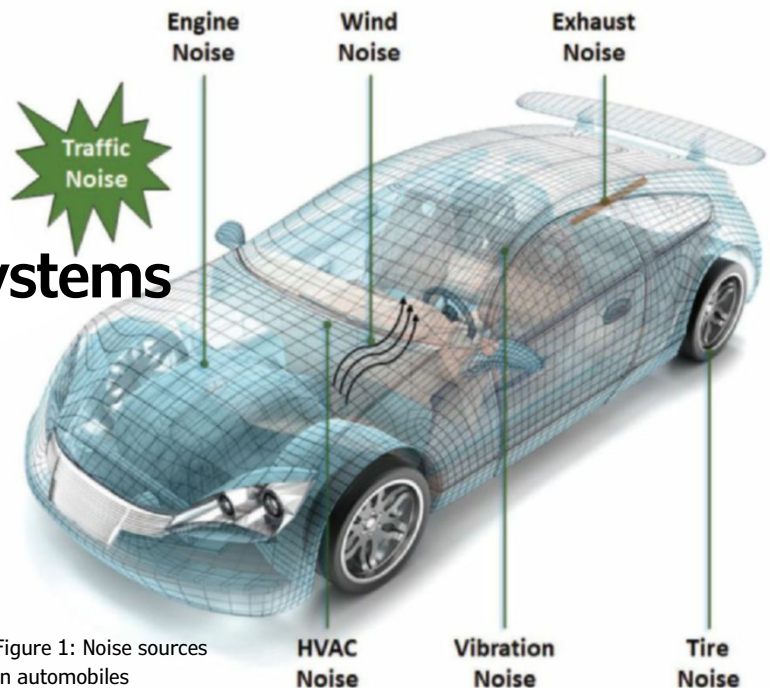


Figure 1: Noise sources in automobiles

The automotive industry has widely acknowledged vehicle electrification as an effective way to improve fuel efficiency and reduce its carbon footprint. While it is true that Hybrid Electric Vehicles (HEVs) use a combination of the traditional Internal Combustion Engine (ICE) and one or more electric motors for propulsion, Electric Vehicles (EVs) exclusively use electric motors. With significantly different acoustic environments inside their cabin interiors, car manufacturers have a strong drive to innovate in noise cancellation and other car audio technologies. Let us examine noise sources in automobiles and talk about the various noise cancellation technologies to better understand the future of sound management in car audio systems.

Understanding Noise Sources in Automobiles

The passengers inside the cabin are subjected to several external noise sources such as road-tire noise and wind noise and several internal noise sources such as engine noise, HVAC noise, exhaust noise, vibration noise, etc. The relative contribution of these noise sources to the total noise depends on several factors, such as the vehicle's speed and acceleration conditions, the quality of roads and tires, the order and stroke of the engine, and the vehicle's aerodynamics, among others (**Figure 1**).

The engine noise in a typical ICE is dominated by the piston crank mechanism and combustion noise. Engine noise is observed at all speeds and depending on the type and stroke of the engine, revolutions per minute (RPM), acceleration-deacceleration conditions, etc., the noise profile will differ. Road-tire noise is generated by the interaction between the road surface and tire

and is dependent on the tire tread pattern and physical profile of the road surface. The road-tire noise is pre-dominant at mid-speeds, between 30mph to 60mph. The wind noise is induced by the aerodynamics of the vehicle that generates turbulence and aerodynamic drag. The wind noise becomes a primary contributor at higher speeds, typically over 60mph. As a combination of all noise sources, the total noise spectrum can be represented, as shown in **Figure 2**. These noise sources have most of the acoustic energy concentrated within the frequency band of 20Hz to 1000kHz.

In HEVs, the engine's noise contribution to total noise is relatively small compared to traditional IEC vehicles. In EVs, the traditional engine noise does not exist. The high-frequency noise from electric motors in EVs have relatively much lower acoustic energy and can be even further attenuated by careful design of transmission loss and acoustic absorption techniques. However, the significant reduction in engine noise in HEV/EV is a double-edged sword (i.e., the noise from the other noise sources become easily perceivable to human ears). With a conservative estimate of 6dB background noise reduction because of the absence of ICE in EVs, the other noise sources stand out and can get annoying to human ears. Also, as HEV/EVs have lower background noise, the car manufacturers get a perfect opportunity to provide an even better acoustic experience inside the car by improving their other audio systems without adding significant cost.

Introduction to Passive Noise Cancellation (PNC)

Passive noise cancellation (PNC) uses noise-isolating materials to absorb and dampen the acoustic energy of the undesired

sound wave (**Figure 3**). In this approach, structural dampening materials such as fiberglass and cell foams are sandwiched in-between car body parts. Special laminated windshield and window glasses, door sound barriers and seals, special dash, floor, roof mats, and carefully designed muffler systems—all dampen the sound waves before they reach inside the cabin.

Introduction to Active Noise Cancellation (ANC)

Active Noise Cancellation (ANC) is a relatively newer approach to reduce background noise inside cabin interiors. As shown in **Figure 4**, an ANC system produces an “anti-noise” signal that destructively interferes with the original noise to remove the background noise.

The ANC system needs external power and uses several transducers (microphones and loudspeakers) and semiconductor components. With modern infotainment systems with built-in loudspeakers for playing audio, the ANC system leverage a significant portion of the available audio system and hence comes at a relatively low-cost adder. As shown in **Figure 5**, an ANC system works by measuring the constantly changing noise inside the cabin using strategically placed microphones throughout the interior cabin of a vehicle. These microphones measure the interior cabin noise and send it to an audio subsystem. This audio subsystem processes the measured signals in real-time using advanced Digital Signal Processing (DSP) techniques with adaptive algorithms to provide an anti-noise signal output to the built-in car loudspeakers. These loudspeakers generate an out-of-phase acoustic field that destructively cancels out the original field and removes background noise.

Comparison Between ANC and PNC

ANC is more effective at frequencies less than 1kHz, whereas PNC is more effective at frequencies greater than 800Hz. The effective frequency regions and the noise attenuation achievable by ANC and PNC approach is shown in **Figure 6**. While PNC is effective for reducing high-frequency noises, ANC is most effective for low-frequency noise reduction. In addition, ANC offers better background noise reduction compared to the PNC method at low frequencies.

Noise sources such as wind noise, engine noise, and HVAC noise have most of their acoustic energy concentrated at < 1kHz. They emit both tonal and broadband noise. The tonal noise characterized by spectral tones is usually annoying to human ears and induces the most fatigue to passengers. The PNC techniques cannot target specific tones in the spectrum without seriously compromising weight,

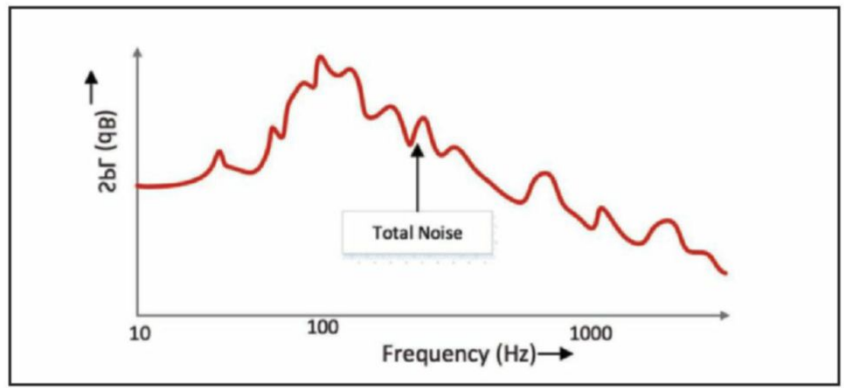


Figure 2: Example of noise spectrum inside vehicles

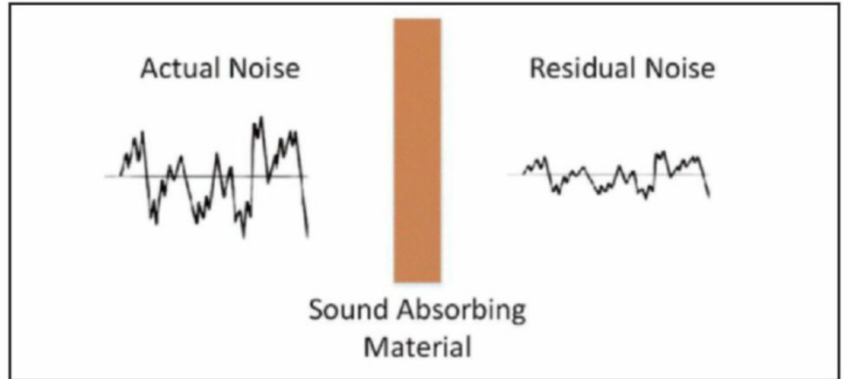


Figure 3: Concept of the PNC system

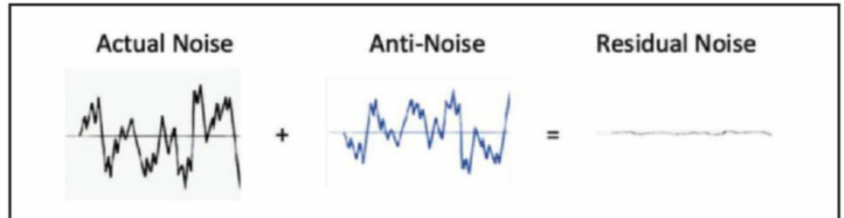


Figure 4: Concept of the ANC system

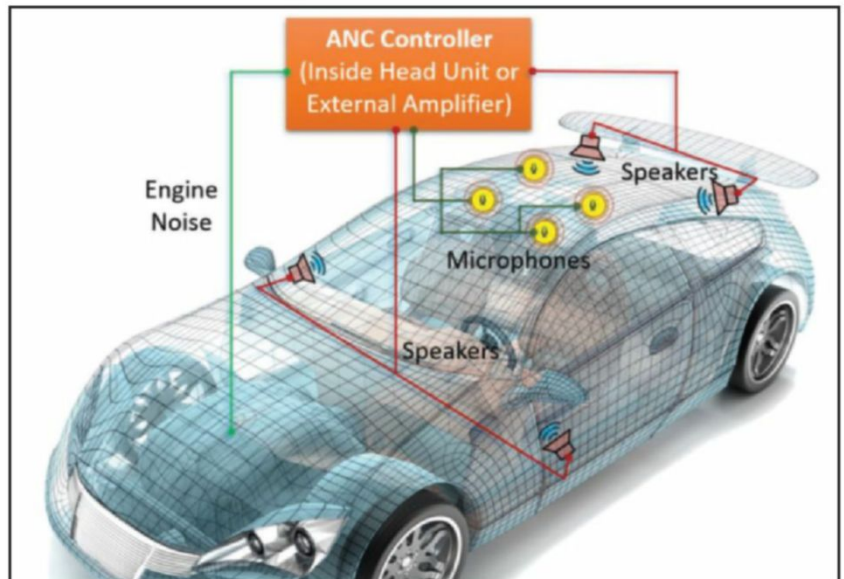


Figure 5: ANC implementation in automobiles

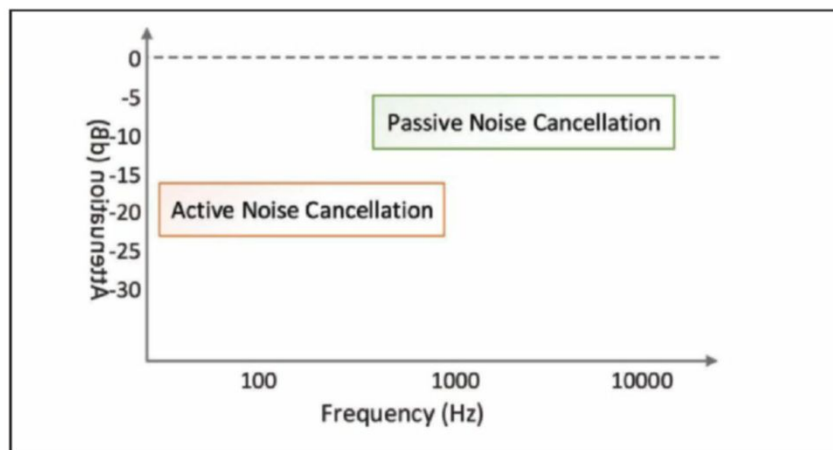


Figure 6: Effective frequency regions for ANC and PNC methods

size, and cost. On the contrary, ANC can offer targeted tonal noise reduction and broadband noise reduction by DSP implementation using adaptive notch filters.

With the advent of HEVs and EVs, as the engine gets less powerful (in the case of HEVs) or is eliminated completely (in the case of EVs), the road and wind noise become even more noticeable. As car manufacturers emphasize lightweight designs to meet ever-increasing fuel economy standards, the ANC method that does not compromise on weight has a very strong value proposition compared to PNC to attenuate the low-frequency tonal and background noise. PNC implementation can get bulky and complex to achieve the same. A typical mid-sized car has more than 100 lbs. of insulating material, and hence any weight reduction using the ANC system is highly appreciated by car manufacturers.

One of the major advantages of PNC is that the sound isolating materials used have multiple purposes

(i.e., they not only attenuate high-frequency traffic noise but also help reduce car vibrations and improve thermal insulation). Hence, PNC method works complimentary to the ANC system and cannot be eliminated.

Bass band frequencies typically range from 20Hz to 250Hz, and mid-band frequencies typically range from 250Hz and over. Since ANC are the most effective in these bass-band and mid-band frequencies, a superior audio music playback experience inside the cars with ANC systems can be expected. The key differences between ANC and PNC techniques are shown in **Table 1**.

Future of Sound Management in Car Audio Systems

ANC enhances the driving and in-cabin passenger experience to a different level, resulting in reduced fatigue and improved comfort. ANC allows for quieter cabins that enable clear communication among vehicle occupants, referred to as in-cabin communication (ICC), and high-fidelity hands-free voice calling using beam-forming technologies.

Because of ANC, quieter cabins also drive car manufacturers to offer even better audio music playback experience (e.g., true 3D surround sound).

Engine Sound Enhancement (ESE), especially observed in small fuel-efficient HEV/EV cars, generate artificial vehicle engine and exhaust sounds through the same audio system. These ESE systems in the absence of “powerful” engines enhances certain engine tones typically induced by intake and exhaust systems. ANC and ESE systems work simultaneously to create super-quiet interiors while simultaneously offering a much sportier driving experience. The

Parameter	Active Noise Cancellation (ANC)	Passive Noise Cancellation (PNC)
Suitable Frequency Range	< 1000 Hz	> 800 Hz
Noise Sources where most effective	Wind, engine, road-tire, HVAC, etc.	High-frequency traffic noise sources such as honks, sirens, etc.
Tonal Noise Reduction	Excellent, targeted reduction possible with advanced DSP algorithms	Average, with high compromise on weight and complexity
Broadband Noise Reduction	Suitable for < 1000 Hz	Suitable for > 800 Hz
Implementation	Complex, need to solve a lot of hardware and software design challenges to achieve good performance	Relatively easy, it's a traditional solution used in automobiles
Weight Addition	Not much	Bulky, for low-frequency noise reduction
Customization across different noise profiles and requirements	Easy. Once implemented, no hardware modifications needed. Fine-tuning the software is usually good enough	Complicated, as it involves complete re-design as per new noise profile and reduction requirement
Thermal Insulation Improvement	No	Yes, the noise isolation materials provide thermal insulation as well
Mechanical Car Vibrations	No	Yes, the noise insulation material can dampen vibrations as well
Power Requirements	Additional power is required for transducers and semiconductor components	No additional power required

Table 1: Comparison between ANC and PNC methods

different drive selection modes such as Eco, Normal, Sport, etc. in HEV/EVs offer specific acoustic in-cabin signatures using ESE and ANC systems.

HEV/EV cars are quieter at low speeds and can pose additional risks to pedestrians and cyclists. Acoustic Vehicle Alerting Systems (AVAS), now slowly becoming a mandatory feature worldwide, is designed to create artificial sounds to alert pedestrians on the road.

Silicon Labs Si479xx automotive radio receivers integrate an audio subsystem with audio DSP, analog-to-digital converters (ADCs), and digital-to-analog converters (DACs). In addition, Silicon Labs provides a software development kit (SDK) that simplifies porting of audio algorithms to the audio subsystem. This high-level of integration offers a greater flexibility to automotive radio suppliers to port audio algorithms for ANC, ESE and other audio post-processing applications.

For additional product portfolio information, please visit www.silabs.com/audio-and-radio.

As vehicle electrification and ANC trends pick momentum, there will be increased adoption of advanced sound and audio technologies in IVI systems.

Figure 7 shows some of the potential future sound management ecosystem that could be available .

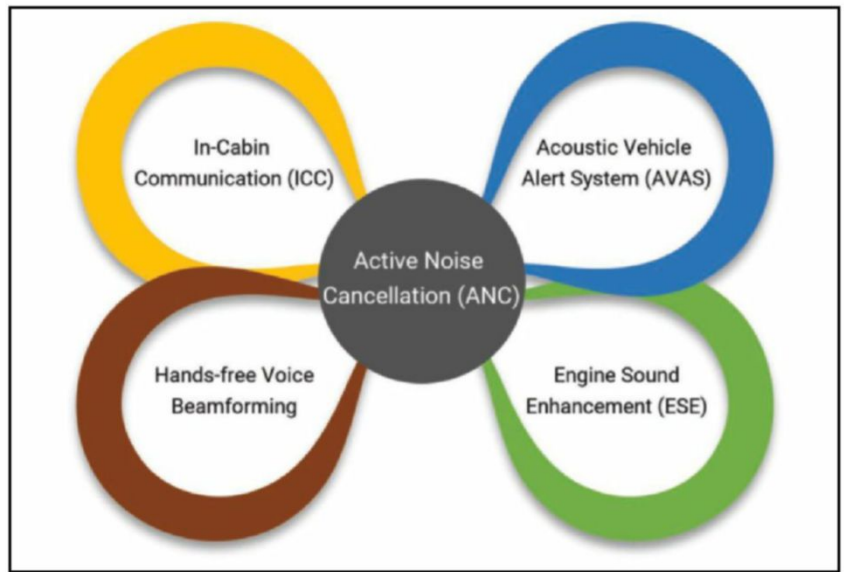


Figure 7: Future sound management ecosystem in automobiles

About the Author

Krunal Maniar is the Senior Product Marketing Manager for Broadcast Automotive Products at Silicon Labs. Krunal brings over 10 years of multi-disciplinary automotive engineering and business expertise, spanning marketing, product definition, and validation at leading technology companies. He earned his Master's in Electrical engineering from Arizona State University.



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Using Line Arrays to Control Directivity

In this Sound Control article, our audio expert details how arrays, if used properly, can be a very useful tool in controlling the directivity of a speaker system.

By
Richard Honeycutt

Whenever two or more loudspeakers' radiation covers the same area, the composite radiation pattern depends upon not just the patterns of each speaker, but also the geometrical relationship of the speakers. Or, to say it another way, two or more loudspeakers radiating the same program material constitute an array. This is true even if there was no intention to create an array.

An example from my own experience is a sound system designed and installed by a colleague of mine. He did not think that a single subwoofer would produce loud enough bass for the venue, so he used two subs—one located on

each side of the stage. When the system was powered up, the operator immediately cut the sub level to the point that the bass in most of the venue sounded anemic. The reason he did so was that the two subs formed an array that produced far too much bass in the longitudinal center of the venue, where the mixer was located. So the operator tried to bring the bass into balance by cutting the sub level. Naturally, the result was insufficient bass at the sides—or anywhere except the central axis. But arrays, if used properly, can be a very useful tool in controlling the directivity of a speaker system.

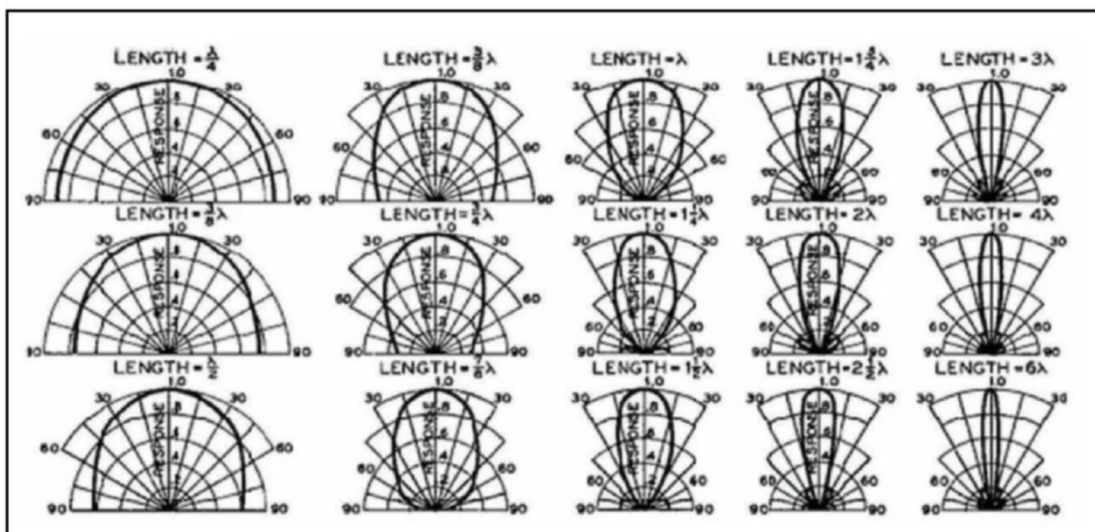


Figure 1: The directivity of a line array depends upon the length of the array.

Arrays

Every few years, the audio world rediscovers a principle that was investigated and usually patented in or before the 1940s, proclaims it to be a great breakthrough, and may even patent it again. One such principle is that of the line array. A true line array is a continuous strip radiator that may be oriented either horizontally or vertically, and which produces directional control through interference effects among separate portions of the array. Only two true arrays are presently available, to my knowledge: the electrostatic strip radiator, and the electromagnetic strip radiators sold by several companies. Numerous manufacturers produce stacks of boxes that are meant to act more or less as line arrays, but these are not true line arrays.

A true line array radiates in phase all along its length. A close approximation to a true line array can be achieved if individual radiators are fed in phase, and the centers of adjacent units are within one-quarter wavelength of each other at the highest frequencies. Arrays that do not meet this requirement will exhibit irregular frequency response and lobing in the radiation pattern.

Figure 1 [1] shows the directional characteristics of a true line array. (The curves are calibrated in fractions: 0.8, 0.6, 0.4, and 0.2 times the on-axis SPL, corresponding approximately to -2dB, -4.4dB, -8dB, and -14dB.) Notice that in addition to the major lobe, at lengths greater than or equal to one wavelength, minor lobes also appear. (Note the patterns for array lengths greater than 1-1/2 wavelengths.) Listeners at the angles corresponding to the lobes will experience extremely irregular frequency response. However, also note that the levels of the minor lobes are greatly reduced when compared to the major lobes. The approximate equation for the angular width of vertical coverage for a vertical line array is $\theta = 2 \sin^{-1}(\lambda / L)$. [2] (The angle will be expressed in radians.)

The first common commercial application of line arrays was the “sound columns” produced by Bogen, University, Argos, and other manufacturers prior to 1970 (**Photo 1**). Most sound columns used from four to seven identical cone speakers—typically 4” to 8” diameter each—spaced anywhere from 6” to 10” center-to-center. This would mean that the critical $\lambda/4$ minimum spacing would apply only up to about 340Hz to 570Hz.

Shure introduced the Vocal Master column in 1967 (**Photo 2**). Incorporating two 10” and four 8” cone drivers, this column speaker outperformed the less expensive competitors. All the sound column

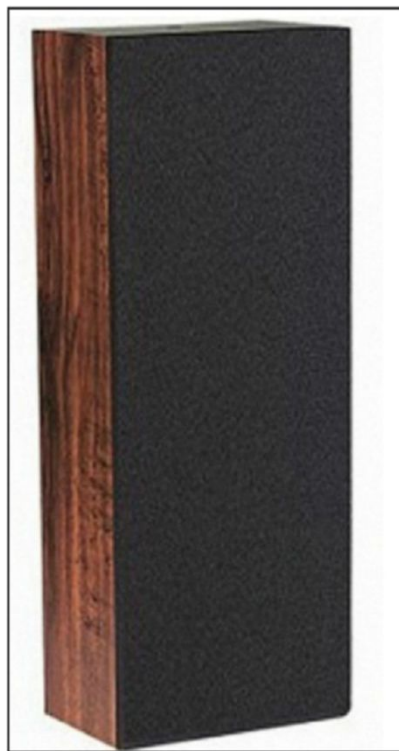


Photo 1: This Bogen “sound column” speaker was used in many school and community auditoria before 1970.

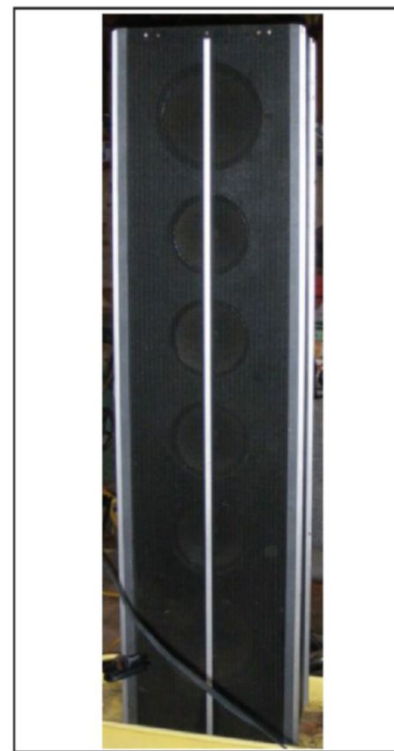


Photo 2: The Shure Vocal Master was an improved line array compared to the earlier “sound columns.”



Photo 3: This JBL modular line array was designed to provide concert sound for large groups, with good vertical control.

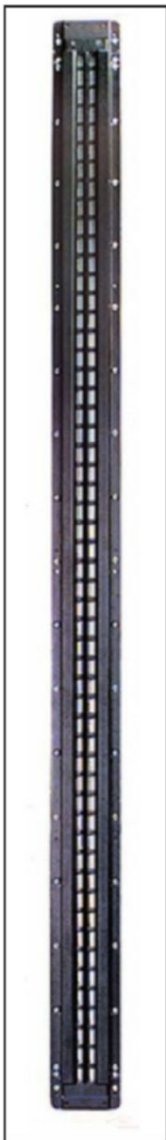


Photo 4: The Bohlender-Graebner ribbon line array was available in four lengths: 28", 40", 50", and 75".

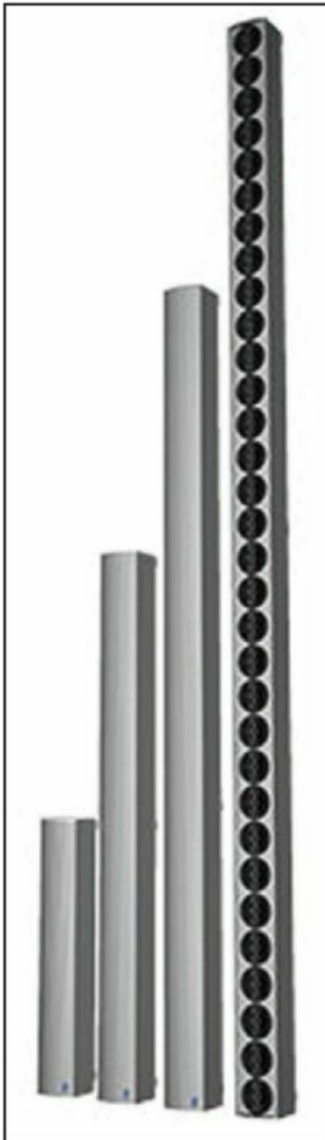


Photo 6: The Renkus-Heinz Iconyx steerable line array was introduced in four lengths.

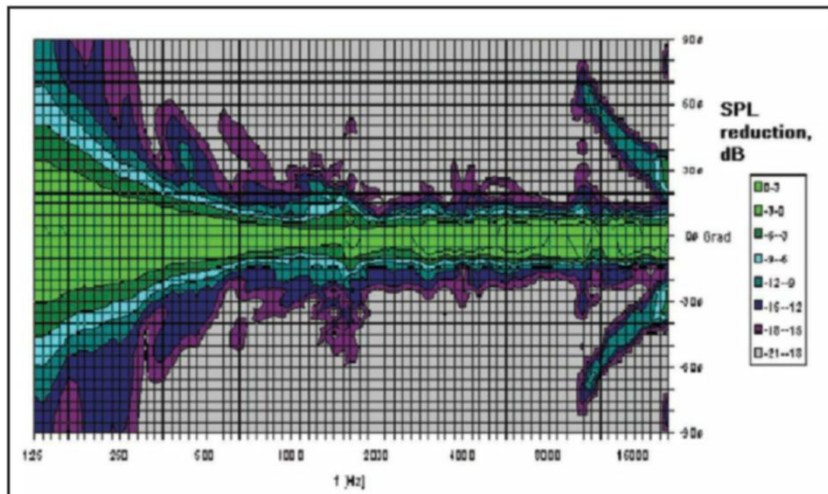


Photo 5: The vertical directivity of the 75" Bohlender-Graebner ribbon line array was significant above 125Hz.

speakers provided some vertical directivity control, but the horizontal directivity was only limited by the size of the cone drivers, larger drivers providing tighter horizontal directivity. The Vocal Master's specified directivity was $65^\circ \text{V} \times 140^\circ \text{H}$, although the frequency at which this directivity was achieved was not specified. Any line array's vertical directivity is limited by the array length, with useful directional control commencing about the frequency whose wavelength corresponds to about $\lambda/4$, as shown in Figure 1. Thus a typical PA sound column, being no more than 4' high, would provide some vertical control down to about 80Hz. Achieving an off-axis response 4.4dB down from the on-axis response would be possible only down to about 283Hz. The Vocal Master was about 5' tall, and thus provided vertical control to a higher frequency: the specified 65° vertical directivity would have been achieved at and frequencies above starting slightly over 1kHz.

In the 1950s and 1960s, Bill Hanley often covered large live-sound venues by stacking Altec A7s in order to create a sort of line array to control the vertical directivity of the speaker system. But factory-made concert line array systems were not to emerge for another 40 to 50 years. In 2003, Marcel Urban, Christian Heil, and Paul Bauman of L'Acoustics published a paper describing "Wavefront Sculpture Technology," a design technique to improve the evenness of coverage of a line array speaker.[3] This technology was incorporated in the L'Acoustics V-Dosc modular line arrays, and other manufacturers began building modular line arrays based upon different embodiments of the same principle (**Photo 3**).

Differing Approaches

Taking a different approach, Bohlender-Graebner manufactured a line array using a single electromagnetic ribbon element, avoiding the need to coordinate phase relationships among many different drivers (**Photo 4**). Although there was some lobing (at levels about 12dB below the axial level), these arrays performed quite well and had an extremely natural sound (**Photo 5**). However, their power handling limitation did not allow their use in high-SPL applications.

In the early 2000s, several companies introduced powered line arrays whose drivers had separate onboard power amplifiers, allowing the frequency response and phase of each driver to be preset separately. This, in turn, allowed each array to have one or more steerable beams to cover complex audience areas, such as orchestra

and balcony seating. **Photo 6** shows four models of one such steerable line array: the Renkus-Heinz Iconyx. The drivers were small cone units with coaxial dome tweeters at first, then later models used horn tweeters in order to provide higher maximum SPL. **Photo 7** shows a cross-section of the vertical coverage of an Iconyx line array configured to provide one beam. The heavy straight black lines indicate head height of the main and balcony seating areas. The curved lines above the seating areas indicate the SPL at the indicated location. Note that the coverage angle decreases as frequency goes up, and that although there is some lobing in all three plots, the lobing is the worst at the highest frequencies, due to the driver spacing being a larger percentage of the wavelength at those frequencies. (All three plots cover a three-octave range, starting at the stated frequency (250Hz, 1kHz, or 8kHz).

The horizontal directivity of a line array depends upon the horizontal dimensions of the radiating elements. In some line arrays, these are cone speakers, perhaps with coaxial dome tweeters; in others, cone woofers with compression-driven mid-high frequency drivers. In the latter, the compression drivers may feed an elaborate piece of “plumbing” (waveguide) that converts the sound wavefront to a specific shape to control directivity. Some line array modules have mid-high horns with movable sides to allow adjustment of the horizontal directivity. When the mid-high drivers are cone or dome tweeters, their directivity increases with frequency. However, most line arrays have horizontal coverage angles between 90° and 150°.

If more than one line array is used in a venue, the wide horizontal coverage of most line arrays can cause problems, especially in long, narrow venues, because of many listener seats being covered by more than one speaker. **Photo 8** shows an auditorium covered by a single line array, and **Photo 9** shows the same auditorium covered by two arrays. Note the overlap in coverage when two arrays are used.

Although covering a single seat from two or more line arrays will cause some comb filtering, the large amount of sound reflecting from the sidewalls tends to mask this effect. As long as the distance from any array to any seat is no more than 40' (50-ms delay) greater than the distance from another array to that seat, covering a seat from multiple arrays is usually not a problem. If the speaker-to-seat distances are greater than 40', the Haas effect will no longer cause

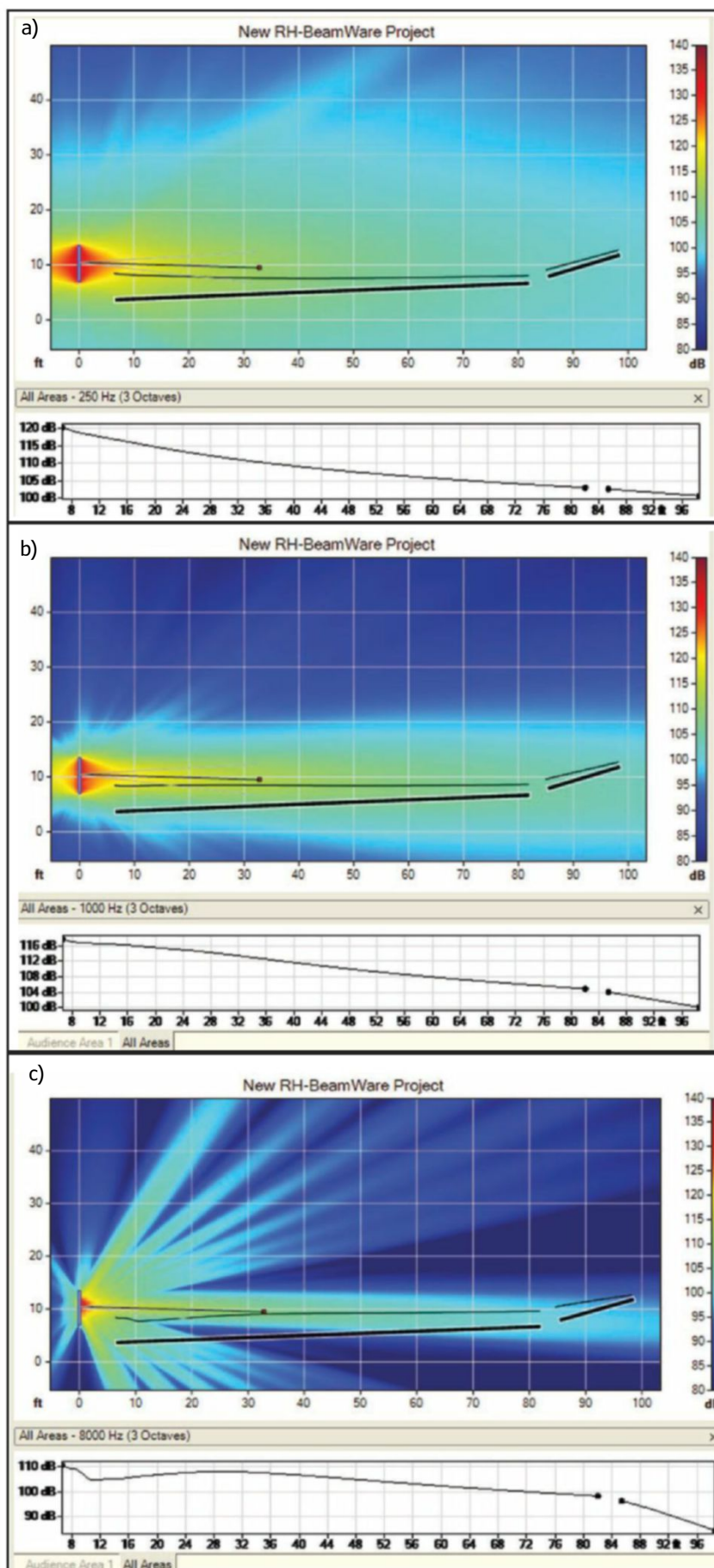


Photo 7: The vertical coverage of a steerable array can be aimed. (The top plot (a) is for 250Hz; the middle (b), 1kHz; and the bottom (c), 8kHz.)

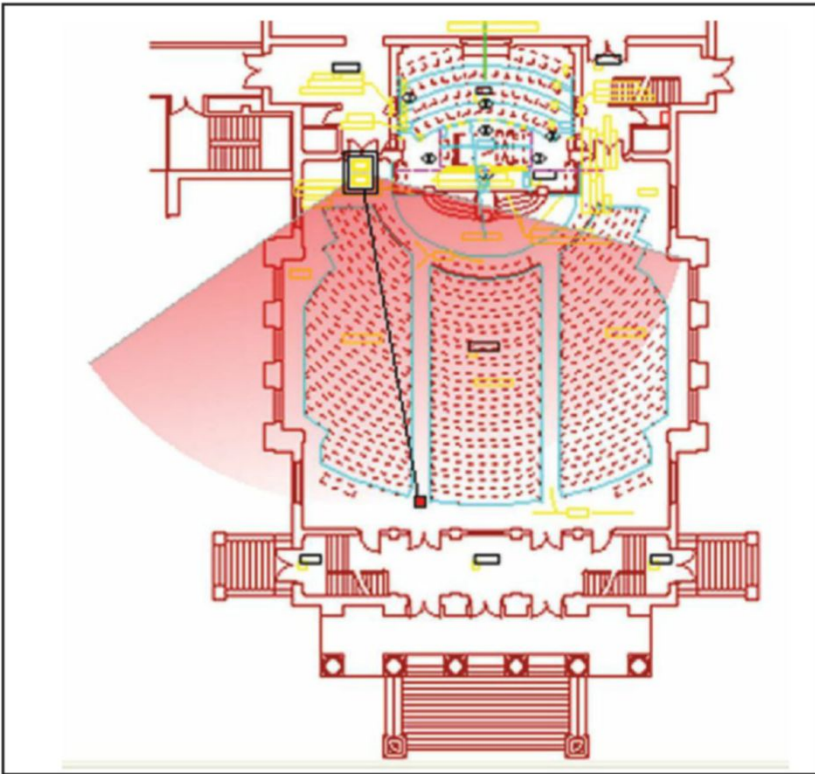


Photo 8: Due to the wide horizontal coverage of a line array, often a single array will cover a venue.

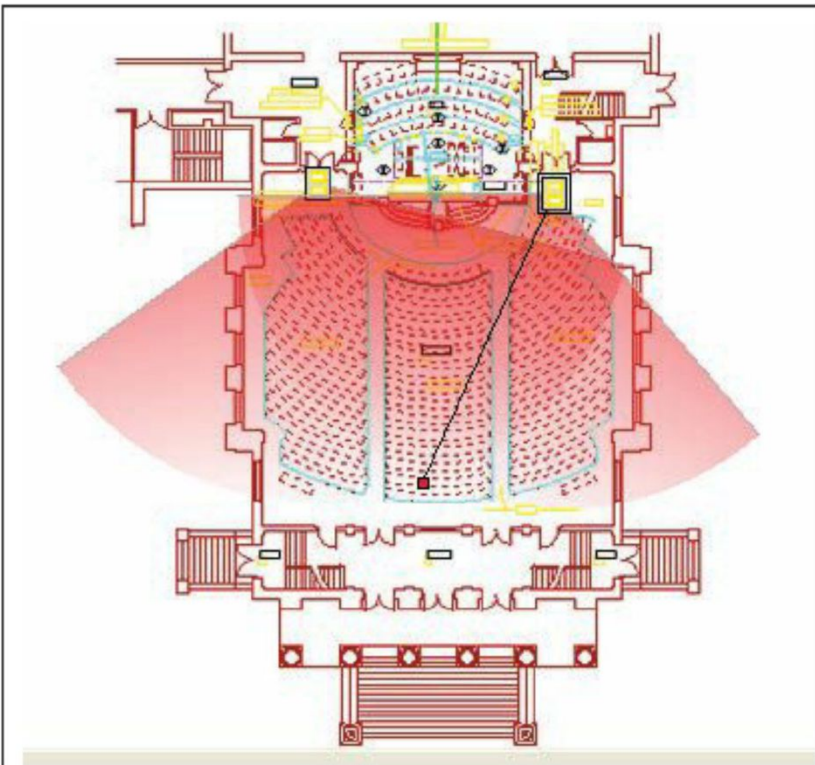


Photo 9: When two line arrays are used, a listener may hear some comb filtering—or in the worst case, echo—due to differences in arrival time of the sound from the different speakers.

the sound from the separate speakers to merge into a single acoustic event, and the listener will hear an echo.

A New Variant

In the year 2000, Don B. Keele, Jr. published a paper detailing the design of a new variant of the line array: the Constant Beamwidth Transducer (CBT).[4]



Photo 10: The JBL CBT1000 is one line array built on the CBT concept.

References

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- [3] M. Urban, C. Heil, and P. Bauman, "Wavefront Sculpture Technology," *Journal of the Audio Engineering Society (JAES)*, Volume 51, Number 10, pp. 912-932, October 2003.
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Photo 11: Inventor Don Keele shows a CBT array made for location on a floor (ground-plane use). There is also a model designed to be suspended. These are available as kits.

Based upon unclassified US Naval research, the CBT concept uses multiple drivers evenly placed around a circular arc.

A CBT array provides constant directivity and requires no DSP, since only the levels fed to the drivers are “shaded” (gradually decreased) as you move from the center to the end of the array; no delay is added to any driver. The coverage pattern is independent of frequency and distance, avoiding the near-field level boost common to other line array designs. JBL manufactures a line of CBT array speakers (**Photo 10**), and Don Keele has designed a CBT array kit for DIYers (**Photo 11**). More information can be found at his CBT page (www.xlrtechs.com/dbkeele.com/CBT.php).

Although no specific loudspeaker design is optimum for all applications, line arrays have significantly evolved since their first analysis by Harry Olson at RCA Laboratories more than six decades ago, and are now a vital tool in the sound-system designer’s toolbox! 

About the Author

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



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KEF LS50 Meta Loudspeaker

A Material Improvement

In September 2020, British brand KEF updated its extremely popular LS50 speaker design and introduced the LS50 Meta, an improved version of the original LS50 passive design, now featuring a new Uni-Q driver that is coupled to an interesting new metamaterial absorber in the back. *audioXpress* requested a pair of the loudspeakers for review. Oliver Masciarotte, Kent Peterson, and their friend Pete Quirsfeld did the listening, and Kent Peterson confirmed their impressions with extensive measurements.



The LS50 Collection update has allowed KEF to introduce the results of its first collaboration with Acoustic Metamaterials Group (AMG), a young company from Hong Kong researching the usage of metamaterials in noise control solutions. The company was set up in 2012 by three young entrepreneurs—Yang Min, Vincent Fong, and Chen Shuyu—who are in charge of research and development, business development, and manufacturing, respectively.

By
Oliver Masciarotte

The original coaxial LS50, with its trademark Tangerine waveguide, was released for KEF's 50th anniversary in 2011. In 2012—seemingly a lifetime ago—I first heard the then-new model at the Rocky Mountain Audio Fest. Superficially, it looked to have a single, full-range driver with an attractive, copper-colored cone. As we now know, it was a two-way coaxial design, borrowing aspects from KEF's flagship Blade. I remember listening and thinking that it gave a memorable performance, even at its very affordable price. Writing up that show as my first article for *audioXpress*, I mentioned that "I got the briefest listen to some surprising speakers, like KEF's LS50 (loved it a lot...)." Since that time, I've continued to hold the passive LS50 in high esteem as a performance benchmark in its price range.

Fast forward to October of 2020, and a press release arriving in my email. The press release announced a new iteration of the LS50, this time with something novel to the industry; the inclusion of metamaterial. In brief, metamaterials are non-traditional physical materials and structures

that are used to control the propagation of waves through and around them. Metamaterials counterintuitively manage to bend the rules of received wisdom, by cleverly manipulating energy in novel and surprising ways.

In the case of the LS50 Meta, the metamaterial takes the form of a quite small "unit-cell" structure repeated and nested in two dimensions to create a larger, disk-shaped region of material having highly optimized acoustical absorption properties. The disk, looking something like a diminutive labyrinth for insects, sits on an acoustically reflecting substrate. The structure, according to GP Acoustics* designers Sebastien Degraeve and Jack Oclew-Brown, is "...a continuum of acoustical resonators, tuned to have a constant number of resonances per octave above a cut-off frequency." The maze-like metamaterial is fabricated from injection-molded polymer. Since the metamaterial addition is the primary change in the new LS50, I will spend a fair amount of time discussing its construction and use in KEF's latest iteration.

I asked Oclew-Brown, KEF's Head of Acoustics,

KEF
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what the magical new feature affords. By email, he told me that the LS50 Meta's new Metamaterial Absorption Technology (MAT) is a metamaterial disk affixed to the rear of the Uni-Q driver. "This disk absorbs up to 99% of the back wave from 620Hz (on up). The result is detail and crispness at a completely new level of performance."

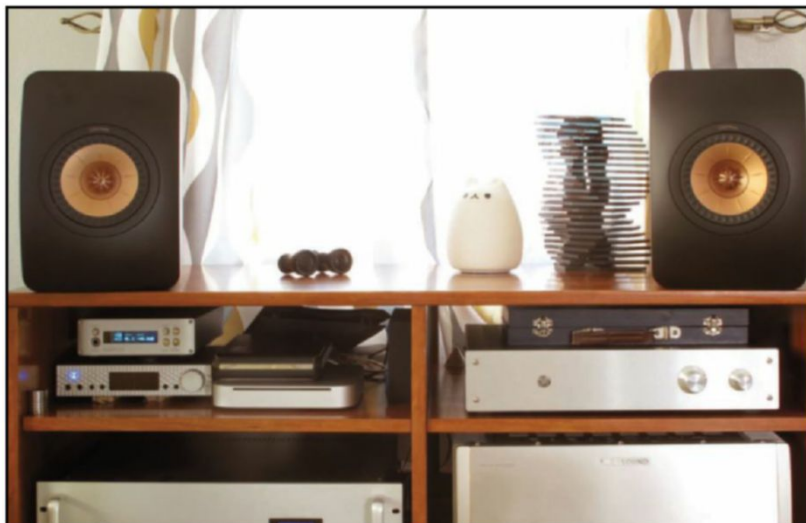
The Audio Engineering Society (AES) Paper 10341, "Metamaterial Absorber for Loudspeaker Enclosures," mentions that, "Since the resonators are very high Q, the absorption approaches 100% close to the resonance frequencies. However, the absorption drops to 50% to 80% at the anti-resonances because the spectral density is very low."

An absorbing material (sponge) is added in front of the metamaterial to create an "absorption valley-filling" effect due to the surface impedance renormalization by the evanescent waves, and their interaction with a highly dissipative medium. The production version is the same as that described in the AES paper. The metamaterial is dual layer, two labyrinths of channels stacked one on the other, with a total of 30 channels. That disc-shaped metamaterial structure then has 1cm of open-cell foam, the "sponge," applied to its face, and the entire assembly is attached to a conical duct. The metamaterial replaces a progressively damped, exponentially tapered horn, applied to the rear of the tweeter to load it and significantly and optimally reduce the energy being launched into the cabinet. Oclée-Brown told me that the original LS50 used their patented vented tweeter design to control the back wave, but the MAT "...is an entirely new level of absorption for the unwanted back firing energy."

At Play

A quick speeds-and-feeds check yields the following: a Uni-Q concentric driver array featuring a 1" (25mm) vented aluminum dome tweeter with MAT, and a 5.25" (130mm) cone, also aluminum, for midrange and low-frequency duties. The crossover frequency is 2.1kHz and the sensitivity is spec'd as 85dB (2.83V/1m, A-weighted). Each 17.2lb. (7.8kg) speaker is nominally 8Ω though the minimum impedance is 3.68Ω at 217Hz, so thoughtful amplifier selection is indicated. The flared racetrack port is positioned on the back, and a nested, two-stage foam plug is included. A single set of five-way binding posts is supplied for hookup, and threaded inserts are positioned on the bottom for use with KEF's \$450 fillable S2 stand. The stand is available in the same matte finishes as the speaker; dark gray, black, white, and a very dark "royal blue."

I started my subjective evaluation by pressing the exceptional Cambridge Acoustics'



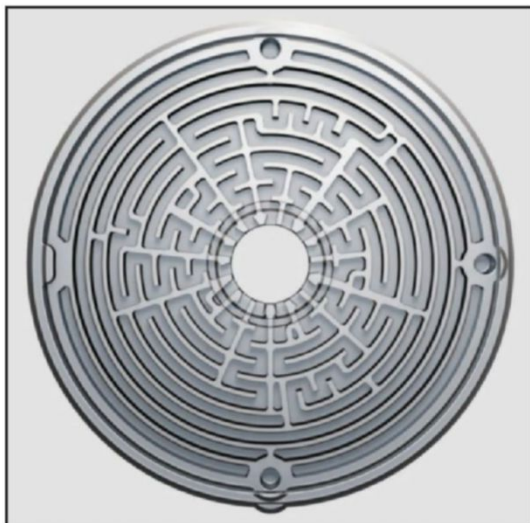
The LS50 Meta's modest footprint makes for easy placement.

Edge W into service. I had reviewed that 100W into 8Ω stereo amp in the January 2020 issue (<https://audioxpress.com/files/attachment/2712>), and was fortunate enough to still have it in the house. The high quality, three-way binding posts on the rear of the LS50 Metas made it easy to wrangle my stiff Anticables Level 3.1 speaker wiring into place, plug in the bananas, and get everything hooked up. I left my home brewed swept tones and pink noise files to run overnight. What had started out as a bit gritty and congested rapidly cleaned up nicely to reveal what I would later come to realize is an unusually good sounding loudspeaker. For my first listen, I had stuffed the dual concentric open-cell foam plugs into the ports in the rear. That produced a tubby aspect to the upper bass that I was none too fond of, so I pulled those to let the speakers breathe. They stayed that way for the remainder of my listening.



The distinctive mid-woofer and concentric waveguide telegraph KEF's brand from afar.

Detailed perspective of the new Metamaterial Absorption Technology (MAT) labyrinth, with multiple ducts tuned to specific frequencies. This physically compact construction has an amazing super power: It absorbs 99% of the unwanted sound radiating from the rear of the driver, reducing unwanted resonances and distortion.



Listening sessions began using the native Qobuz app. I then quickly switched to the same playlist and songs in Audirvana. The improvement in clarity and transient response with the combination of painstakingly crafted player app and innovatively engineered LS50 Meta was immediately evident. The same goes for Tidal. Switching from Audirvana to Tidal's native app resulted in a shallower, shrunken soundstage and slightly darker top end, and that with an MQA half unfolding in Tidal's app.



Detailed view of the new KEF Uni-Q driver with the MAT absorber labyrinth applied.

About the Author

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



While listening to the title track from Bruce Springsteen's latest, "Letter To You" [Qobuz 96k], I could clearly perceive a left-channel rhythm guitar strumming away just to the left of the left-hand loudspeaker. The LS50 Meta throws a wide, not tall soundstage, at least in my untreated room. I preferred the sound of the LS50 Meta directly on-axis vertically for the widest soundstage and flattest response. Vertical dispersion is fairly narrow, Kent Peterson's measurements (see the accompanying article) should provide more insight there. I do recommend listening with your ears on about the same plane as the tweeter for best overall frequency response.

Despite its rather compact, very pleasing dimensions, the speakers launch a satisfying, surprisingly tight low end into the room. I heard usable level from a 30Hz warble tone played at nominal SPL. While not reproducing the last half octave, the bass is scaled properly, with good pitch definition and snappy transient response on kick and other big percussion. On *The Comet is Coming's* submerged dub opus, "Birth Of Creation" [Qobuz 96k], the synthetic drums and synthesizer were clearly delineated and distinct from each other. On Sarah Tandy's "Bradbury Street" [Tidal 44.1 MQA], Mutale Chashi's bass line came through as slightly vague. By contrast, the naked bass behind Zara McFarlane's vocals on the intro to Peter Edwards Trio's live performance of "Feed The Spirit" [Tidal 88.2 MQA; *Jazz re:freshed Live*, Volume 1 compilation] was clean and continued as nicely delineated through the track. To me, this revealed differences in production between tracks rather than the loudspeakers blurring distinction at low frequencies.

Granted, a speaker as small as the LS50 Meta cannot plumb the depths as larger speakers can. Your brain fills in some of that missing energy, but I didn't find them wanting.

At the other end of the scale, I have to say they are polite, not aggressive, in a good way. On Robert Glasper's single, "Better Than I imagined" [Tidal 96k MQA], featuring H.E.R. and Meshell Ndegeocello, the snare and hi-hat were sharply defined without etch or harshness. The reverb sheen and bloom sat nicely in Glasper's and engineer John Rooney's tasteful production. For my how-loud-can-they-play test, I enlisted Soundgarden's evergreen "Black Hole Sun" [Qobuz 96k] via the Avishai Cohen Listens playlist. Despite playback volumes that call for short-term exposure before the onset of permanent damage, there was not a hint of compression.

In the Studio

I wanted to provide a real-world comparison and, as I don't own any speakers close in price or

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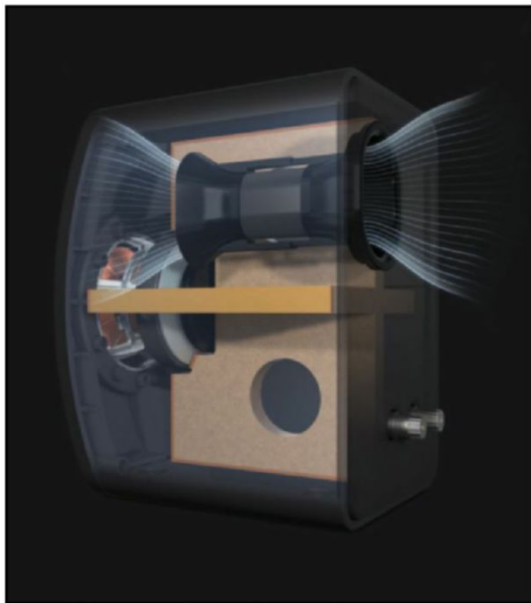
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Other enhancements from the original LS50 include an offset flexible bass port.



type, I hauled a pile of gear over to Kent Peterson's commercial studio. 8VB Recording in Minneapolis, MN, counts two pro stand mounters among its inventory; Sennheiser's Neumann KH 80 and the larger PreSonus Sceptre S8. Both are ported, powered two-way designs. Since the LS50 Metas are not powered, we wrestled the Edge W into place under the mixing console and ran my set of Anticables up to the Metas.

Though the sound of the Presonus speakers were new to me, I have some familiarity with the Neumann's house sound, having done seemingly countless demos at an Audio Engineering Society (AES) convention using their compact KH 310 family. A bit after getting everything dialed in, Kent's business partner, drummer Pete Quirsfeld, arrived to join us in our nitpicking endeavors. Pete sniffed around the Metas and proclaimed them handsome, my better half would not disagree. In short order, we

What are Metamaterials?

Manmade structures called metamaterials have significantly upgraded our ability to steer waves and energy in electromagnetism, optics, acoustics, and mechanics. Our current understanding of acoustic metamaterials stems from previous work done in the realms of electromagnetism and photonics. Radar cloaking and superlensing, compact, man-made structures that operate on a scale much less than the relevant wavelength of a traditional lens, resulted from that research. Starting early in this century, the principles learned from those and prior efforts resulted in the creation of the first modern acoustic metamaterials. Optical and acoustical metamaterials share many similar approaches to implementation. By incorporating novel, carefully chosen subwavelength structures into macroscopic materials, acousticians can create devices with exotic sound-altering properties.

As Sebastien Degraeve and Jack Ocleo-Brown state in their Audio Engineering Society (AES) Convention Paper 10341 "Metamaterial Absorber for Loudspeaker Enclosures":

"Acoustic metamaterial absorbers can realise previously unattainable absorption spectra with sub-wavelength dimensions approaching the theoretical minimum...To shrink the size of the rear absorber, one possible approach is to use resonant acoustical absorbers such as Helmholtz oscillators or quarter-wavelength ducts.

Acoustical resonators are extremely efficient sub-wavelength absorbers but they have the disadvantage of being narrowband in character. Adding damping increases the bandwidth but limits the absorption. An ideal solution would be to arrange a structure containing many high-Q resonators optimised to provide a wide bandwidth of overall absorption. In this context, acoustic

metamaterials can deliver unconventional effective properties without the constraints normally imposed by nature."

There it is in a nutshell, those last two sentences tell the tale. Cluster together a collection of sub-wavelength tuned structures, and their lumped behavior acts in a novel manner. In the past two decades, the steady advance of additive manufacturing techniques in concert with improvements in computational sciences has resulted in the fabrication of physical objects with elaborate topologies that defy Newtonian physics and the recognized canon of legendary acousticians with their non-intuitive behavior. Imagine that someday an advanced hobbyist will build a full-frequency acoustical pipe organ in the footprint of a coffee table.

As all speaker builders know, the bulk impedance of traditional absorption is inconsistent when applied, relatively large physically and a bit of a pain to work with. As Steve Cummer, professor of electrical and computer engineering at Duke University, states on YouTube, "The materials that are used (for sound-proofing or amplitude reduction) right now are (physically) pretty big and not easy to deploy, and absorb essentially all frequencies of sound, all pitches. With metamaterials, we have the ability to design much more compact, thinner structures that absorb only the frequencies, the pitches, that we want to absorb, leaving the rest of the sound... unaltered."

In 2006, a Duke research team demo'd an optical cloaking device. Then, in 2014, Cummer and his band of merry experimenters announced the world's first three-dimensional acoustic cloaking device, a perforated plastic metamaterial shell that can be placed over a small object to make it acoustically invisible. The Romulans would be proud.

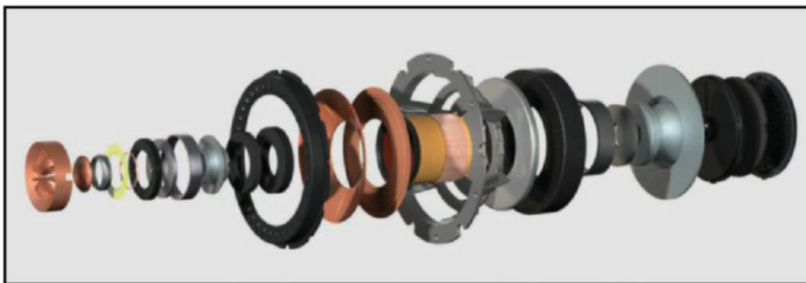
fired up Tidal's browser playback mode and started listening to a range of styles during the evening, from De Staat and Outkast to Brad Mehldau and Joe Henderson. What struck us all was the similarity of voicing between the Neumanns and the KEFs. The KEFs had a touch more upper-mid presence, while the Neumanns had a tighter, more extended bottom, but their similar midrange honesty was notable. These are both examples of relatively uncolored truth, which we found surprising since one is a consumer product. That KEF voicing speaks to a philosophy I share with them, an audio "Hippocratic Oath." It's an aim to do no harm to your music, conveying the intent without warping the timbre, tarting it up, or adding layers of distortion for good or ill.

Following a good bit of listening to that pairing, we swapped out the Neumanns for the PreSonuses and were immediately struck by the difference in midrange and bass. The Sceptre S8 focuses on the midrange, with an in-your-face presentation that takes no prisoners. It's sort of an NS-10 on steroids. The larger cabinet volume of the Sceptre S8s made for a more extended low end, but their shouty character swamped any other aspect when trying to make a comparison. I also detected a throaty telltale with the PreSonus kids, their tweeter is horn-loaded, which was absent in the other two. After several hours of swapping seats and furrowed brows, we all agreed that we could easily live comfortably with the Metas.

The Verdict

After all was said and done, there's little I could find at fault with Oclew-Brown's creation. Yes, they are "bookshelf" speakers but, being rear ported, would not be happy in a bookshelf. They also do not go terribly low, nothing that KEF's Kube Series sub (\$550 to \$750) or the newly released KC62 subwoofer with Uni-Core technology (\$1,499), can't cure. Also, their high end is somewhat limited. They start to roll off slightly at about 15 kHz and so do not have gobs of "air." Yet, within their usable range, they provide surprisingly flat, well-behaved frequency response and low distortion resulting in a winning, balanced character.

For some, this is an aspirational product and, for others, it's a bargain. The LS50 Meta sits in that value sweet spot of nice price, exceptional performance without significant qualification, and a clean visual appeal for those who care about appearances as well as sound quality. Very few speakers that I know of targeted at consumers portray the neutrality and intent as well as the second-generation LS50 passive. Designer Oclew-



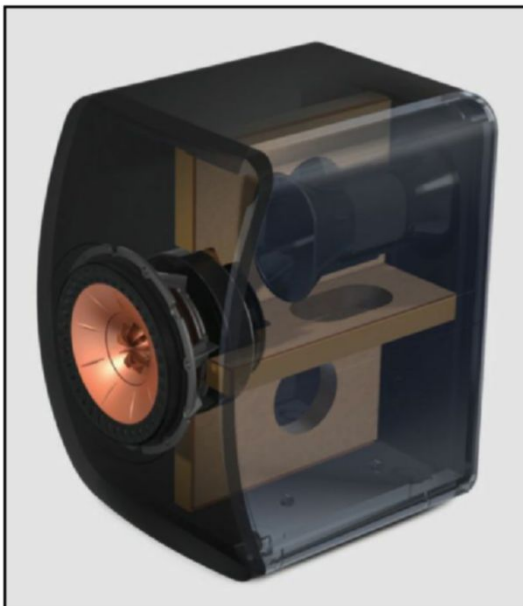
The new KEF LS50 Meta also features a specially designed 12th generation Uni-Q, benefiting from a new cone and motor system configuration.

Brown told me that, "We certainly try and get (the on-axis response) nice and tidy but we're more focused on delivering an even sound into the room in all directions. Our CEA2034 (spinorama) acoustical testing data and resulting graphs have many different curves, all of which are smooth and monotonic. This is the key to a great sounding loudspeaker."

Amen brother—great sounding, compact, and good looking. The LS50 Meta has a wonderful combination of attributes, all of which combine to reinforce my original impression. I l-o-v-e these speakers and, at \$1500 a pair, they would serve any audiophile in good stead for many years. Very highly recommended. 

Author's Note:

* Gold Peak Group, the Hong Kong-based corporate parent of GP Acoustics (UK), is an Asian multinational whose holdings include KEF and Celestion.



The newly optimized cabinet size, which is elegantly curved in the front, features an improved back panel with recessed port and speaker terminals, and rigid cross bracing with constrained layer damping.

KEF LS50 Meta

The Measurements

After conducting a serious listening session of the KEF LS50 Meta with Oliver Masciarotte and Pete Quirsfeld, Kent Peterson took the speakers to Warkwyn to determine if their listening impressions could be confirmed with actual measurements.

By Kent Peterson

The KEF LS50 Meta was measured at Warkwyn's facility using the Klippel Near-Field Scanning (NFS) system (**Photo 1**). The NFS delivers a 360° acoustical radiation balloon allowing for an examination of the radiation pattern, on- and off-axis frequency response at any angle or distance, and any directionally important information, which may help when determining where to place this coaxial speaker into your listening environment.

For the measurements of the LS50, a calibrated ACO Pacific free-field microphone with the 4048 pre-amp and a 7052E capsule were used. All on- and off-axis data is referenced at 1m for sensitivity (2.83V), and then increased to 3m for some evaluations in the listening field.

Measurement points around the speaker totaled 1800 and were processed with a resolution of 18 points per octave and from 20Hz to 20kHz. The length of the stimulus was 0.90 seconds with an average of 4.

Sensitivity

The LS50 Meta comes with a port plug, so one of the initial measurements was an assessment of the plug's effect on frequency response with and without the port plug. **Figure 1** shows that result on-axis and at 1W/1m.

With the port open, the LS50 provides a 1dB increase in amplitude at 140Hz and offers an approximate 1/6 of an octave gain in low-frequency response below 100Hz, and an increased benefit 10dB down from 60Hz to 40Hz. That being the case, the results of our measurements here will be based on the port being plugged as this is the default configuration in which Warkwyn received the speakers.

Horizontal On- and Off-Axis Frequency Measurements

Examining the frequency curve on-axis, the LS50 has an impressively smooth response from 135Hz up to 3kHz. Please note the Y-axis, 5dB scale, which translates into a ± 2 dB range of unsmoothed deviation within these frequencies. At 3.5kHz, we see a 3dB rise ending at 10kHz, where the frequency response gradually descends 2.5dB ending at 11kHz. Another peak is then encountered at 16kHz where the device rolls off to its measurement limitations.



Photo 1: The KEF LS50 Meta is shown mounted for measurements.

Taken as a whole, the complete frequency response exhibits a ± 2 dB variation throughout most of the range, with a dip that spans 1.75 octaves, from 1kHz to 3.5kHz. This might result in a slight reduction in definition, but considering the -1dB to -2dB in level, and the increase at 4kHz, sibilance doesn't suffer greatly.

Examining the off-axis frequency response curves at 15°; this co-axial system holds together admirably (**Figure 2**). Even 30° off-axis, the frequency response holds together very well.

Horizontal and Vertical Contour Maps

The unwrapped horizontal contour map is an examination of the frequency to radiation strength in a "top-down" fashion (**Figure 3**). As the head moves to the left (or down), it is moving to the left around the speaker until the point it reaches the -180° position on the X-axis and behind the speaker. As it moves right (or up in the graphic), it is experiencing the frequency response and radiation around the right side of the speaker.

As a co-axial system, frequency response on-axis is very good from 18kHz down to 3kHz and for a 40° angle. At 3kHz, a

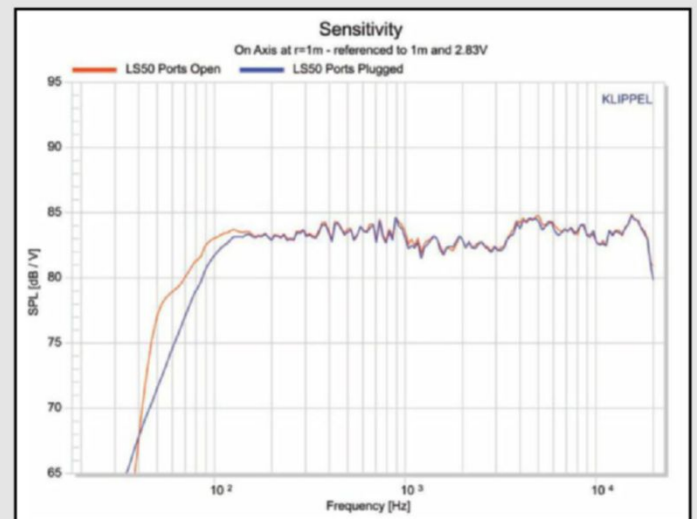


Figure 1: The KEF LS50 frequency response was measured showing port plugged vs. port open.

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2dB to 3dB dip in frequency response is encountered but this recovers at 1.5kHz.

The radiation pattern is good up to 30° off-axis and to a 6dB down point. This should result in an even and relatively wide listening widow though the vocal region.

Another slight dip in response is encountered from 250Hz down to 150Hz where the pressure level recovers and then drops off gradually to 40Hz. Again, this analysis is based upon the ports being closed on the system. With the ports open, an additional 1dB to 2dB in gain is encountered at 40Hz.

The beauty of a coaxial system is the symmetry from horizontal to vertical. The vertical contour map below bears this out (**Figure 4**).

Here we are examining the vertical cross-section from the perspective of the left side of the cabinet. As the head moves up, we are going up and over the cabinet, and conversely down underneath the cabinet and to the back at 180°.

The frequency response is almost identical to the horizontal radiation map with the exception of an asymmetrical loss between 100Hz and 180Hz and at -100°. This would indicate a perceptual difference in low frequency, say kick drum or bass, from sitting on axis to standing and at 3 meters. Of course, your room influence will compensate for some of this difference and, accounting for the CEA 2034 vertical listening window, fall well within $\pm 10^\circ$. At these same frequencies, the system is quite even from left to right as seen in the horizontal mapping. **Figure 5** details the CEA 2034 results.

CEA 2034 Results

CEA 2034 defines the listening window as an average response of nine frequency response curves within the $\pm 10^\circ$ vertical and $\pm 30^\circ$ horizontal angular range. As we have illustrated with the on- and off-axis measurements, the LS50 Meta has a respectable frequency response within these ranges. On-axis, the system has a slight 2dB dip between 1kHz and 3.5kHz. This is not improved in the listening window. However, there is 1dB to 2dB reduction

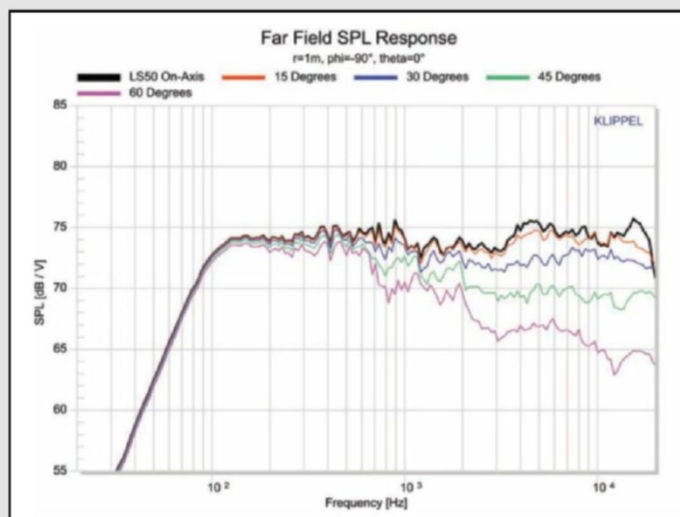


Figure 2: Horizontal sensitivity is measured on-axis at 15°, 30°, 45°, and 60° (no smoothing).

in frequency response from 3.5kHz to 10kHz, which provides further smoothing. And the bump found on the on-axis curve at 16kHz is reduced in the listening window 2dB resulting in an over-all smoother response that the single on-axis frequency response illustrates.

The Sound Power DI confirms the LS50's higher directivity from 1kHz and upward. Ultimately, the CEA 2034 assesses the LS50 Meta as a well performing system within the listening window and in a "typical listening room."

Listening and Commentary

In our usual nocturnal fashion, Oliver Masciarotte and I met in a dark alley on November, Friday the 13th (!) to heft a Cambridge Audio Edge W for amplification, and the KEFs to enjoy some time listening at 8VB Recording in downtown Minneapolis. The DAC used was a Presonus Studio 192 running at 24-bit/192kHz via Oliver's Anticables Level 3.1 speaker cables. We were joined by Peter Quirsfeld, owner of young fresh ears and drummer/percussionist for the lovely band Humbird.

8VB typically monitors with powered Neumann KH120s (5.25"/1" dome tweeter two-way) as well as larger, powered Presonus Sceptre 8's (8"/1" compression driver coaxially mounted in 90° x 90° horn). Size-wise, the KH120s were similar to the LS50s so that was our go-to for comparison. The KEFs were placed on the outside of the KH120s, and the KH120s were on their side with tweeters on-axis and positioned to the outside of the woofer (**Photo 2**).

The Listening Experience

The listening position is centered between the systems, at tweeter ear height and within 2 meters from the speakers. To

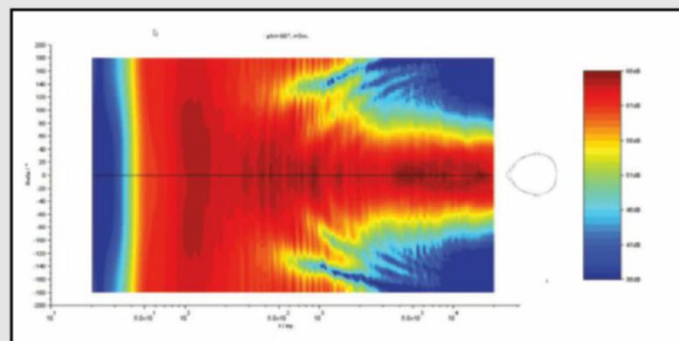


Figure 3: Unwrapped horizontal contour balloon, 3m

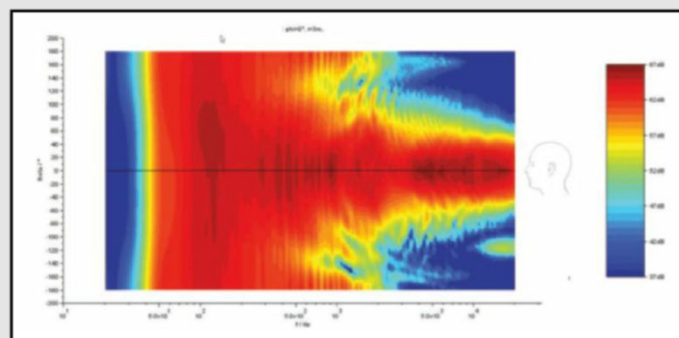


Figure 4: Unwrapped vertical contour balloon, 3m

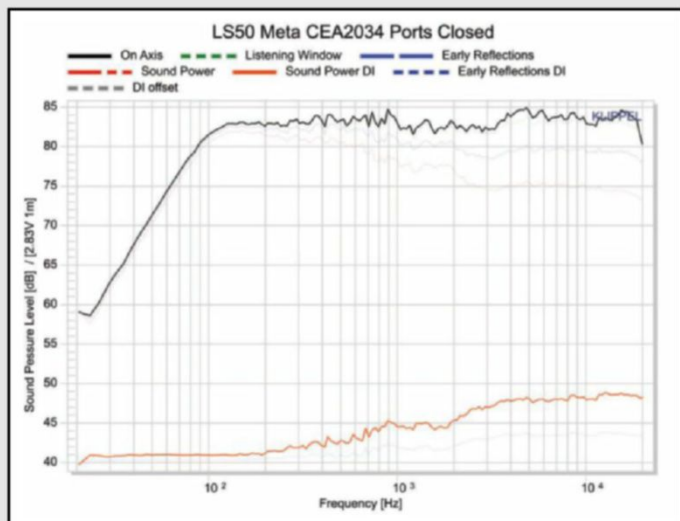


Figure 5: CEA 2034 response curves

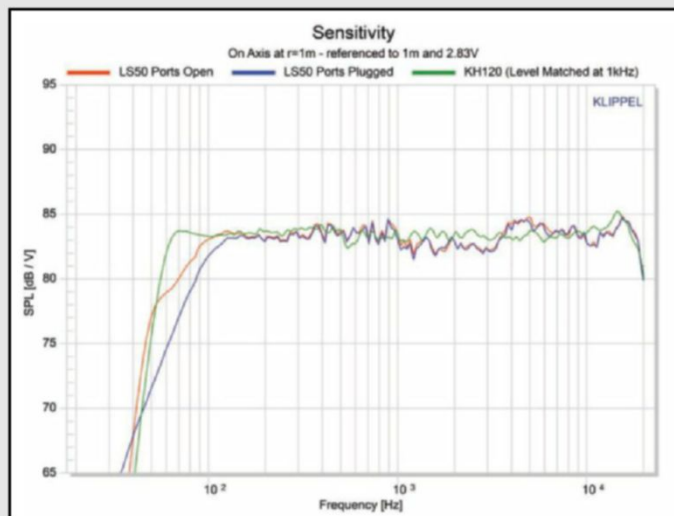


Figure 6: Neumann KH120 and KEF LS50 are shown in listening position.

be fair, the mix position at 8VB is not your “typical” listening environment as it is a recording studio control room with plenty of absorption behind the speakers, as well as laterally in the room. Immediately behind the listening position is a comfortable leather couch with 6’ x 6’ RPG Skyline diffuser on the rear wall. Above the listening position, a 2’ deep x 4’ x 6’ absorptive cloud hangs. Additionally, the NC of the room approaches 20. So, while not typical, nicely controlled and very quiet.

Listening spanned the gambit from Blue Note jazz, to hard rock, pop, and some ambient. Each of us, Oliver, Peter, and I spent a considerable amount of time right on-axis. During my time on center, I resorted to some of my go-to recordings—Thomas Dolby both old and new due to his always tight bass and sub structure and beautifully recorded piano and vocals; some Stan Getz to hear the definition of his stark and dry sax; and some recordings I am currently working on to assess the differences between what I hear in a mix between the KEFs and the Neumanns. During times either sitting on the couch or standing and moving left to right within the mix space, I found the KEFs to be very similar to the Neumanns especially in the high-frequency from 1kHz and up

to 3kHz. The boost in frequency response of the KEFs however, from 3.5kHz to 10kHz, was very apparent when switching back and forth. This created a perceived clarity I wasn’t necessarily hearing from the KH120s. This was confirmed by both Peter and Oliver while on-axis and moving side to side.

Low-frequency output of the KEF’s was good. However, when switching back and forth to the Neumanns, there was a noticeable low-frequency improvement with the Neumanns with greater overall output and depth.

Thankfully, I realized that I had some Klippel measurement data on the Neumann KH120s that I could use in comparison to the measured KEFs. **Figure 6** shows that measurement comparison.

After examining the comparison curves my listening impressions were confirmed. The KH120 has greater low-frequency “extension” than the LS50’s even with the ports open. The rise in response from 3.5kHz to around 9kHz of the KEFs, resulted in a greater impression of clarity and certainly brightness. Not unpleasing but clearly audible in comparison to the Neumanns. Again, we were listening in an acoustically treated room intended for accuracy. In a “typical” environment and as illustrated in the CEA 2034 curves, this would be diminished with harder reflections.

As we packed up for the evening, we were all quite impressed with the KEFs. For me in particular, not only did I like how they sounded, I also love the physical presentation. They feel fantastic and are quite striking. For a week I struggled with how I could hide them from my wife until a time I could honestly tell her “Oh, those old things? I’ve had them forever.” Kudos KEF for some great sounding and gorgeous looking speakers, I really enjoyed them. 

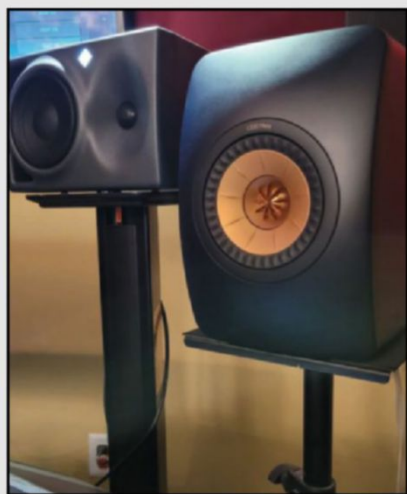


Photo 2: Neumann KH120 and KEF LS50 are shown in listening position.

About the Author

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

Measuring Vibration with Piezoelectric Sensors (Part 3)

The Piezoelectric Amplifier

By
George Ntanavaras



In the third and final part of this article series, George Ntanavaras describes the design and construction of the piezoelectric amplifier and presents the measurements he took to evaluate the system.

I designed a piezoelectric amplifier especially for the PZ-1 sensor, but other piezoelectric sensors could be connected although some modifications might be needed. The electronic diagram of the piezoelectric amplifier is shown in **Figure 1**.

A detailed description of the operation of the circuit follows:

- At the input of the circuit, there is a 3-pin connector for the sensor—the first is the positive input (IN-1); the middle is the ground pin (IN-2); and the third is the negative pin (IN-3).
- The differential charge amplifier is built around op-amp IC1A. Resistors R1 and R20 offer some protection against possible over-voltages at the input of the amplifier. Capacitors C1-C4 and C19-C22 adjust the gain of the amplifier while resistors R2-R5 and R21-R24 are used to properly bias the op-amp. I used four parallel connected capacitors and four series connected resistors to achieve better matching for the total values of the positive and negative side

since this affects the common mode voltage rejection of the circuit.

- IC1 has to be an ultra-low input current operational amplifier and I used in this position the CMOS dual op-amp LMC6482. It has a input current of 20fA, a rail-to-rail output swing, a high input impedance, a large common-mode range and a high CMRR. The last two are very important for the differential circuits. According to its datasheet, the circuits designed with the LMC6482 can reject a larger range of common-mode signals than others making them an excellent choice for operation in noisy environments.
- IC1B provides an additional gain of 16dB to the signal. C31 with R26 form a low-pass filter at around 40kHz to reduce the bandwidth and the high frequency noise.
- Switch S1 selects between 3Hz, 16Hz, or 160Hz for the cut-off frequency of the first-order high-pass filter that is formed with capacitors C6, C10, C11, and R15. The IC2A buffers the high-pass filter and through switch S2 provides

different gains between 0dB, 10dB, and 20dB by selecting resistors R32, R33, R34 as feedback for the IC2A.

- The components around IC2B and IC3A form a two-stage sixth-order low-pass filter with the -3dB point at 12kHz. This gives an attenuation of about 37dB at 31kHz and suppresses the very high Q resonance of the PKGS-00GXP1-R sensor to achieve a more flat response.

For a better understanding of the next circuits, let's refresh our memory with the basics of the simple harmonic oscillation theory. Suppose an object oscillates periodically with a frequency f and the instantaneous value of the excursion (=displacement) is given as the function of time by $x(t)$. We can calculate the velocity $v(t)$ of the object as the first derivative of the excursion and the acceleration $a(t)$ as the first derivative of the velocity or the second derivative of the excursion.

Now let's think the opposite, supposing that the acceleration $a(t)$ is known. The velocity can be calculated by integrating the acceleration signal over time while the excursion by integrating the velocity over time. This is our case—the sensor gives the acceleration and we need an integrator to calculate the velocity. This can be realized very easily with an electronic integration circuit based on an op-amp, a capacitor, and a resistor. It is similar to an active low-pass filter with a very low cut-off frequency.

The amplifier includes two such integrator circuits in series—one for the calculation of the velocity and another for the excursion.

The op-amp IC3B with R6 and C5 form the first integrator that gives at its output the velocity of the input signal. Resistor R7 is in parallel with the integration capacitor C5 limiting the gain of the circuit at the very low frequencies to a finite value.

The op-amp IC4A with R31 and C23 form the second integrator that gives at its output the excursion (=displacement) of the input signal.

Switch S3 selects the type of the signal that will be connected to the output of the amplifier. Resistors R27, R29, R36 form dividers with R28 properly adjusting the level of the output signal. IC4B is a unity gain buffer that drives the selected signal to the output connector.

The conversions factors between the level of the output voltage and the absolute values of the measured quantities were defined as follows:

- acceleration 1 mVrms corresponds to 1m/sec²
- velocity 1 mVrms corresponds to 0.01m/sec
- excursion 1 mVrms corresponds to 0.01mm

These factors are valid only for the the PZ-1 sensor supposing that the Murata sensor has its nominal sensitivity. The large tolerances of the Murata sensor used in PZ-1 and PZ-1-sh sensors, ($\pm 15\%$ for its charge sensitivity and $\pm 30\%$ for its capacitance), cannot allow a very accurate output, so the above factors should be taken only as indicative. The amplifier's gain was adjusted so that when the PZ-1 sensor is connected at its input, the voltage at the acceleration output will be similar to the voltage

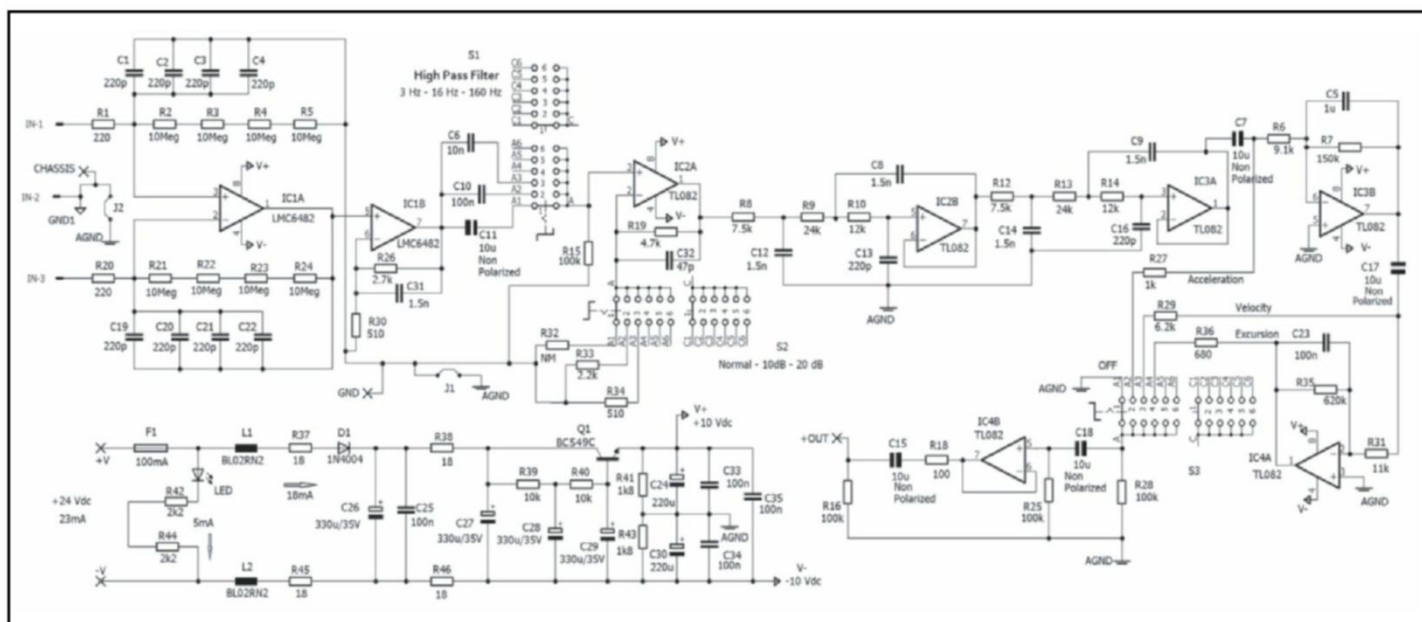


Figure 1: This is the electronic diagram of the piezoelectric amplifier.

About the Author

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

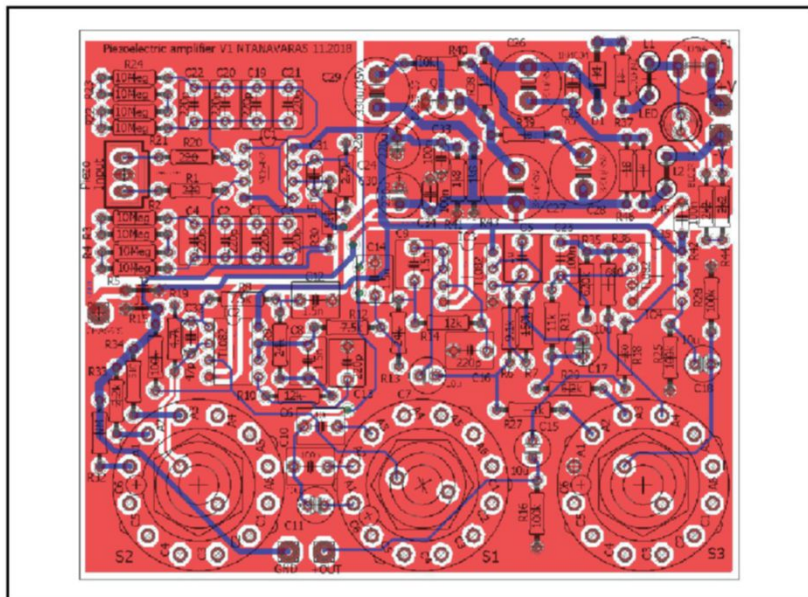


Figure 2: The dimensions of the PCB are 84mm x 101mm, and this board shows the placement of the components.

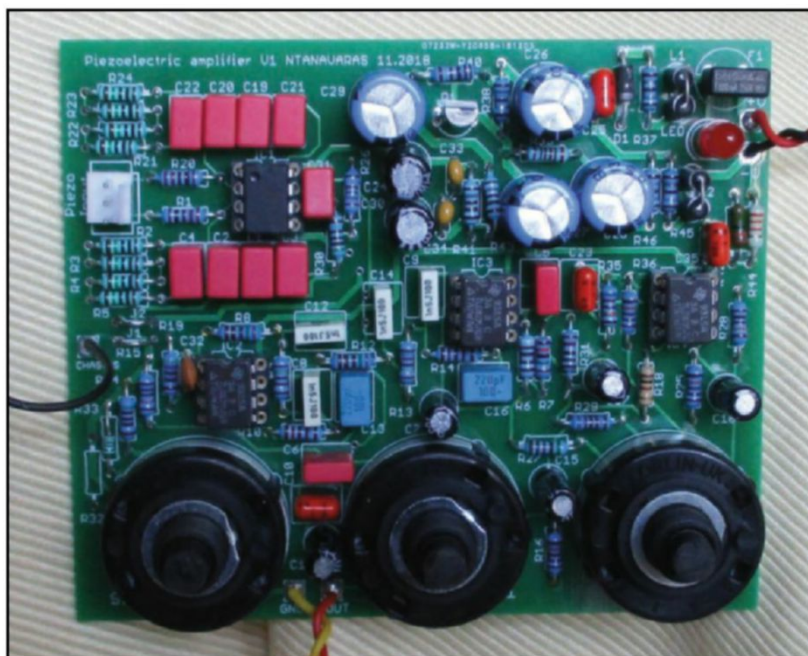


Photo 1: The assembled PCB shows the three rotary switches S1, S2, and S3 soldered directly on the PCB to facilitate the construction of the amplifier.

that will be produced by the ACH accelerometer when connected to its amplifier.

The gains of the integrators for the velocity and the excursion were adjusted to meet the previously mentioned conversions factors.

The amplifier needs 24 VDC for its operation and has a low consumption at about 23mA out of which the 5mA are used for the power LED indication. I used a small Desktop AC adapter from XP Power, type VET18US240C2-JA. It is a 18W switching power supply with 24 VDC/750mA output but any other power supply with low ripple will be suitable.

Fuse F1 is used for protection of possible excessive currents in case of a fault and when the external power supply has no short circuit protection. L1 and L2 are ferrite beads with lead wires to attenuate the very high frequencies for suppression of EMI noise. Diode D1 assures that the input voltage will be connected to the amplifier with the correct polarity. Resistors R37, R38, R45, R46 with capacitors C25-C27 form low-pass filters to additionally suppress the high-frequency noise of the power supply. Transistor Q1 forms a capacitance multiplier, which provides at its output a very low noise DC voltage of about ± 10 VDC for the supply of the circuit.

Resistors R41 and R43 are equal and their common point is the virtual ground of the circuit. Capacitors C24, C30, C33, C34 decouple the supply voltages improving the stability of the op-amps.



Photo 2: For the completed amplifier. I used the same metal box from Hammond and the aluminum frontplate that I had designed for the ACH-01 amplifier.

The Construction of the Amplifier

I designed the PCB for the amplifier using the Demo version of the Eagle Layout editor. This demo version of the program can be downloaded free of charge from the Cadsoft website (www.cadsoftusa.com). The demo version is fully operational except a limitation on the maximum dimensions of the PCB, which was not a problem for this project.

I ordered a high-quality PCB manufactured from FR4 composite material with a 1.6mm thickness and 35µm copper solder on both sides and silkscreen on the top side. The dimensions of the PCB are 84mm × 101mm and the placement of the components is shown in **Figure 2**. All the components are placed on the top side of the PCB.

Photo 1 shows the assembled PCB. The three rotary switches S1, S2, and S3 are soldered directly on the PCB to facilitate the construction of the amplifier.

I used good quality IC sockets for the op-amps on my prototype, but this is not necessary unless you are planning to test different types of op-amps. The three switches are fastened directly on the front panel and support the whole PCB. The output

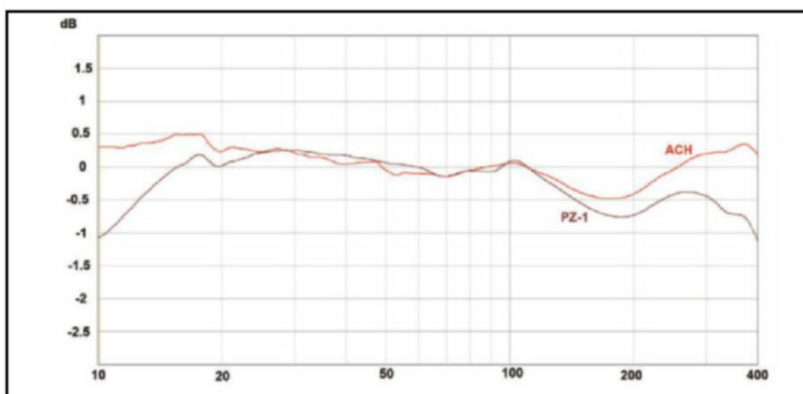


Figure 3: The graph shows a comparison between the acceleration response of the Peerless cone measured by the ACH and the PZ-1.

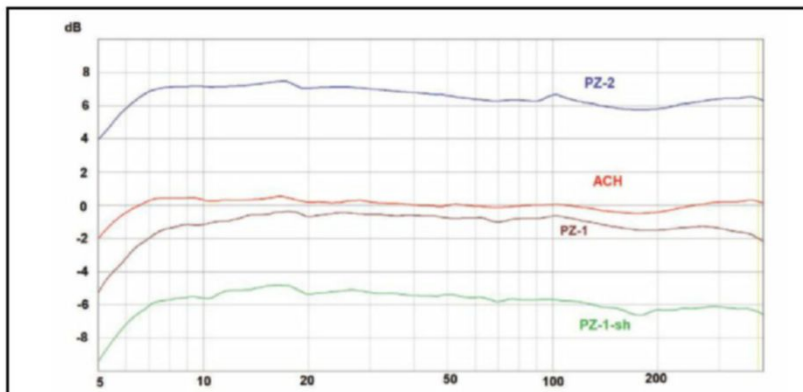


Figure 4: This graph details the acceleration of the Peerless cone as measured with the PZ-2, PZ-1, PZ-1-sh, and ACH without normalization.

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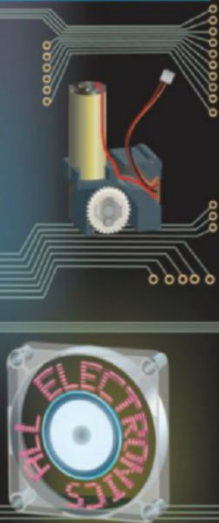
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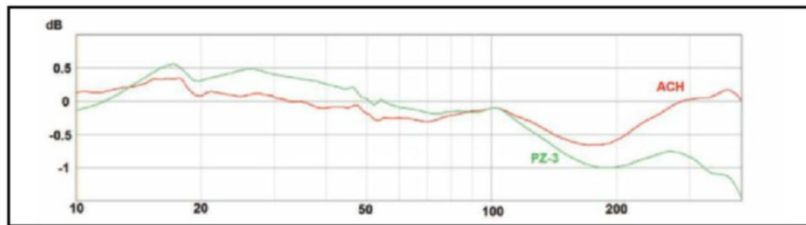


Figure 5: In this graph we compare the acceleration response of the PZ-3 and ACH sensors after they were normalized for equal level at around 100Hz.

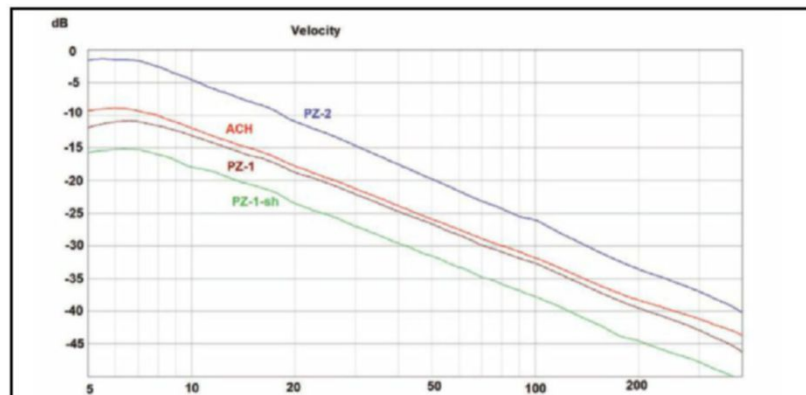


Figure 6: This graph details the velocity of the Peerless cone as measured with the four sensors without normalization.

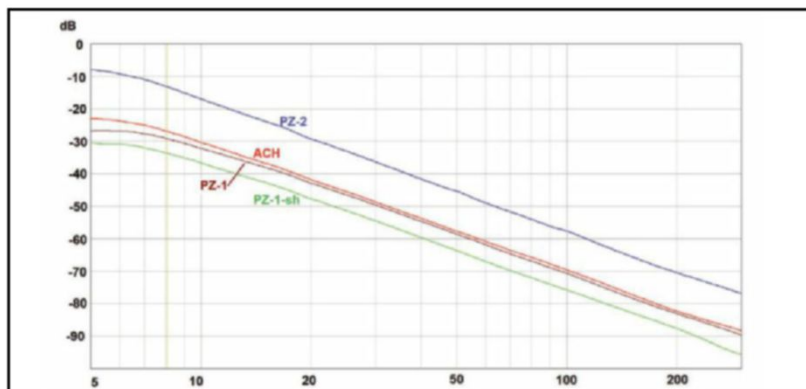


Figure 7: Here we see the excursion of the Peerless cone as measured with the four sensors without normalization.

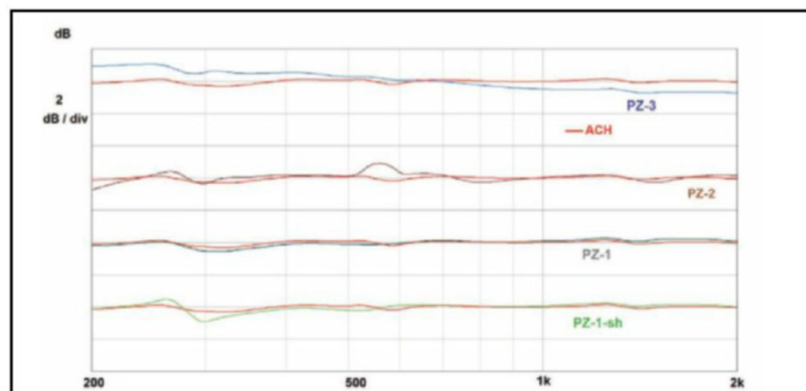


Figure 8: In this graph we see the acceleration of the Visaton exciter as measured with the four sensors.

connector and the supply connector are placed on the front panel.

The Parts List details the components required to build the amplifier. **Photo 2** shows the completed amplifier. I used the same metal box from Hammond and the aluminum frontplate that I had designed for the ACH-01 amplifier. This is an engraved front panel, which replaces the original top cover of the Hammond box. I placed two adhesive labels to cover the engraved text for the input of the ACH and the power supply as shown in Photo 2, since they are different for this amplifier. The ground point of the PCB of the amplifier is connected with a short wire to the metal box chassis to ensure better shielding.

The Sensor Placement

Proper placement of the piezoelectric sensor on the vibrating surface is very critical—a sensor installed incorrectly can give very wrong results.

The ACH manufacturer recommends to use a quick setting viscous methyl cyanoacrylate adhesive or any epoxy adhesive. This should be applied sparingly to the clean surface and should be allowed to set.

This is an excellent method for permanent placement but not for temporary use. Although the manufacturer recommends to avoid soft adhesives, such as double-sided tape because they can adversely affect the ACH performance, this is the only method I have found to use for temporary measurements. My preference is a standard thin double-sided tape from Tesa, which is strong enough to hold the sensor in good contact with the surface but allows it to be carefully removed. This is something that requires some experience but it works.

As I already have mentioned, another very important point is that the part of the cable that is close to the sensor should be adhered firmly to the surface and not allowed to move at all.

Measurements with Sensors

For the final evaluation of the system, I will present some of the measurements that I took with the piezoelectric sensors attached to the Peerless cone for the frequency range between 5Hz and 400Hz and on the Visaton Bass Exciter for the frequency range between 200Hz and 2kHz.

Figure 3 shows a comparison between the acceleration response of the Peerless cone measured by the ACH and the PZ-1. The two curves were normalized for equal level at 70Hz for easy comparison. They have very similar response, within 0.5dB, between 17Hz and 220Hz. Probably, they have very similar response above 220Hz, but

the break-up mode of the subwoofer's cone (not shown) starts to influence each measurement in a different way since it occurs at very different frequencies, 720Hz for the ACH and 910Hz for the PZ-1.

Figure 4 shows the acceleration of the Peerless cone as measured with the PZ-2, PZ-1, PZ-1-sh and ACH without normalization. The PZ-2 is 6dB to 7dB more sensitive than the ACH. The responses of the ACH and PZ-1 are very close, with the PZ-1 less sensitive for about 1dB. Considering the large tolerance of the Murata sensor, this is a very nice result. The shielded sensor PZ-1-sh is 5dB to 6dB lower in sensitivity than the ACH. Both the PZ-1 and PZ-1-sh have less than 1dB difference from the response of the ACH, when their responses are normalized.

The PZ-3 sensor is much more sensitive than the other sensors, almost 35dB higher. **Figure 5** compares the acceleration response of the PZ-3 and ACH sensors after they were normalized for equal level at around 100Hz. They are very similar.

Figure 6 shows the velocity and **Figure 7** shows the excursion of the Peerless cone as measured with the four sensors without normalization. The responses are very close.

I also measured the distortion of the Peerless cone's acceleration and the results were very similar for all four sensors.

Figure 8 shows the acceleration of the Visaton exciter as measured with the four sensors. Here I placed the response of the ACH normalized at the same level to the responses of each sensor for easy comparison. Again the response of PZ-1 and PZ-1-sh are almost identical to the response of the ACH. The small deviation in the response of the PZ-1-sh around 280Hz is probably some minor resonance due to the placement.

Figure 9 shows the velocity, and **Figure 10** shows the excursion of the Visaton exciter as measured with the sensors. The responses were again normalized to have the same level. It should be noted that the measurement of the excursion is very difficult at the higher frequencies because the signal levels are much lower in level.

Another example that will interest a lot of loudspeaker builders is shown in **Figure 11**—a panel vibration measurement of a loudspeaker cabinet. It was measured by attaching each accelerometer to the same point at the front panel of a well-braced cabinet. I used the ARTA software producing a pink noise signal and a power amplifier to drive the loudspeaker at a high sound level. The piezoelectric amplifier gain was set to 20dB gain because the signals were very low in level. The red curve

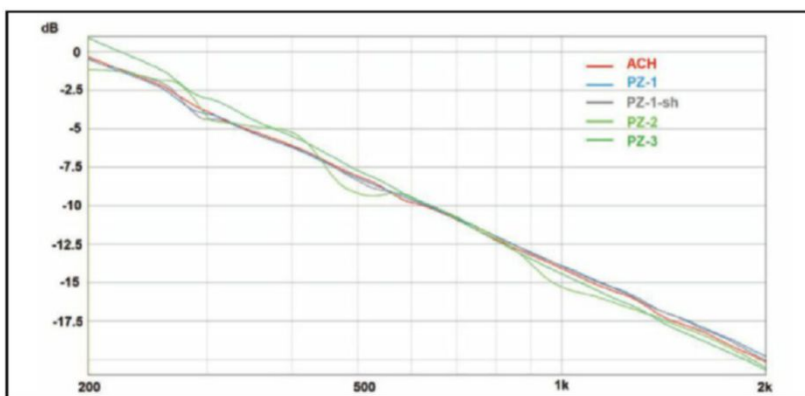


Figure 9: This graph shows the velocity of the Visaton exciter as measured with the sensors.

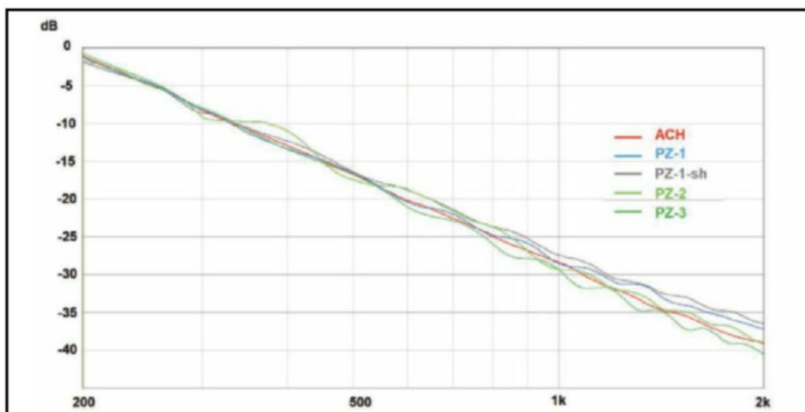


Figure 10: In this graph, we see the excursion of the Visaton exciter as measured with the sensors.

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represents the measurement with the ACH, the blue curve represents the measurement with the PZ1, and the green curve represents the measurement with the PZ-1-sh. The curves shown were offset by several decibels for clarity and they were very

similar, clearly showing the main panel resonance at around 300Hz. An interesting point is that in the response of the PZ-1-sh sensor, the 50Hz mains interference is almost eliminated.

Conclusion

This was a very long journey on which I have learned a lot of interesting things. I believe that the outcome resulted in a very simple-to-build system that is able to generate useful measurements, concerning the acceleration, the velocity, and the excursion of a vibrating surface of a loudspeaker cone, a loudspeaker panel, or a turntable. If you are looking for an easy way to begin exploring vibration measurements, building this system will be a good start.

Author's Note: I have a few PCBs available for the construction of the piezoelectric amplifier and the PZ-1 sensor. Anyone who is interested can please send me an email at: gntanavaras@gmail.com

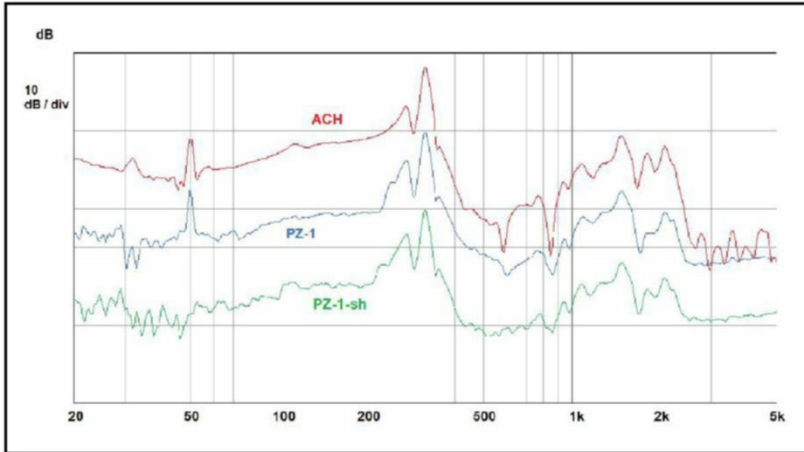


Figure 11: I measured the panel vibration of a loudspeaker cabinet by attaching each accelerometer to the same point at the front panel of a well-braced cabinet

Parts List for the Piezoelectric Amplifier

Part	Value	Notes	Part	Value	Notes
C1, C2, C3, C4, C13, C16,			R8, R12	7.5kΩ	Metal film, 0.25W, 1%
C19, C20, C21, C22	220pF, 100V, 2.5%	Polypropylene	R9, R13	24kΩ	Metal film, 0.25W, 1%
C5	1u	Plastic film	R10, R14	12kΩ	Metal film, 0.25W, 1%
C6	10n	Plastic film	R15, R16, R25, R28	100kΩ	Metal film, 0.25W, 1%
C7, C11, C15, C17, C18	10u / 16V	Non Polar Electrolytic	R18	100Ω	Metal film, 0.25W, 1%
C8, C9, C12, C14, C31	1.5n	Plastic film	R19	4.7kΩ	Metal film, 0.25W, 1%
C10, C23	100n	Plastic film	R26	2.7kΩ	Metal film, 0.25W, 1%
C24, C30	220u / 16V	Electrolytic	R27	1kΩ	Metal film, 0.25W, 1%
C25, C33, C34, C35	100nF	Ceramic	R29	6.2kΩ	Metal film, 0.25W, 1%
C26, C27, C28, C29	330u / 35V	Electrolytic	R30, R34	510Ω	Metal film, 0.25W, 1%
C32	47p	Ceramic	R31	11kΩ	Metal film, 0.25W, 1%
D1	1N4004	Diode	R32	Not placed	
F1	Fuse 100mA	Fuse for PCB	R33, R42, R44	2.2kΩ	Metal film, 0.25W, 1%
IC1	LMC6482	Op-amp	R35	620kΩ	Metal film, 0.25W, 1%
IC2, IC3, IC4	TL082	Op-amp	R36	680Ω	Metal film, 0.25W, 1%
IN	Header, vertical	3-pin, 1 row	R37, R38, R45, R46	18Ω	Metal film, 0.25W, 1%
L1, L2	BL02RN2R1M2B	Ferrite Beads	R39, R40	10kΩ	Metal film, 0.25W, 1%
LED	red	5mm	R41, R43	1.8kΩ	Metal film, 0.25W, 1%
Q1	BC549C	Transistor			
R1, R20	220Ω	Metal film, 0.25W, 1%	S1, S2, S3	Switch, 2-Pole, 6-Position CK1060	
R2, R3, R4, R5, R21,			Metal box	Hammond 1590XXBK	
R22, R23, R24	10 MΩ	Metal film, 0.6W, 1%	Knobs for the switches		
R6	9.1kΩ	Metal film, 0.25W, 1%	Output connector	3.5 mm Jack, chassis	
R7	150kΩ	Metal film, 0.25W, 1%	Power supply connector	Panel Mount, plug 2-pole, 6.3mm	

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Noise is Everywhere—Even in MC Input Transformers (Part 2)

Additional Proof for the Skeptics

In this article, Burkhard Vogel continues to discuss the signal-to-noise (SN) measurements of moving coil (MC) phono amps, demonstrating that it is good practice to measure a transformer with an input load, and that a MC step-up transformer (SUT) produces noise because of its insignificant but non-zero impedance. He concludes his article with a step-by-step approach to noise measurements, simulations, calculations, and conclusions.

By
Burkhard Vogel

In Part 2 of this article, I will continue to prove why I think it is good practice to measure a transformer with an input load and that an MC step-up transformer (SUT) produces noise because of its insignificant but non-zero impedance.

I intend to prove that this kind of transformer always produces more or less significant amounts of—at least—thermal (Johnson) noise, depending on their specs or measured data, their input load R_0 , and the input noise situation of the following moving magnet (MM) phono amp. By the way, microphone

input transformers suffer from the same noise problem. Here, we continue our discussion with the LL9662.

LL9226 Connected to the Input of a Balanced MM Phono Amp

Now we can confirm the given measurement findings with the practical example shown in the **Figure 1**, which is the fully balanced test configuration, described in depth as Amp1 in Chapter 17 and Chapter 18 of my book, *Balanced Phono Amps* [2].

The corresponding math and simulation results on MCD-WS 2 were created with two OP37G op-amps. **Table 1** presents the overall non-equalized SN results for the three op-amp types, mentioned at the beginning of the article.

I hope **Table 2** is also self-explanatory. We can see that, except for lines 4 and 7, there are only very small positive differences between the measured and calculated/simulated SN results. Lines 6, 11, and 16 show the decreasing worsening effect if we increase the input voltage noise of the op-amps. Later, I will explain the steps to get the numbers in the tables and what they have to do with CNI, the current noise impact of a transformer.

I guess, the negative signs in boxes G4 and G7 come from two sources. First, I did not match the op-amps. Second, if we do not exactly match the

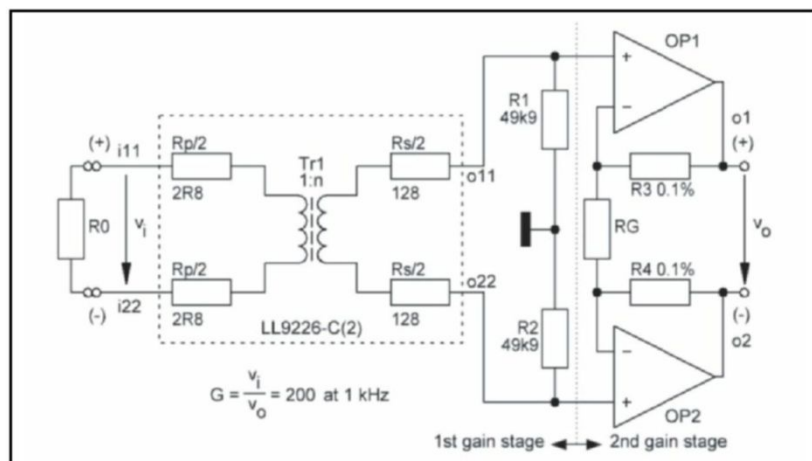


Figure 1: Balanced test configuration with LL transformer and op-amp driven based balanced amp input

1/A	B	C	D	E	F	G	H	I
2	Amp1 : type of op-amp	Measured vs. calculated and simulated signal-to-noise ratios in B_{20k} at the o/p of the balanced Amp1—warm and cold LL9226-C & excl. transformer				meas vs. sim = D - F [dB]	units	Remarks about action possibilities
3		configuration	sim	calc	meas			
4	LT1128C sim with LT model	Amp1 & warm trafo	-98.92	-98.92	-98.78	-0.14	dBV	meas+sim+calc
5		Amp1 & cold trafo	-105.78	-105.82				sim or calc
6		Worsening = line 4-5 in D&E!	6.86	6.90	7.1		dB	calc in D&E, meas in F
7		Amp1 no trafo & i/p short to gnd	-106.61	-106.61	-105.89	-0.72	dBV	meas+sim+calc
8		CNI = line 7-5	0.83	0.79			dB	calc
9	OP37G sim with buvo model	Amp1 & warm trafo	-95.99	-95.99	-96.13	0.14	dBV	meas+sim+calc
10		Amp1 & cold trafo	-98.24	-98.24				sim or calc
11		Worsening = line 9-10 in D&E!	2.26	2.25	2.3		dB	calc in D&E, meas in F
12		Amp1 no trafo & i/p short to gnd	-98.29	-98.28	-98.48	0.19	dBV	meas+sim+calc
13		CNI = line 12 -10	0.05	0.05			dB	calc
14	OPA627B sim with buvo model	Amp1 & warm trafo	-93.89	-93.89	-94.30	0.41	dBV	meas+sim+calc
15		Amp1 & cold trafo	-95.15	-95.15				sim or calc
16		Worsening = line 14-15 in D&E!	1.26	1.26	1.4		dB	calc in D&E, meas in F
17		Amp1 no trafo & i/p short to gnd	-95.15	-95.15	-95.68	0.53	dBV	meas+sim+calc
18		CNI = line 17 -15	0.00	0.00			dB	calc

Table 1: Measured, simulated, and calculated results with LL9226-C and three different op-amps in balanced mode as shown in Figure 1, $R_0 = 0\Omega$

op-amps the resulting bias input currents, managed by a special bias current cancellation mechanism, may work against each other [10]. Thus, extra cancellation current noise will be generated, which then increases the input referred voltage noise via R3 and R4.

Figure 2 shows the measurement setup. **Photo 1** and **Photo 2** present important parts of the measurement setup with the PFMA = Galvanically isolated Measurement Amp (Chapter 28 and Chapter 29 [2]) and the RIAA decoder with only $\pm 0.1\text{dB}$ deviation from the exact RIAA curve (Chapter 23.3 [3]).

General Step Approach

There are two goals:

1. Get the numbers in the tables

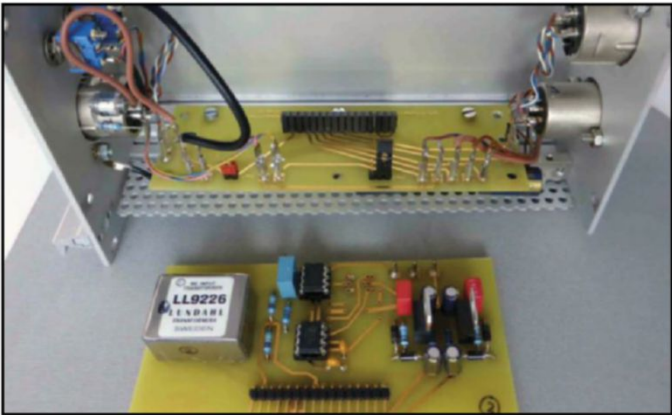


Photo 1: Test case and Amp1

2. Find the SUT’s noise production immediately
Note: The results and delta calculations are all presented in the same lines of both tables.

Fulfilment of Goal 1:
Step 1: Measure, simulate, or calculate the warm state of the whole amp chain as shown in Figure 1 and Figure 5 from Part 1 of this article (May 2021) See lines 4, 9, 14 in each Table 1 found in both Part 1 and Part 2 of the article series.

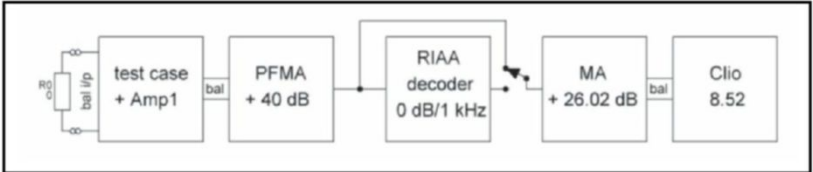


Figure 2: Measurement setup for column F in Table 1 and Figure 6

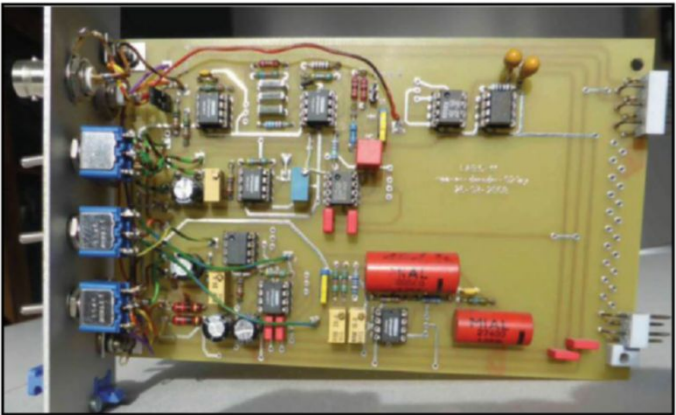


Photo 2: RIAA decoder (lower half) and encoder (upper half)

Step 2: Simulate or calculate the cold state of the whole amp chain as shown in Figure 1 and Figure 5 from Part 1 of this article (May 2021)
See lines 5, 10, 15

Step 3: The calculated deltas between Step 1 and Step 2 results are the worsening W from the cold state through warming-up of the coil resistances to the warm state
See lines 6, 11, 16

Step 4: Measure, calculate, or simulate the second gain stage with input shorted to ground.
See lines 7, 12, 17

Step 5: No matter if the coil resistances produce thermal noise or not: as long as there is an input current noise produced by the second gain stage, there will be a voltage noise generated by the transformer's R_0 & R_p and R_s & R_i —dependent un-balanced impedance $Z_{i.tot}(R_0)$ ($2 \times R_x(R_0)$ in the balanced case) times this input current noise. I called this effect the CNI. It can easily

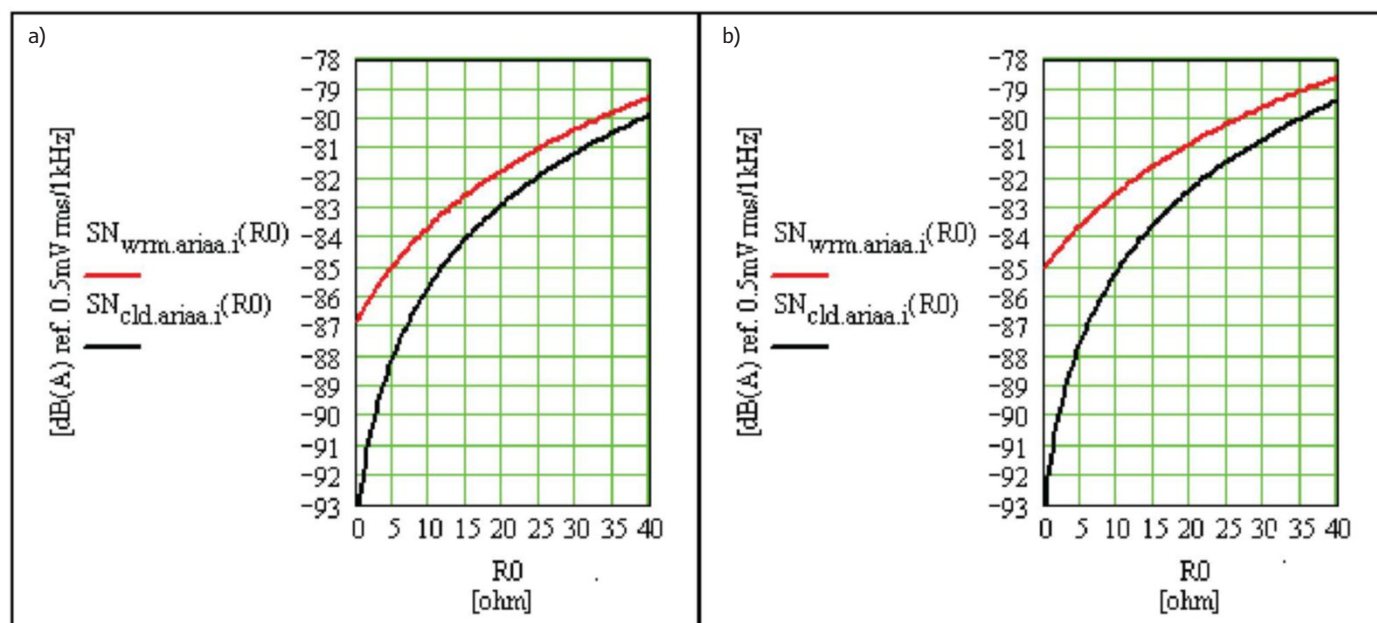


Figure 3: SN of LT1128C + LL9662-C (a); and + JT-44K-DX-ds (b)

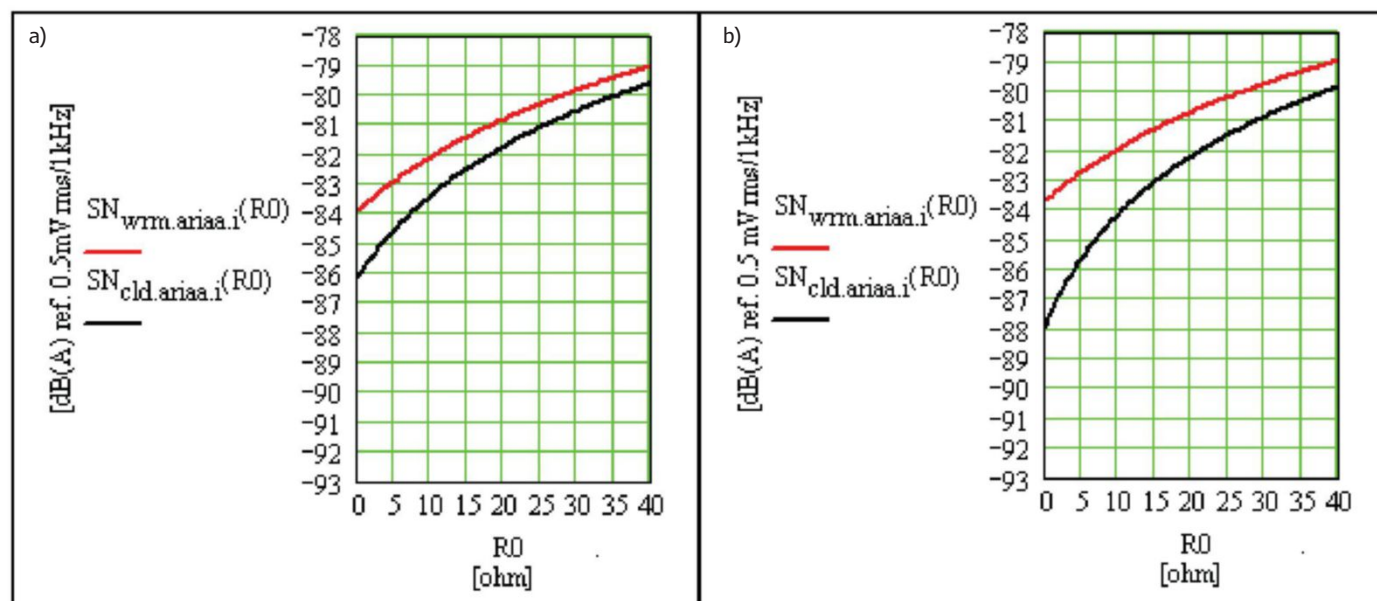


Figure 4: SN of OP37G + LL9662-C (a); and + JT-44K-DX-ds (b)

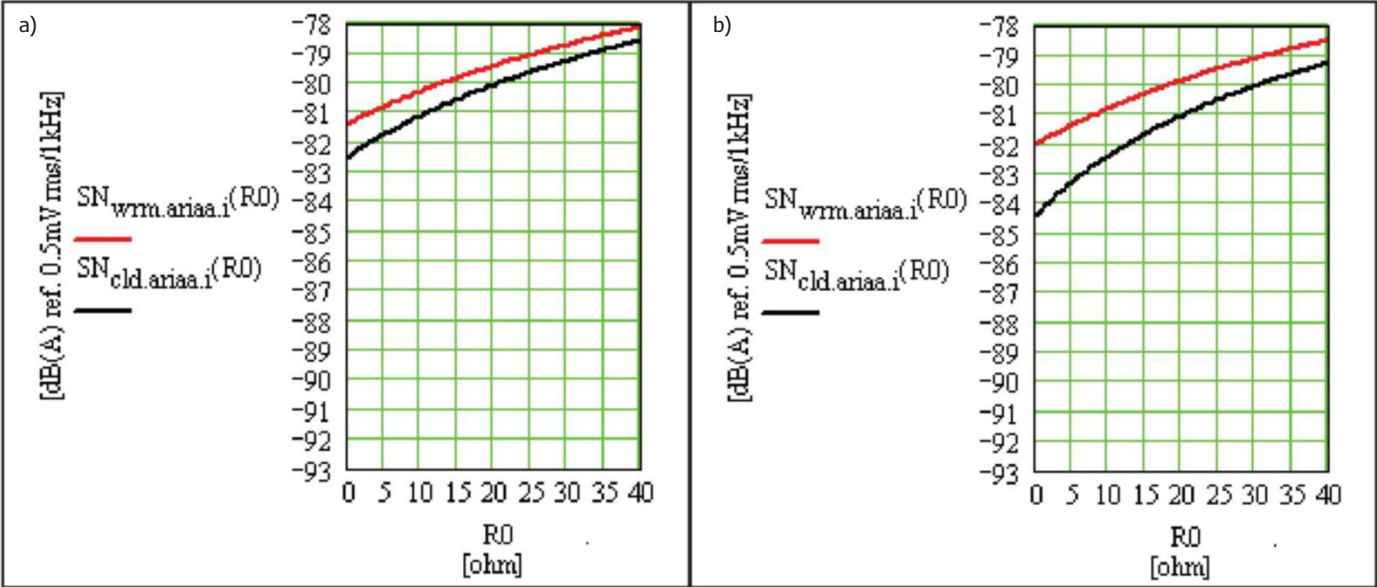


Figure 5: SN of OPA627B + LL9662-C (a); and + JT-44K-DX-ds (b)

be calculated (e.g., as shown with the tables). That is the point where the A situation (= the opposing viewpoint from the beginning of the article) becomes very questionable. In this connection, it is not the transformer itself that produces the noise; it is because of its identifiable impedances $Z_{i.tot}(R_0)$ or $R_x(R_0)$ are always part of the whole amp chain. See lines 8, 13, 18

Fulfilment of Goal 2:
Step 6: Transfer of all voltage noise results into the input referred RIAA equalized and A-weighted signal-to-noise ratios ($SN_{ariaa.i}$) of the warm and cold amp chains. The corresponding traces vs. R_0 are shown in **Figure 3**, **Figure 4**, and **Figure 5** in dB(A) with reference to a typical input signal voltage of 0.5 mVrms/1kHz (red = warm = upper trace, black = cold = lower trace). At specifically chosen values of R_0 the differences of these traces represent the Worsening Figure W in decibels (dB). The effect of W cannot be ignored and it does not change from no-RIAA to RIAA equalizing and to an additional A-weighting!

Those readers who want to follow the math should know that LT1128/LT1028 and OP37/OP27 are op-amps with 100% correlated input referred current noise densities (because of their mentioned bias current cancellation circuitry), whereas OPA627 and many other relatively old op-amps (e.g., NE5534 and NE5532) have 100% un-correlated input current noise densities. This is reflected on the MCD worksheets by the phrase “Plus correlated current noise of the op-amps taken into account” at equation (5). However, in the cases presented here the difference

between “taken into account” and “not taken into account” can be ignored. Nevertheless, care must be taken because this is not always the case.

Summary
Table 2 shows the worsening results of the survey. We’ve gathered the results of six different

1/A	B	C	D	E
2	Worsening Figures W [dB]			
3	transformer →	LL9229-C balanced		JT-44K-DX-ds single-ended
4	type of op-amp ↓	meas	calc	calc
5				
6	LT1128C	7.1	6.90	8.25
7	OP37G	2.3	2.25	4.25
8	OPA627B	1.4	1.26	2.65

Table 2: Worsening Figures W for six different cases of combinations of amplifier and SUT according to Figure 1 and from Figure 5 from Part 1 of this article (May 2021)

About the Author

Burkhard Vogel obtained his academic degree Dipl.-Ing. (Telecommunications) in 1973 from Darmstadt University of Technology, Germany. He then decided to follow a general management career path in the high-tech and IT industry in Germany, Switzerland, and Austria. In the recent decades, fruitful discussions on tube audio and electronic noise topics with customers, employees, colleagues, and friends led to a broad range of works on these issues. They were published in several magazines including *Electronics and Wireless World* (EWW), *Linear Audio* (LA), and *Design & Design* (D&D), together with many additional comments in EWW, LAs in EWW, LA, and D&D. Finally, edited by the Springer Nature publishing house, he has six books (currently number 7 is “under construction”) that round up the gained knowledge.



op-amp/input transformer circuits and we could prove three of them by measurement results in column C and via traces shown in **Figure 6** by application of Figure 2. At, for example, 1kHz, the deltas between the red and green curves (or

between the black and orange RIAA equalized ones) become the measured worsening value given in Table 2's box C-7. Of course, it is practically the same value already calculated in Figure 4a at $R_0 = 0\Omega$ or with equation (8) from MCD WS 2.

Column E gives the calculated worsening figures with the JT-44K-DX-ds transformer. The difference for the OP37G case is shown in the calculated **Figure 7**. It presents all possible noise curves we can gain from Figure 5 (shown in Part 1 of the article, May 2021)—if we would measure them. At any chosen frequency, the difference between the two corresponding and rather parallel curves becomes always 4.25dB, the Worsening Figure W. With a difference (or Worsening Figure) of 2.25dB with the LL9226-C the same collection of traces is shown on MCD WS 2. The "+66" represents the overall gain in Figure 2. It shifts the set of curves to the same position shown in Figure 6.

Consequently, the following statements succinctly describe the findings about the noise impact of MC SUTs connected to MM phono-amps:

1. As long as there are coil resistances, SUTs produce at least thermal (Johnson) noise, no matter which kind of material they represent and not taking into account other noise sources (e.g., mechanical).

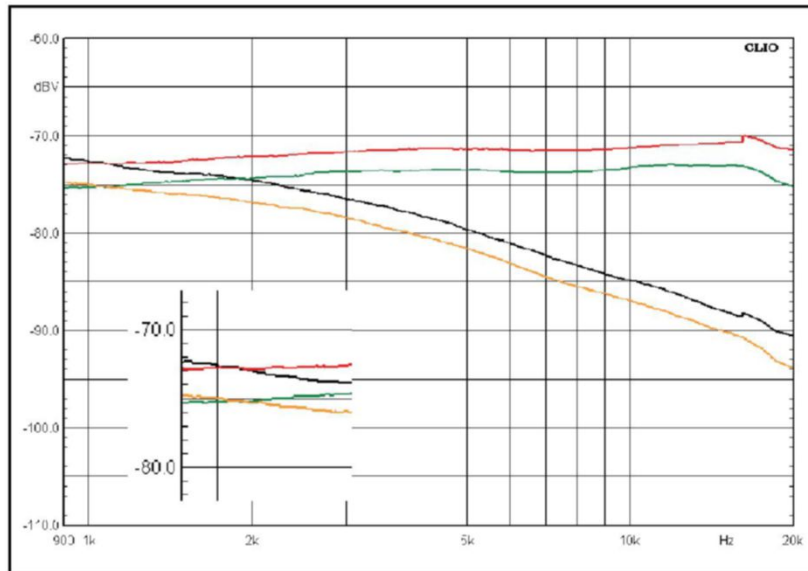


Figure 6: Measured output noise traces of Figure 1, above 900Hz, plus a zoomed region at 1kHz, and LL9226-C connected to a balanced MM phono-amp input: non-equalized cold (green = lower flat trace) and warm (red = upper flat trace) version, RIAA equalized cold (orange = lower bent trace) and warm (black = upper bent trace) version

Project Files

The Mathcad Worksheets found in the Supplementary Material section of the audioXpress website present the complete math around the two different step-up transformers (stand-alone under points 1. ... 4.) and their connection to two different and active amplification methods, un-balanced and balanced (under points 5. ... 12.):

MCD-Worksheet 1: JT's transformer JT-44K-DX configured alone with datasheet values of its DC resistances and turns ratio plus its connection to an OP37G driven un-balanced MM phono-amp input

MCD-Worksheet 2: LL's transformer LL9226 "C" configured alone with measured values of its DC resistances and gain plus its connection to an OP37G driven balanced MM phono-amp input

To download the Mathcad Worksheets, visit <http://audioxpress.com/page/audioXpress-Supplementary-Material.html>

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2. Additionally, as long as the following MM phono amp produces an input noise current, there will be a current noise impact CNI.
3. Compared with a stand-alone situation, the input referred noise of a following MM phono-amp lowers the noise impact of the SUT. However, together they decrease the input referred SN of the amp chain.
4. Compared with an input load of $R_0 = 0\Omega$ any value greater than 0Ω lowers the noise impact of the SUT. However, R_0 's extra noise worsens the input referred SN further.

=> Unfortunately, the noise impact of a SUT never disappears!

Final Remarks

Assuming we take the same SUT and we compare the single-ended circuitry with a fully balanced one, we will find that the single-ended amp chain shows a bit better input referred SN results than the balanced amp chain. This can easily be checked now by application of LTSpice or by my math approaches. Then, careful measurements should lead to equivalent results.

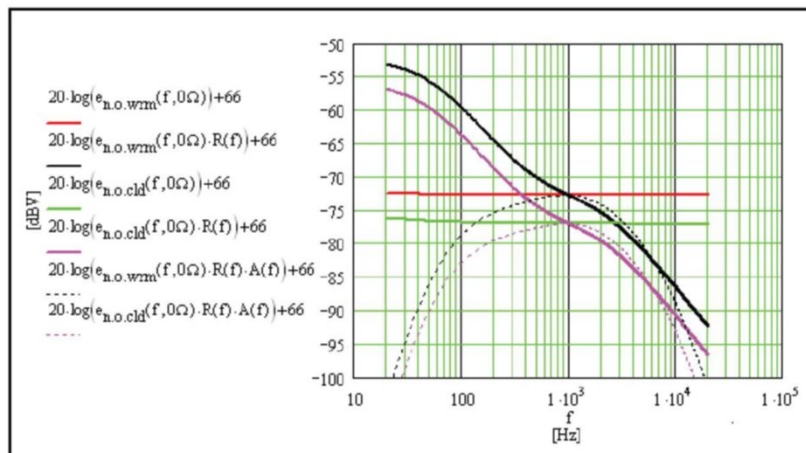


Figure 7: Calculated output noise produced by the JT-44K-DX-ds connected to a single-ended MM phono-amp input: non-equalized cold (green = lower flat trace) and warm (red = upper flat trace) version, RIAA equalized cold (magenta = lower RIAA trace) and warm (black = upper RIAA trace) version; RIAA equalized plus A-weighted cold (dotted magenta lower trace) and warm (dotted black upper trace) version

Readers who miss Spice-based schematics and models of op-amps should know that modelling of the noise of older op-amp types will be treated in depth by the content of my next book: *Slopes and Levels—Spice Models to Simulate Vintage Op-Amp Noise*. Diving deeper into this world would have greatly increase the size of this article. 



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The Science and Physics of High-End Mic-Preamplifier Design (Part 5)

The Best Practice

By
Mike Meehan

In the previous articles, Mike Meehan considered the physics that play a fundamental part in microphone pre-amplifier design, explored its inhabitation in the real practical world to retain the fidelity of the mic signal along its path from transducer to output, and looked at what (theoretically) contributes to perfect active mic preamps. Having looked at “the bigger picture,” he concludes his article series with the golden rules of low-noise, mic preamp design.

You can’t polish mud—but you can maybe find diamonds beneath it.

I like this quote. A lot. But a mic preamp is not a diamond. And you definitely can’t polish it after the fact. It is the job of the designer to find the mother lode through mathematics and logic, and good practice, and trial-and-error, and exhaustive bench-testing, and get it right as soon as the mic signal hits the preamp input.

To briefly recap what we have discussed throughout this five-part article series, a top-flight preamp comprises the following:

- RFI protection (two or more cascade stages)
- Phantom powering circuitry
- Phantom power fault protection
- Optional Mic/Line input switching
- Z_{SOURCE} impedance matching network(s)
- Mic preamp IC (and gain control network)
- Differential line driver stage
- Good power supply
- Scrupulously-careful PCB design (this is absolutely vital)

Before looking more closely at the various sections, it is important to realize that the topologies presented here, and any specifications subsequently intimated, are wholly dependent both for success

of performance and veracity of specification on peripheral component choices and physical layout. The significance of both cannot be underestimated. At this critical level of design, the difference in performance between a layout adhering to a strictly-defined, carefully-considered design mantra, and one that translates verbatim from an electrical schematic, is profound.

Scrupulous attention to detail, and full awareness of the unwelcome intrusion of the real-world environmental constraints—coupled to physical limitations of components and layout—differentiates between a good design and an excellent one, and an excellent design and a world-class one.

Unfortunately, even the most sophisticated SPICE modeling techniques omit real-life effects, and since no model can simulate all the parasitic effects of discrete components, and easily that of a PCB layout, prototypes must be built and test-bench proven before production. Auto-router PCB layout software can cause havoc with grounding arrangements so that PCB vias should be avoided on the board, if possible—certainly for signal paths. Every connection within the finished design must be scrutinized for integrity, every component type, and value justified on cost, performance, and applicability criteria, and every location of it on a physical board layout evaluated thoroughly against key design pointers.

The Art of Low-Noise Design

- High common mode rejection ratio (CMRR) capability (an absolutely vital aspect of any design) often depends as much on *absolute symmetry of component placement* as on exact component value match—thus, a precise mirror image of both PCB track layout and physical component placement across both of the legs that comprise the differential input is likely required. (Capacitive differences due to mismatches in routing/wire length may cause unequal noise pick-up between the input differential pair, and affect overall CMR.)

It is easily forgotten that any resistors used to feed phantom power to the microphone—typically +48V via 6k81 values—also must be very closely-matched or CMR performance will be compromised. 1% resistors in the phantom supply network should be matched to 0.1% tolerance to provide the best performance, with 1% 100ppm and 50ppm metal film resistors preferred to the more common 5% 200ppm tempco carbon film types—they provide greater initial accuracy, better long-term stability, and higher stability at extremes of temperature.

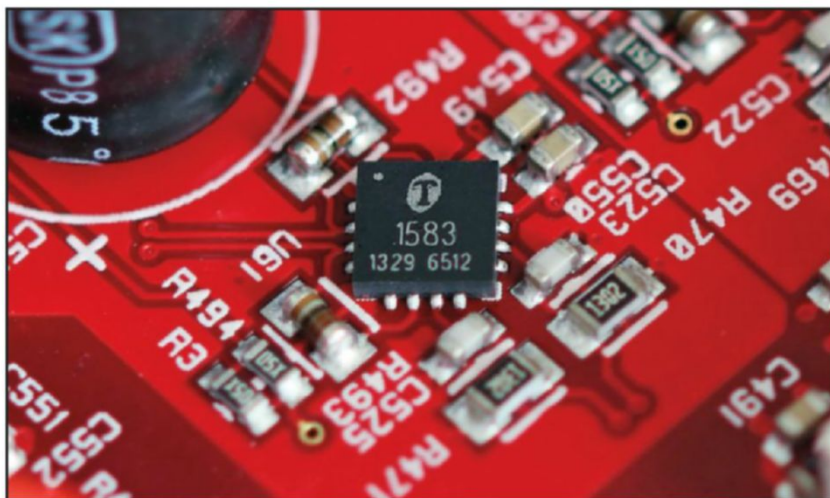
- Consider every earth connection as a possible costly cause of loss of performance. Calculate ground error signals for worst case/highest load scenarios and use the above guidelines to mitigate against them. Remember, as far as PCB traces are concerned, short and thick does the trick.
- Connections to the differential signal lines—resistors feeding phantom power, RF filtering via C or RC layouts to ground—sometimes unwittingly provide RF paths into the system via the earth connection. Great care must be taken with the routing and cross-section of ground traces, with star earthing techniques where at all possible and the “ground-follows-signal” philosophy where this topology proves impracticable or impossible.
- Transistor junctions rather unfortunately behave as effective RF demodulators—guard appropriately against this eventuality. The devices, whether discrete or internal to ICs, often have f_T values extending up to 300MHz—use spectrum analysis up to the low RF regions to ensure that designs aren’t oscillating there.
- Rigorously apply proper RF design techniques throughout the design, with copious high quality decoupling for both AF/RF.
- Adopt a comprehensive ground plane system.
- Use the very highest quality components in all parts of the design—specify military-grade parts throughout, and hand-match via an LCR bridge or similar any passive components (capacitors



German microphone manufacturer Neumann recently introduced the V402, a new state-of-the-art dual-channel microphone preamplifier with integrated headphone monitoring. This is Neumann’s first-ever stand-alone microphone preamp—although the company has created several generations of top-quality preamp modules for its mixing consoles, such as the now legendary V476B of the 1980s. Built to much higher standards, the V402 preamplifier is conceived as a perfect complement to all high-class studio microphones, including Neumann’s.

and resistors). Bulk or metal foil resistors are available with tempcos of <1ppm, low reactance, and excellent tolerance (to 0.005%), but are expensive; wire wound provide a good compromise, offering excellent TCs, close tolerance (and lowest noise), although reactance and bulk must be considered.

- Capacitor quality plays an important part in overall signal quality—choose types appropriate to the task in hand. Use by-passed non-polarized



The THAT 1583 is a versatile, high-performance current-feedback instrumentation amplifier (CFIA) suitable for differential microphone preamplifier and bus summing applications. It improves on previous, traditional current-feedback designs (viz. THAT’s 1510 and 1512) by offering a more versatile configuration that can yield lower noise at low gains, lower distortion overall, and higher slew rate. Amplifier gain is determined by three external resistors and the 1583 is designed to mate with THAT’s series of Digital Preamplifier Controller ICs to produce an optimized, digitally controlled preamplifier.

audio-grade aluminum electrolytics where high values ($>10\mu\text{F}$) are required in the signal path, (or use more-expensive high-value film PTFE or polypropylene types, if available). So use electrolytic capacitors of the required large value, then add a $1.0\mu\text{F}$ or $0.1\mu\text{F}$ (or both) film capacitor in parallel, the theory being that low frequency will be handled by the large electrolytics, and high frequency (where the smearing would be most audible) will be handled by the small film capacitors. Elsewhere, use high-grade metallized polypropylene film or similar (polystyrene), except for decoupling, where high-quality ultra-stable COG dielectric ceramic types should be used in conjunction with bypassed low-impedance electrolytics—if possible, omit

them from the input signal path via switching whenever possible. Having low Z_{INTERNAL} , ELNA Cerafine, or Dynacaps for the PSU, and SILMIC's (or Solen's) types for the signal path are exceptional choices. (And remember—any capacitor's inclusion is responsible for a number of sonic (dielectric absorption) and operational shortcomings, including failure of input devices because of a number of mechanisms, namely current inrush when transducers are unplugged, which causes Zener breakdown of active devices directly connected). Note also that the use of higher voltage/same value capacitors (e.g., 105°C rated rather than 85°C types) can often reduce distortion by an order of magnitude (and improve reliability).

- Avoid confining the specification of quality components to only those residing directly in the signal path—those elsewhere affect the signature of the audio.
- Use a quality fiberglass PCB having foil thickness and pad size sufficient to allow damage-free desoldering of failed components that aren't socketed.
- Adopt a minimalist approach throughout—avoid including features likely little-used by sound engineers but which add cost, complexity, and another means of introducing coloration to the signal (and possible failure points).
- Deploy only the very best turned-pin types where the design mandates them (sockets add unwanted inductance and capacitance). Remember that ICs or transistors driven hard produce heat, worsening noise performance. (A 50°C increase in temperature worsens noise performance by some $1\text{nV}/\sqrt{\text{Hz}}$, and diminishes slew rate—and GBP too). ICs soldered directly to the PCB—particularly metal can packages—dissipate heat much better and consequently run cooler and quieter—so ensure the casing has plenty of side and top-and-bottom ventilation slots.
- Avoid wires where at all possible in the design—they provide an unwanted variable.
- Do not run traces carrying low-level sensitive signals next to PSU lines.
- IP/OP connectors should be soldered directly to the board (or via inter-PCB soldered Molex connectors).
- Consider the microphone connecting cable and any connectors as an intrinsic part of the CMR problem—even slight mismatches in the individual core lengths of a twisted pair and other often-neglected factors negate by an appreciable amount careful physical layout and component matching of designs. Test CMRR via kosher



The Solid State Logic (SSL) Alpha VHD-Pre is considered one of the best solid-state mic preamps in the market today, allowing for up to 75dB of gain. Available in rack-mount and 500 series modules, it is an example of a very flexible preamplifier thanks to its Variable Harmonic Drive (VHD) circuit, originally designed for the famous SSL Duality studio console.

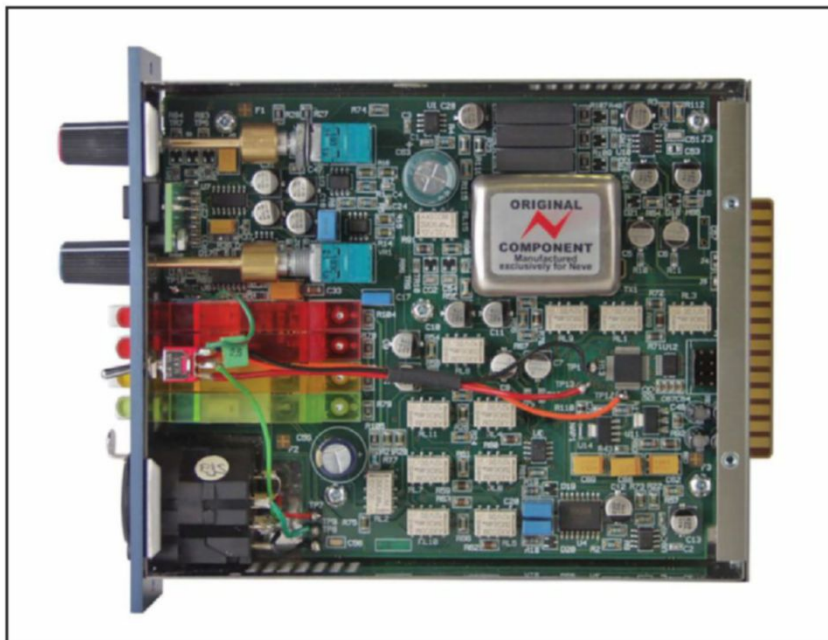
About the Author

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



scenarios (i.e., with real microphone cables and patch-bays—and over authentic distances).

- Maximize headroom by operating active components at the highest voltage levels recommended by manufacturers.
- Optimally-engineer the PSU for low-noise/ proper tracking on bipolar rails. Also, don't under-estimate what current transients will demand of the circuitry if the sound is not to be smeared during loud passages, particularly when circuitry is driving low source impedances over long distances, and several input stages are being fed. 4,700 μ F, 35V audio-grade capacitors with low ESR are a good choice in the power supply unit (PSU). Voltage doublers can be used to derive the phantom +48V power rail from a center-tapped toroidal 18-0-18V transformer.
- Expend rigorous design effort on the PSU. Falling power supply rejection ratio (PSRR) of op-amps with rising frequency (at 6dB/octave), coupled to regulator ripple ratio, and regulator input capacitor size, insidiously add noise via supply rails. Analyze these well into the megahertz (MHz) region—IC regulators have been known to oscillate at 3MHz, causing an unpleasant



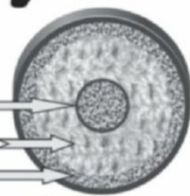
Considered one for the world's greatest recording consoles—the Neve 88RS also featured the most appreciated microphone preamps, now available to any project studio in convenient 500-series format modules.

“sheen” to be added to the audio, but which escapes detection in AF analysis. Likewise, supply bypassing via RC decoupling can be effective but load current changes will modulate the voltage at

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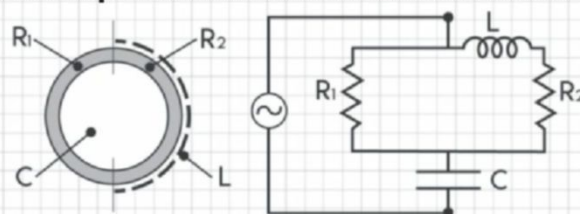
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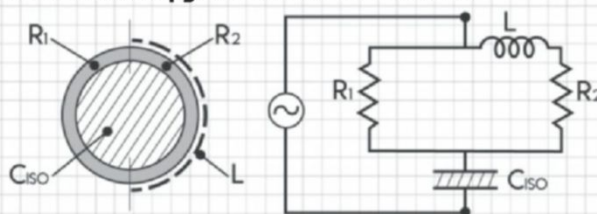
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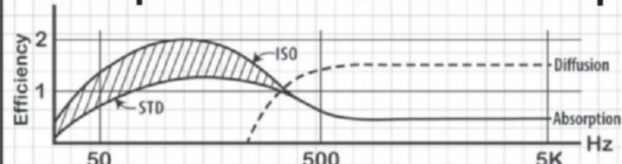
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The Focusrite ISA One portable microphone amplifier combines the circuits originally designed for the famous Focusrite Forte studio console, designed by Rupert Neve for the brand. The microphone preamp survived and is today a common feature in many studios all over the world, using the same Lundahl LL1538 input transformer and network.

the op-amp's supply pins: keep tracks short and fat—decouple within 0.1" of the device.

- Wherever possible, operate a star distribution for power as well as ground, maximize track the cross sectional area (CSA), and ensure that any switched indicators, relays, and so forth are fed from a dedicated supply line (i.e., not shared with sensitive audio circuitry) or use another PSU. Switched Mode PSUs (SMPSUs) deservedly have their niche—it is not high-end low-noise audio. Whatever the commercial attraction (particularly for the high voltage phantom power supply rails), their use must be avoided if at all possible.
- Use custom audio toroidal transformers—these offer many benefits over IE-core types: smaller footprint, easier mounting, reduced noise level, lower magnetic flux gathering, and inherently lower stray magnetic fields. Additional silicon-iron or mumetal shielding provides fuller electrostatic and magnetic immunity.
- The IEC/NEMA mains inlet should be an RFI filter type.
- Consider using an outboard PSU, which is somewhat less-than-elegant, but removes a hum creating component from the preamp enclosure. The dual analog supply rails, phantom rail, logic supply, and 0V can be accommodated on a 5-pin Neutrik XLR connector

Crescendo

Noise is the all-pervading enemy of electronic circuitry, with increasing density of usage of the electromagnetic spectrum exacerbating further the problems in this area. In audio and other fields (notably RF and satellite communications), the first device that the input signal contacts determines noise (and distortion) performance from then onwards, with any noise generated locally in this area of the audio path tending to swamp noise in succeeding stages.

Thus, failure to use a good low-noise, low-distortion design, or perhaps worse still, to use a

good design but realize it badly in practice, notably through inattention to the finer details (e.g., mechanical layout/physical construction/passive component choice) can completely negate the benefits to be had from downstream audio path integrity.

The subject of low-noise audio design is extensive—thus, deep knowledge about it appears scattered and sometimes not easily accessible, affording the skill an arcane quality. This article has not pretended to address anything like the complex and comprehensive subject that is professional low-noise analog audio design. Rather, the intention has been to demonstrate that, with the right tools and more than a modicum of knowledge (attainable and easily-comprehensible by the competent engineer), good design is readily achievable. Topflight design, however, is more elusive, and requires careful choice of component, extensive bench-testing, thorough product evaluation, and a certain degree of trial-and-error.

After the actual microphone itself, the microphone pre-amplifier constitutes the most important area of design in the audio input signal chain, and designs must address this, regardless of cost.

I leave the last pithy words to the acclaimed British audio designer, Ben Duncan:

"Readers may do well to sharpen their awareness of the fact that audio products for music creation are amongst the most demanding and rewarding things that human beings can make. Competitive functionality and good ergonomics are high arts in themselves, as is ensuring high sound quality."

And so, ending on that good high note (like the very best concert)—I couldn't agree more.

Author's Acknowledgments:

A series of in-depth technical articles of this nature could only be written with the kind cooperation and extremely generous help and indulgence of a number of key senior pro-audio industry folk. In no particular order, grateful thanks are extended to Gary Herbert, CTO THAT Corp. (and designer of the THAT1580 IC); Dr. Rosalfonso Bortoni, Senior Applications Engineer, THAT Corp. (in particular for his patience with me in answering my endless stream of questions); and Ken Nevard, Marketing Manager, THAT Corp.; Larry Hathaway, Production Manager, Jensen Transformers Inc.; John Hardy, the President at The John Hardy Co.; Ian Du Rieu at The Leon Audio Co.; Uwe Beis, BEIS Electronics; and Pete Jobur, SMS Electronics, Ltd.

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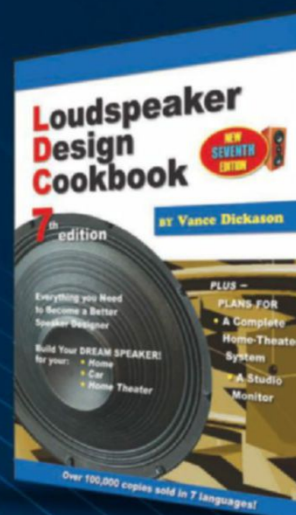
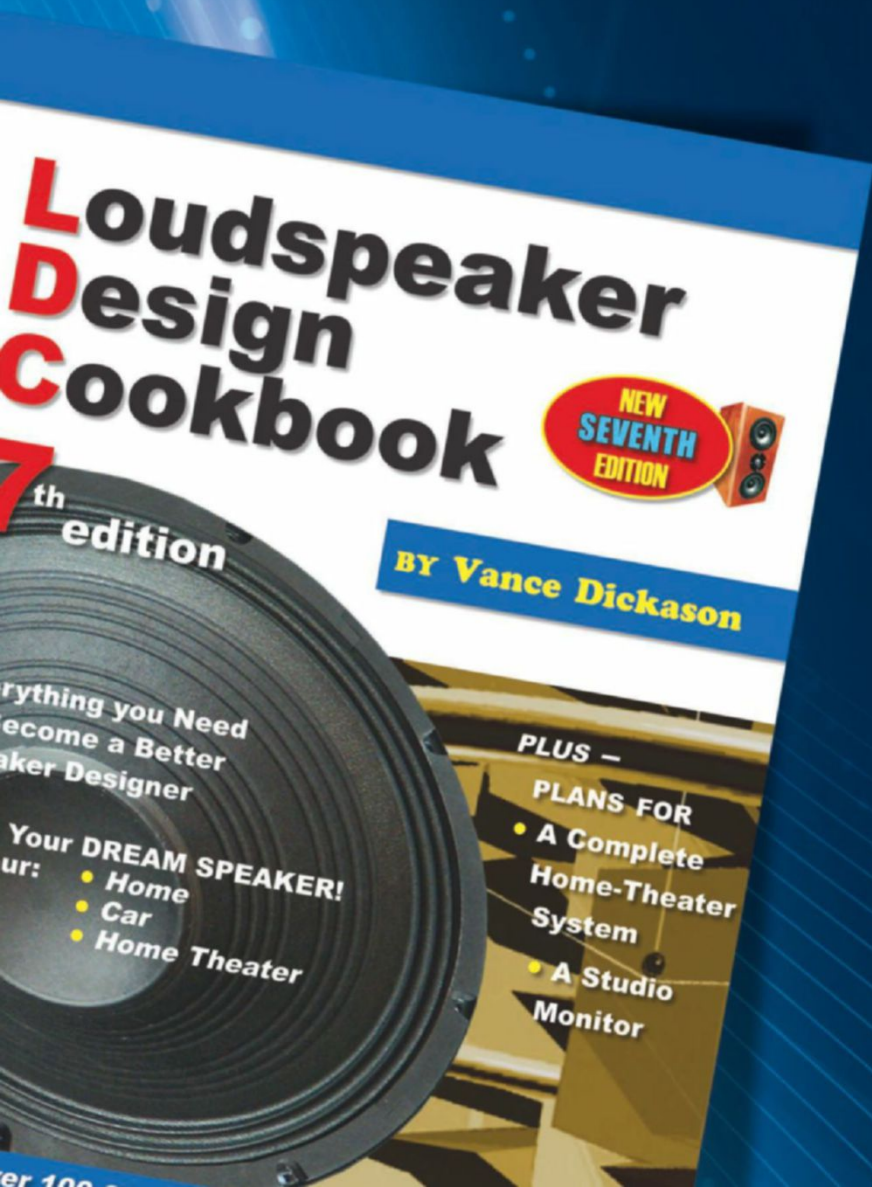
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Class-D Amplifier Output Filters For Next-Generation Audio Devices

As consumers demand even more audio performance in ever-smaller packaging, Infineon's new MERUS™ family of Class-D amplifiers uses an innovative multilevel technology to deliver new levels of efficiency across the whole audio frequency spectrum while significantly relaxing the requirements for filtering. Together with the reduced BOM that comes from simplified filters, these improved efficiency levels open up new possibilities for the audio system designer.

By
Rien Oortgiesen

(Application Engineering Manager for Class-D Audio,
Infineon Technologies Denmark)

The availability in recent decades of low-cost, fast switching MOSFETs has driven the popularity of switch-mode, or Class-D amplifiers, which have significantly lower power consumption, smaller size, and lower heat generation than the previous generation of Class AB amplifiers. With the growing trend toward more compact and more portable audio devices, developers are increasingly leveraging these benefits of Class-D amplifiers.

However, although these amplifiers can achieve efficiencies of 90 percent or better at high output powers, this drops to around 50 percent at typical music listening volume levels. Additionally, traditional Class-D amplifiers require LC output filters to reduce electromagnetic interference (EMI) and improve electromagnetic compatibility (EMC). Further filters are also often required on the switching side of the output filter to dampen switching edges and further improve EMI and EMC. These constraints on efficiency and the additional costs, power consumption, and PCB real-estate requirements of the filter components impose limitations on the benefits of using Class-D amplifiers.

These limitations have motivated Infineon's design engineers to develop an innovative new generation of devices, the MERUS multilevel

Class-D amplifier series (MA12040, MA12040P, MA12070, MA12070P), which are highly efficient at all listening levels. Along with these improved efficiency levels, the enhanced EMI and EMC management of the MERUS amplifiers enables filter-less operation for many types of applications. For more demanding scenarios, such as long speaker cables, these amplifiers can still operate with small and inexpensive filters, leading to a reduced bill of materials (BOM) and a more compact product footprint.

A New Generation of Class-D Amplifiers

The Infineon MERUS family uses a proprietary, multilevel switching technique to achieve low power consumption at idle and near-idle operations as well as to deliver exceptional EMI and EMC performance. In many cases, the MERUS series of audio amplifiers can drive speakers directly, with no need for output filtering, enabling lower application costs and smaller product sizes. Even where filtering is necessary—for longer speaker cables, for example—the filters can be much simpler than for traditional Class-D amplifiers—again leading to lower costs. On top of this, the superior efficiency of these amplifiers translates directly to a massive reduction in heat generation—meaning no need for heatsinks,

even for higher-powered applications. Simply put, this new breed of multilevel Class-D amplifiers gives designers a head-start in the quest to meet customer demands for high-quality audio in ever-smaller packages.

The MERUS audio amplifiers use a multilevel half-bridge power stage as shown in **Figure 1**, with each half-bridge having four MOSFETs (as opposed to the two used in traditional Class-D amplifiers). Each of the four MOSFETs is driven by an individual PWM input and a “flying” capacitor (C_{FLY}) between the top and bottom MOSFETs (constantly charged by separate circuitry), maintains a fixed voltage potential, effectively functioning as an additional supply rail. This design enables the half-bridge to generate three PWM output levels: 0 V, $\frac{1}{2}$ PVDD, and PVDD at the output switching node.

By establishing multiple PWM output levels from a single supply voltage, the half-bridge stage offers significant flexibility to the designer, allowing the amplifier to be configured for optimal power performance across a broad spectrum of audio applications. In fact, the full potential of this multilevel technology is realized in the bridge-tied load (BTL) configuration, where two input and two output terminals are used per channel (**Figure 2**).

With this configuration, five output levels are presented to the speaker load at a quadrupled frequency, meaning smaller individual voltage steps (**Figure 3**), giving a more faithful representation of the input waveform than is possible with traditional Class-D amplifiers. At the same time, the BTL configuration gives a doubling of the voltage swing to the load (compared to a single half-bridge configuration) and so provides four times as much output power. The five output levels of the BTL configuration also enable low near-idle power consumption and beneficial noise properties, removing the need for a bulky output capacitor.

Significantly Reduced Filtering Requirements Bring Huge Benefits to the Designer

Multilevel switching also leads to a considerable reduction in ripple current in the audio system output compared to traditional Class-D amplifiers. With the inherently more efficient switching pattern of the three-level amplifier, there are three states where out-of-band switching residuals are entirely canceled. At $-\frac{1}{2}$ PVDD, 0 V, and $+\frac{1}{2}$ PVDD, the two half-bridge outputs are either both at 0 V or are perfectly “mirrored” 50 percent duty-cycles, reducing the ripple current in the audio system output by a factor of 4. With 5 level switching, there

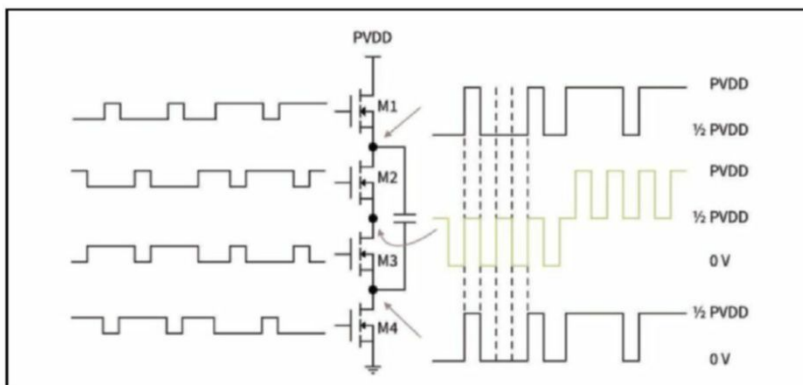


Figure 1: Each half-bridge in the MERUS amplifiers uses four MOSFETs and a “flying” capacitor to generate a three-level output.

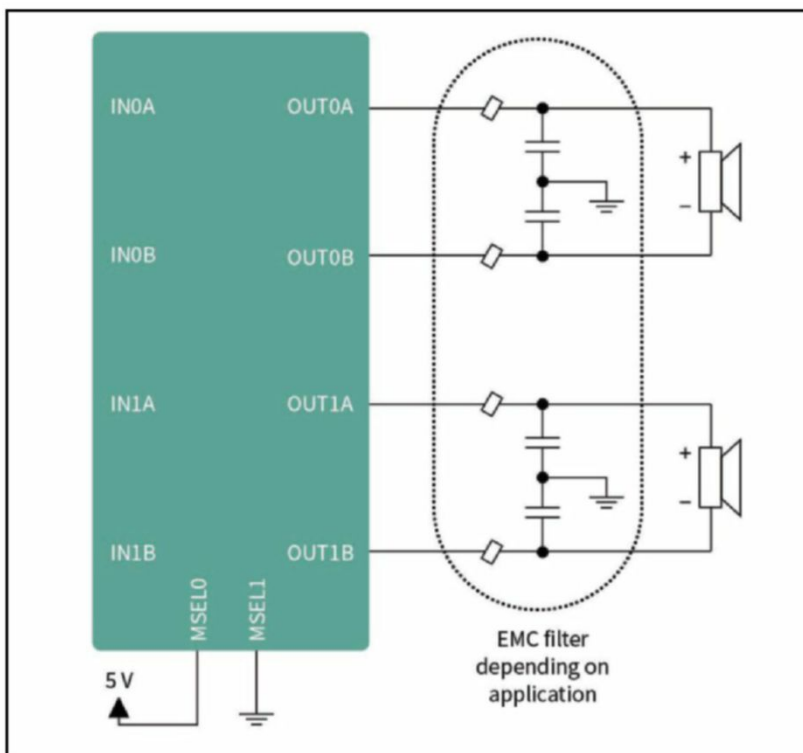


Figure 2: The full potential of multilevel technology is realized in the BTL configuration.

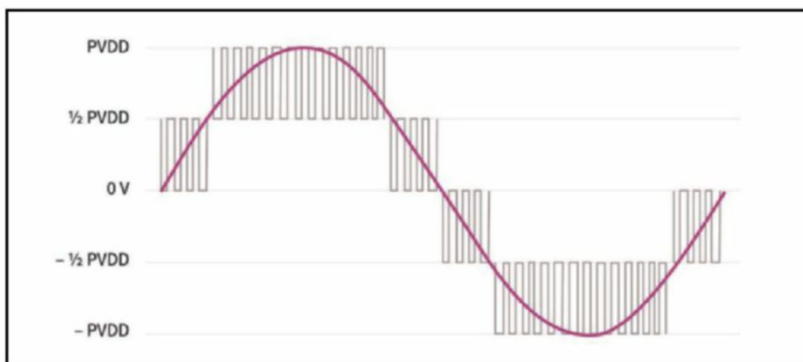


Figure 3: The BTL five-level output signal eliminates external output stage filtering and minimizes the overall power loss, with an output signal that closely approximates the input.

are five such states, giving a further reduction in output ripple current (**Figure 4**). This reduced ripple current removes the need to suppress out-of-band switching noise, enabling the amplifier to operate either with a simple ferrite filter or small LC filter or, in many cases, with no filter at all.

In traditional Class-D amplifiers, LC filters are required to remove the high-frequency switching component of the output signal, passing only the 20Hz to 20kHz audio frequency range to the loudspeaker, and also to improve efficiency by limiting the residual ripple current at idle conditions.

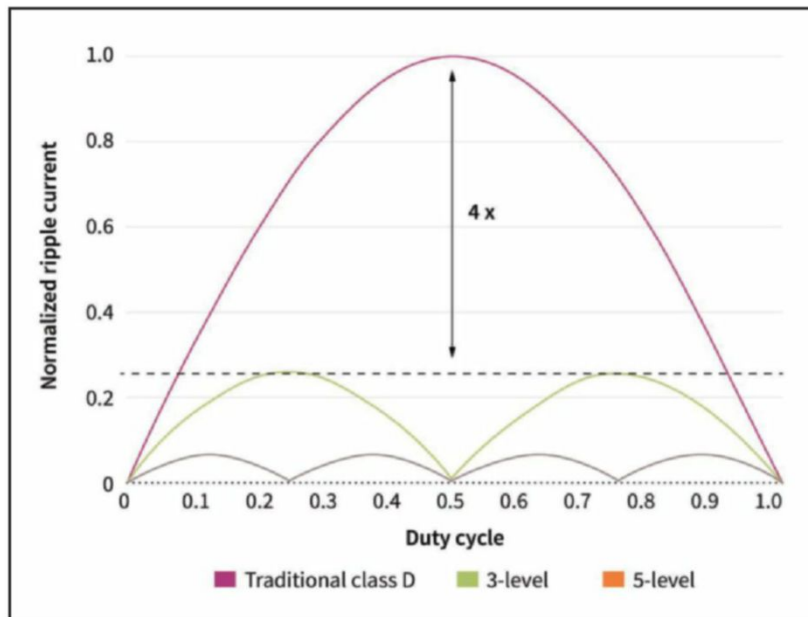


Figure 4: Multilevel switching reduces ripple current by a factor of 4 and better, compared to traditional Class-D amplifiers.

	Ferrite Filter	LC Filter 1	LC Filter 2
Speaker cable length	<60cm	>= 60cm	>=60cm
EMI	Good (***)	Best (*****)	Excellent (****)
Footprint area	34mm ²	320mm ²	300mm ²
Corner frequency	–	Normal (<100kHz)	High (>1MHz)
P _{LOSS}	Negligible	Small	Negligible

Table 1: Filtering requirements are dependent upon the application.

About the Author

Rien Oortgiesen is the Applications Engineering Manager for Class-D Audio at Infineon. Rien's experience includes IC circuit design, mixed signal system design, and embedded audio solutions and applications. Rien holds a Master of Science degree in electrical engineering from Twente University in The Netherlands with a specialization in analog-mixed-signal IC design. Rien became part of Infineon Technologies as part of the acquisition of Copenhagen based company Merus Audio.



However, filters come at a cost, with the associated components contributing to an increased BOM, taking up additional board space, generating more heat, and impacting the end product's physical size.

Therefore, the reduced filtering requirements of the MERUS products deliver tangible benefits to the application designer, including reduced component counts and lower power losses. Since cooler operation is a positive knock-on effect of lower power loss, there is no need for heatsinks. The PCB itself often provides enough heat conduction, enabling reduced product form factors. The resulting cooler operating conditions also tolerate sustained high-volume levels and lead to less thermal aging—and consequently overall higher application reliability.

Filter Selection is Determined by the Needs of the Application

The actual filtering requirements of a MERUS design are determined by the application's specifics, as summarized in **Table 1**.

Output filters can be divided into two types: ferrite filters and LC filters. While the ferrite filter is smaller and cheaper than the traditional LC filter, it has lower attenuation levels and is therefore not suitable for every Class-D amplifier or speaker cable lengths. **Figure 5** illustrates the potential space-saving benefits of ferrite filters. As shown in Table 1, applications based on MERUS technology will pass radiated emissions tests with a ferrite filter only, for speaker cable lengths up to 60cm. In most cases, a ferrite filter will be sufficient for active speaker applications with pre-determined and relatively short speaker cable lengths.

The BLE32PN300SN1L is an example of a ferrite that is well-suited to high-performance and high-power applications based on MERUS technology. Applications using this device have been shown to pass radiated emission tests with a margin of 5dB or better across the 30MHz to 1GHz range, with speaker cable lengths up to 40cm. (The tests also pass for 60cm cables, although with reduced margins). This performance comes with a cost, however, as this device is expensive compared to other ferrites. If using an alternative device, care must be taken to ensure that the ferrite can handle the required power and current with acceptable levels of distortion and power loss.

For these cable lengths, Infineon also recommends the use of a 1nF, 50V COG/NP0 type capacitor in the output filter for best audio performance. Although lower cut-off frequencies can be achieved by increasing the capacitance, values greater than 1.5nF risk higher power losses and output noise.

Where speaker cables are longer than 60cm, Infineon recommends the use of LC filters, and the filter design is influenced by both the EMI and the impedance seen by the multilevel amplifier. The LC1 filter has the best EMI performance (Table 1) and works best with designs that use multiple amplifiers. On the other hand, for single amplifier designs, the LC2 design meets the EMI requirements while exhibiting lower PLOSS and also offering the benefit of a smaller footprint.

Next-Generation Audio Devices

As traditional Class-D amplifiers are hitting the limits in terms of efficiency and size for next-generation, portable audio devices, Infineon's new MERUS family of Class-D amplifiers uses an innovative multilevel technology to deliver new levels of efficiency across the whole audio frequency spectrum while significantly relaxing the requirements for filtering. Together with the reduced BOM that comes from simplified filters, these improved efficiency levels open up new possibilities for the audio system designer. Higher efficiencies and reduced component count translate into lower heat generation and reduced

space requirements. With fewer components and no need for heatsinks, the audio functionality can comfortably fit into smaller enclosures. Higher efficiencies also mean lower power consumption, supporting more prolonged battery operation, and cooler operation reduces thermal aging, improving product reliability and lifecycles. Finally, reduced BOM brings down product costs, driving the profitability and competitiveness of the end product.

With the MERUS family of amplifiers, Infineon is behind the delivery of the next generation of audio innovation. To explore how contemporary audio designs can benefit from Infineon's cutting-edge solutions, please visit the MERUS website. 

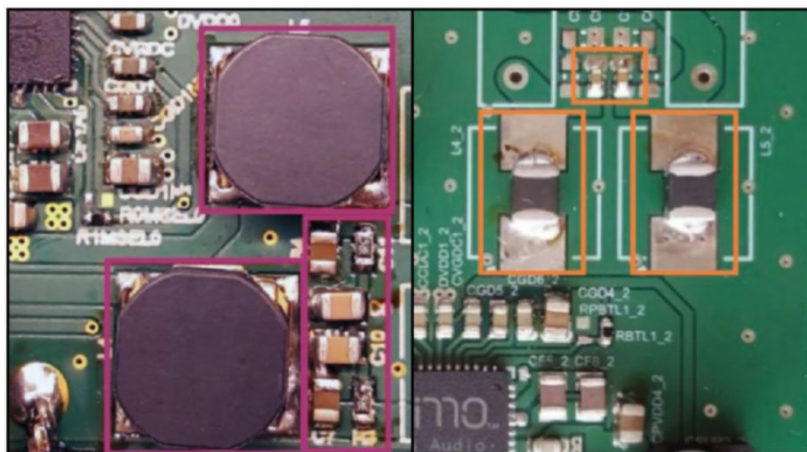


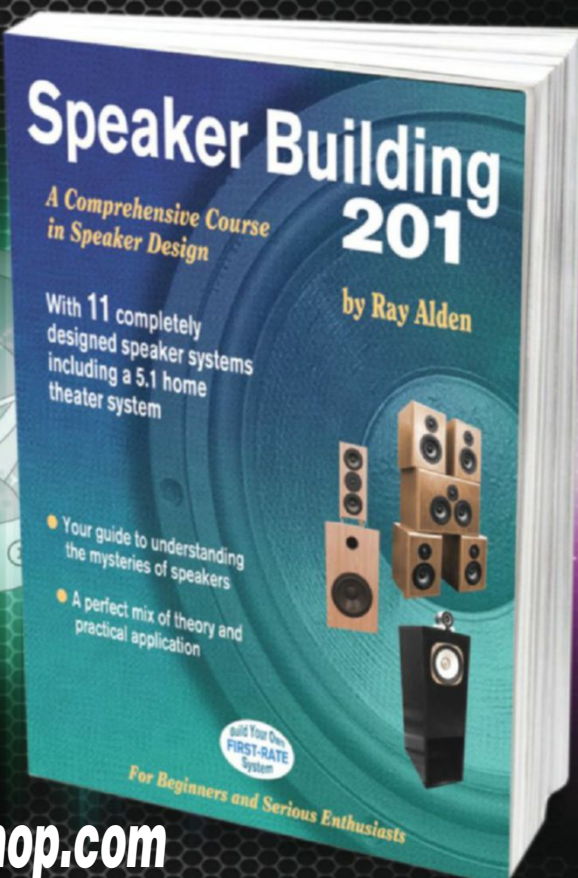
Figure 5: LC filter (purple) vs. Ferrite filter (orange) PCB footprint for BTL output



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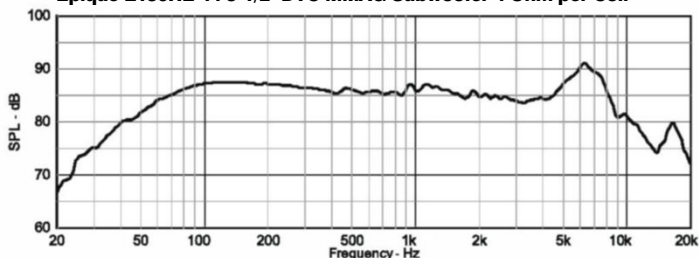
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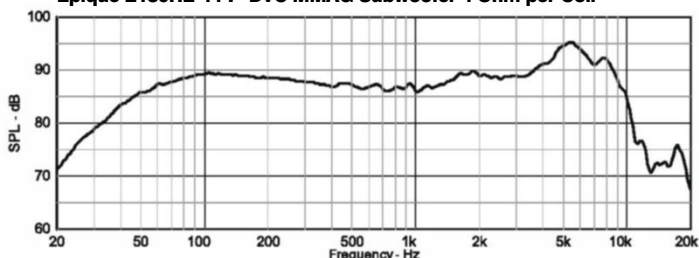
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