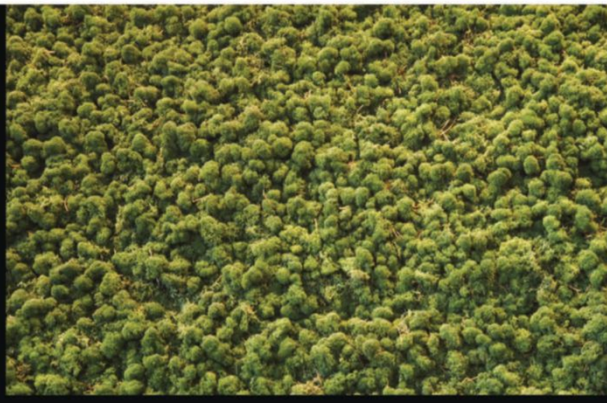
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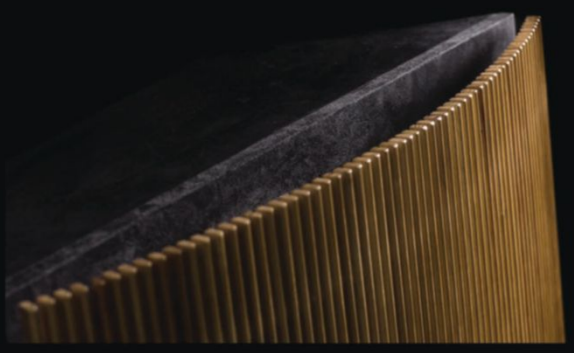
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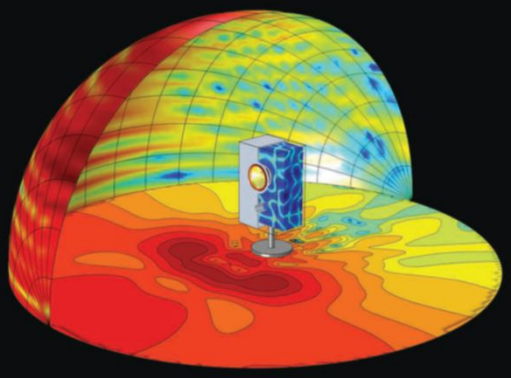
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Science and Technology in Acoustics

At almost every audio industry show we attend, from hi-fi shows to professional audio and broadcast, we regularly meet with established vendors of acoustical solutions and services. However, we are also surprised to find new companies and many new products that are increasingly more sophisticated, aesthetically pleasing, and show a solid approach to the needs of a diverse scope of clients. Consistently, in all our conversations, we discuss a range of topics, which usually tend to focus on the common challenges of acoustics, and the common concerns with low-frequency absorption. Those companies also shared the fact that their investments in new research and the use of computer simulation and visualization tools is becoming mandatory, even sometimes for the most modest and standardized projects. Also, in past years, we have witnessed a common approach from all the acoustic panel vendors, to offer first-approach room treatment kits that result in a very effective solution to solving the basic acoustic challenges.

At those more standardized levels, where common challenges are better understood, acoustic companies are now showing a tendency to approach projects more from the aesthetics perspective, offering carefully designed solutions—including this year, the trend toward "ethnic" patterns for different markets, and integration with architectural lighting solutions.

While there's an increasing awareness of the really difficult challenges, and the need for a scientific approach, all companies will tell their clients that, unless a project is planned from the construction stages, solving the really difficult acoustical problems become exponentially more expensive, while the efficiency of the solutions themselves becomes progressively limited—the law of diminishing returns.

Some companies are trying to design products that address as many challenges as possible (without causing new ones), trying to be more effective in areas that the entire industry recognizes are very difficult to improve. As detailed in a dedicated article in this edition, the ongoing research by Artnovion leads the way in gaining a few precious decibels of absorption below 100 Hz, and that's no small accomplishment.

On the other hand, computerized modeling tools are getting much more accurate at describing and defining acoustical interactions, enabling users to predict behaviors of materials and installations, which would be very expensive to measure in actual chambers. That's precisely what we describe in a piece from COMSOL, detailing the latest functionalities and capabilities of its Acoustics Module, enabling acoustic engineers to analyze the full range of acoustic frequencies from the lowest bass notes to ultrasound.

On the opposite end of the acoustics challenges, there are ongoing efforts to have variable acoustics in any room—or auditorium. Recently, during the 144th Audio Engineering Society (AES) convention in Milan, Italy, I met with E-Coustic Systems USA (LARES Associates, Inc.). They are early pioneers in this field, developing the original Lexicon Acoustic Reinforcement and Enhancement System (LARES) almost 30 years ago. Apparently, E-Coustic Systems is now leveraging the dramatically improved processing power of current platforms to explore new dimensions in active acoustics technology. This is extremely exciting and an area that I believe will soon become common practice in combination with "passive" acoustic treatments.

Our cover for this edition highlights a case study that perfectly summarizes how acoustics needs to be an integrated approach, carefully married with many other disciplines, including electro-acoustic systems. When the business and personal audio solutions specialists at Plantronics decided to create new reference corporate offices in the US and Europe that's precisely what they discovered. After thorough research, they successfully designed an integrated approach to ensure natural harmony within working spaces, leading to the creation of a new business division that currently promotes the integrated Habitat Soundscaping system. I am certain our readers will be fascinated—as I was during my visit to one of Plantronics' campus, in Hoofddorp, Netherlands. I was very impressed with this integrated approach to Smarter Working and Biophilic Design and the completely fresh perspective it conveys.

J. Martins
Editor-in-Chief

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COLUMNISTS

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

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

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

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

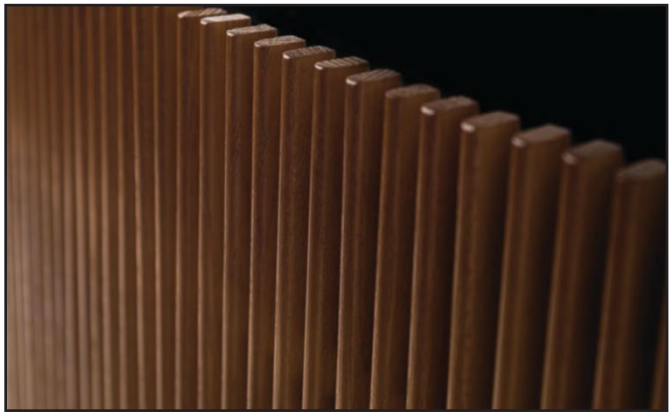
Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at White Sands Missile Range. In 1957, he started Testronic Development Laboratory, which became TDL Technology, to develop audio electronics. All product sales and services were terminated on December 31, 2015, but the TDL website is still online with a variety of audio information and downloads.

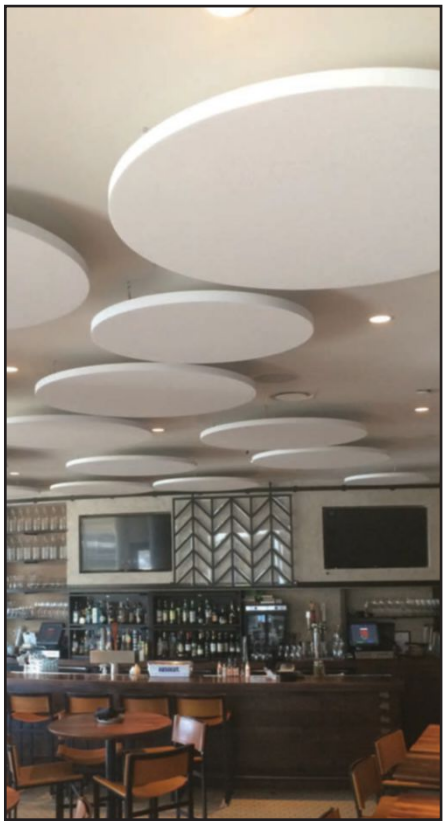


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Noise distraction is the number one complaint for open plan office workers according to a recent study from Oxford Economics. After experiencing its own challenges with noise in an open office, Plantronics tested and developed its own solution to noise and distraction using biophilic acoustics or acoustics based on elements from nature. The article details how Plantronics arrived at its Habitat Soundscaping solution.

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Show Report

High End 2018

Higher-End Audio Electronics in Munich



Yes, the High End in Munich, Germany, is my favorite show for everything high end audio. There, we indeed find everything. As all the manufacturers are there and all the trends are on display, we always find ourselves facing the question: Where to start? That is great fun, as well as a challenge. The following is a report of our main findings for the year.



By
Ward Maas
(The Netherlands)

For the 2018 High End Show, the weather outside was warm and summer-like, and the mood inside the show was also more relaxed. In previous years, I used the tactic of walking around the Munich Order Center (MOC) on the first day to try to get a feel for the layout and discuss with colleagues what was really interesting. The next two days I would spend following up on all the leads. I would use the last day to check if I missed something important.

However, this year was different. It seemed everyone had something really special to showcase. Or better, everyone had multiple new products and was introducing multiple fascinating technologies. So, the 2018 show turned into a four-day marathon that concluded with an extreme thunderstorm, which washed away the sweat of a once again very exciting High End.

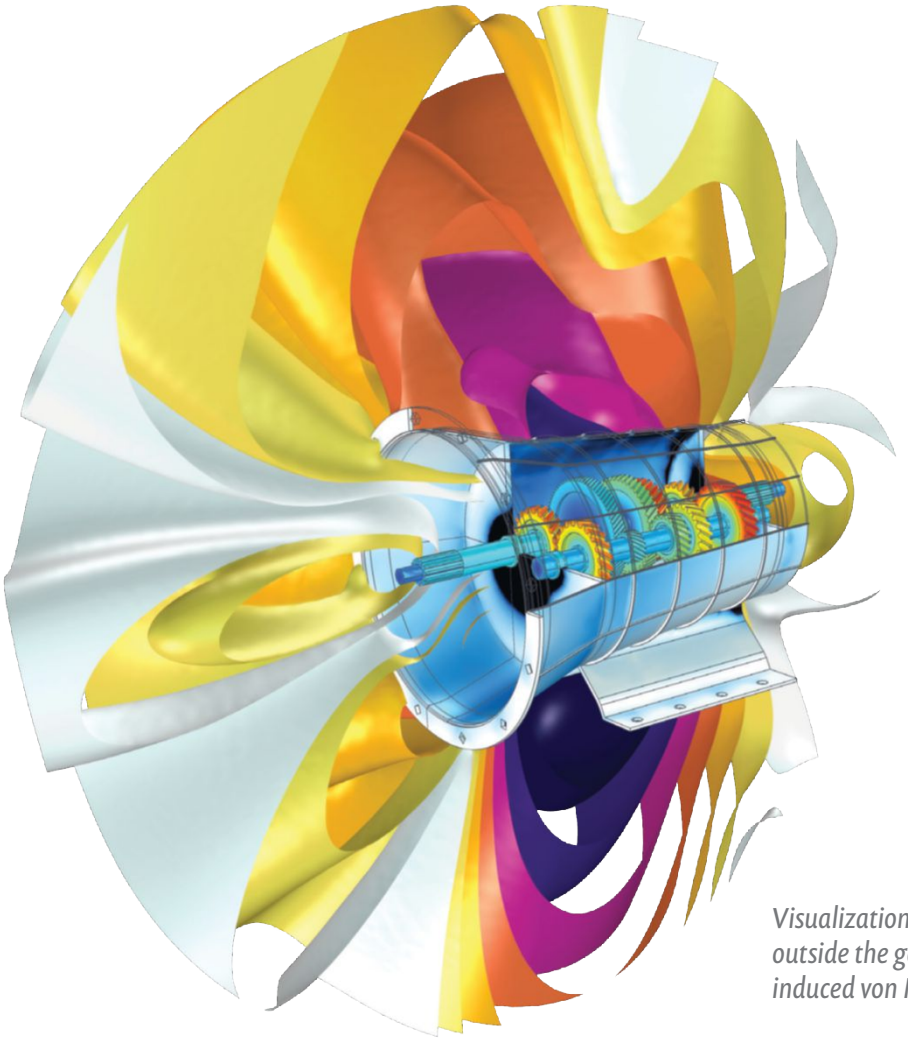
Although there was officially a slight decline in visitor numbers compared to the previous year, it was not noticeable on the first (dealer and press) day. After that first day, you could maneuver better and we could actually discuss products in-depth. It seems that German consumer visitors had to choose between spending those summer-like days with family and visiting the High End. This certainly prompted many potential visitors from the region to stay away from visiting the show.

Better Loudspeakers

High-quality loudspeakers were certainly a hallmark for the show this year, and in this article, we will only focus on the new products.

PMC Audio (https://pmc-speakers.com) introduced the "fact fenestria" (fact being a completely new consumer series from the British company, renowned in the professional studio monitor world). It is a large loudspeaker completely designed from scratch. The fenestria design took five years of development. Special are the Laminair ports placed on the end of twin Advanced Transmission

Simulated NVH testing you can see and hear.



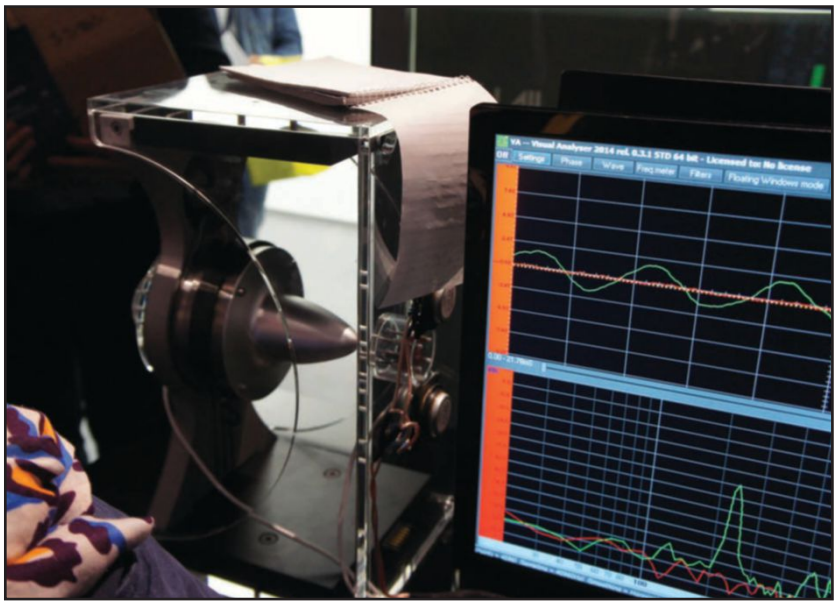
Visualization of the noise pressure level outside the gearbox and vibration-induced von Mises stress in its housing.

The most effective approach for reducing the noise radiation from a gearbox is to perform a vibroacoustic analysis to learn how the design can be improved. Noise, vibration, and harshness (NVH) testing is an important part of the design process, and it can be simulated with multiphysics software.

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comsol.blog/NVH-simulation

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To highlight the vibration-free qualities of the fact fenestria three-way speakers (seen here in detailed illustrations), PMC had a laser vibrometer in the demo of this completely new high-end speaker series.



Now even more powerful, the Kii THREE gained a powerful bass extension upgrade.

Lines (ATL) as the brand calls its proprietary bass-loading technology. These Laminair ports are, just as with F1 cars, intended to make turbulence free airflow possible. The four bass drivers would normally cause interference with the mid- and high-frequency loudspeakers. Therefore, the complete mid/high section, named "the nest," has been acoustically isolated from the remaining structure. The four bass drivers can produce so much acoustic energy that PMC designed a pendulum system, similar to those used for earthquake protection on skyscrapers, to completely eliminate unwanted vibrations. During the demos, the "fact fenestria" showed a powerful and controlled impulse behavior.

Kii Audio (www.kiiaudio.com) had a display at the High End and another display at the Hifi Deluxe showcase, in the nearby Marriott Hotel. The Kii THREE already has a solid reputation as an extreme transparent system with a precise bass reproduction. The introduction of the BXT modules with eight bass loudspeakers per side made clear that there is still something to gain. The demos once again showed the absolute ease with which these loudspeakers can reproduce high-quality music.

California-founded and China-based ESD Acoustics (www.esdacoustics.com) was remarkably active, especially for a company exhibiting at the High End for the first time. Not only did ESD Acoustics introduce the massive Dragon five-way active system with Carbon Fiber Horns, it also did not shy away from the use of Field Coil Loudspeakers (no permanent magnets but electromagnetic ones with a power supply). Furthermore, during a lecture, they introduced a new loudspeaker cone material, new amplifiers, and a new DAC. The amplifier and DAC cabinets looked like perfectly lacquered Chinese boxes. All of that was on display in a rather cramped demo room, making photography useless. All in all, their incredible enthusiasm was only dampened by a lack of European experience. It would not surprise me if this is an indication that more original Chinese contributions to high-end audio can be expected.

And talking about creating an impression, saying the presence of Ascendo Immersive Audio (www.aia-cinema.com) was remarkable is an understatement. How often do you encounter a 50" driver at a show? Albeit only a display, the giant woofer was a crowd-grabber (and selfie inducer…) and an effective way to convince visitors to enter and spend time in their large demo room. The German company decided not to set up a fully working installation, as it did last year (creating complaints from

neighboring demo rooms), and instead opted to show some of their cinema-oriented loudspeakers in refrained "hi-fi mode." Still, looking around their demo room, one could see that bass and infra bass is still their thing. Their amplifier products, which are not that impressive aesthetically, are fully Audio Video Bridging/Time Sensitive Networking (AVB/TSN) networkable and all the systems are DSP controlled. As immersive is, at this moment, the buzzword in the pro audio world, it is not difficult to understand why Ascendo wants to be recognized as a reference in immersive theatre systems. Wow!

Lyravox (https://lyravox.com) is, of course, known for its all-in-one ultra-luxurious soundbars. In Munich, however, the emphasis was on the "normal" Karl, Karlotta, Karlina, and Karlos loudspeakers, with a highly innovative technological approach. Ceramic drivers combined with Pascal bulletproof amplifiers capable of playing a world of digital audio formats. In short, examples of 21st century loudspeaker design. Interesting and impressive. The Karlina and Karlos are the faces of the new "affordable" products.

Amplifiers and Electronics

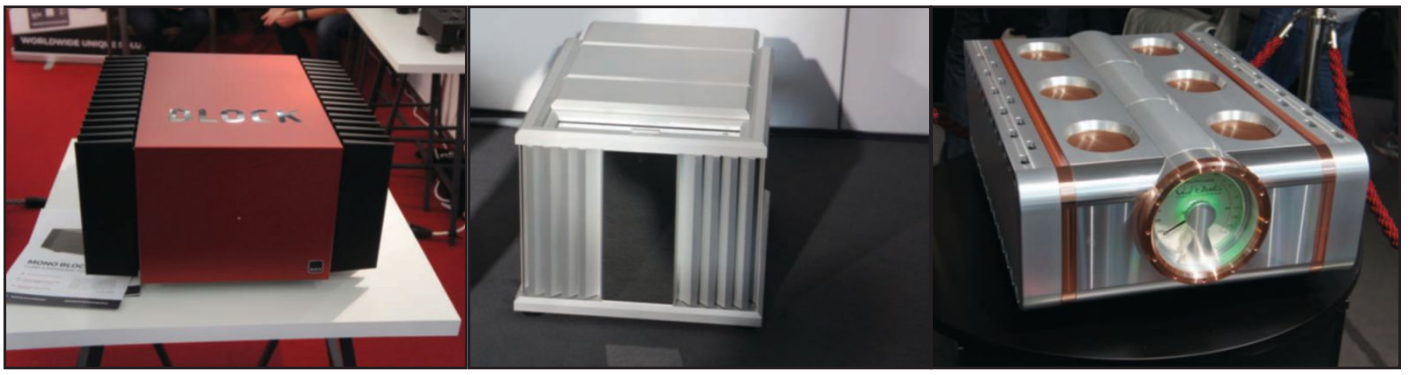
No High End is complete without the monster amplifiers. Block Audio (www.blockaudio.com) showed a very nicely styled dark red amplifier named Mono Block... An amplifier with no less than 200 W in Class A. It is not for energy saving audio enthusiasts as the idle power is 500 W. If that heats you up, there is an eco-mode of 50 W. Further specifications are very nice as well. Specifications for the new Burmester 159 (www.burmester.de) are so far unkown. However, it is the predecessor of the famous 909 MK5, the first real new Burmester amplifier in a long, long time. A weight is mentioned of 175 kg... for a mono amplifier! Although the new D'Agostino Relentless (http://dandagostino.com) is topping that with a weight of 220 kg. However, it brings you a staggering 6000 W at 2 Ω.



While many were attracted by the giant woofer on the back, Ascendo had its Live 15 loudspeaker front and center in the room, a complete hi-fi statement from a home theater specialist.



The Lyravox room and the company's new extended range of luxury loudspeakers are on display.



The Block Audio Mono Block, the Burmester 159, and the real monster—the Dan D'Agostino Audio Systems Relentless Monoblock.

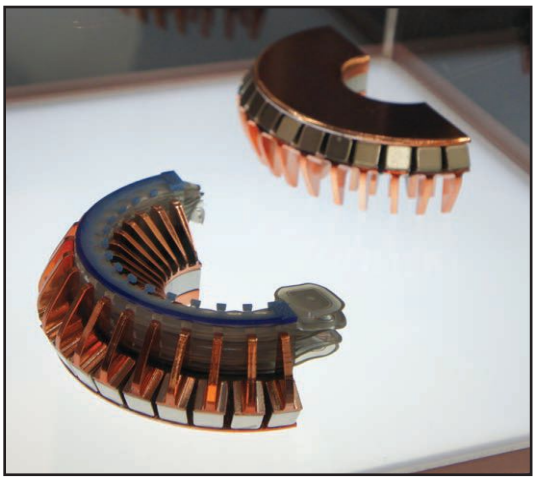


The Mola-Mola Tambaqui DAC

Looking at the more down-to-earth technologies, Hypex (www.hypex.nl) showed the complete Fusion plate Amp series for the very first time to a larger audience. There are no less than eight different models ranging from the FA 251 of 250 W in 4 Ω unit to the FA503 with two 500 W units and one 100 W unit. They are, of course, equipped with all the features one would expect from a plate amp.

Also, on display at the Hypex stand were the Mola Mola products (www.mola-mola.nl). Very interesting was the introduction of the Mola Mola Tambaqui DAC. Basically, it is the same DAC that was used as an option available for the MAKUA preamplifier. It features a PWM DAC output stage with 32-stage discrete analog FIR, asynchronous upsampling to 3.125 MHz/32 bit, seventh-order noise shaper clearing 80 kHz band, and all types of inputs, including AES/EBU, USB, Ethernet (Roon, DLNA), Bluetooth (A2DP, aptX), and I²S over HDMI. Outputs are balanced and supported formats include PCM up to 384 kHz/32 bits, DoP up to double speed and Native DSD up to quad speed.

Astell & Kern (www.astellnkern.com) is an example of a typical Asian manufacturer, with a very large product range in Portable Audio, Home Audio, and accessories—all very high quality—and expanding every year. So much so, it's hard to keep track. In Munich, their large product collection was on display and could be heard as well. New and of notice was the A&futura SE100, equipped with SABRE ES9038Pro DAC, which can handle 32 bit/384 kHz PCM audio and Quad DSD. With an Octa-core CPU, 5" touch screen, 128 GB internal storage, support for fast charging and a battery life of 10 hours, this is a true high-end portable system. Also new, was the "standard" class A&norma SR15, which uses a Cirrus Logic CS43198 and a Quad-core CPU. The A&norma SR15 can handle 24 bit/192 kHz PCM audio and single-rated DSD.



Arya Audio Labs took the original Christensen Audio transducer concept and decided to make it commercially available as the AirBlade. The shape of things to come.

LINDEMANN (www.lindemann-audio.de/25-jahre-lindemann) did not let its 25th anniversary go by unnoticed. With the slogan "The best of 25 years," LINDEMANN drew attention to its Limetree series of products. These small products fit the trend of putting limited essential wished-for functionality in a smaller cabinet instead of providing the bulkier "do it all" products. In the Limetree series, there is a Phono preamp, a headphone amplifier, a USB-DAC, a network player, and a network bridge available. Everything at affordable prices.

An example of a company that has been attending the High End show for years is The Bespoke Audio Co. (www.thebespokeaudiocompany.com). Really a very understated booth, it is easy to pass it unnoticed. What you might notice, however, is the transformer winding machine. It is not just there as a prop, but it is actually used. They use it to produce the heart of a rather exceptional preamp. A preamp with a transformer volume control. The lady operating the machine explains what she does, why she does it, and why that creates an absolutely top class product. Yes, you can have an in-depth coil winding discussion with her during the show. Brilliant!

And speaking of electronics, as the saying goes, "when there is a knot in the garden hose, little water will emerge out of it." The German company EFUSE (www.efuse.de) realized that the same applies for a super mains cable where the apparatus electric fuse (a very thin melting wire) forms the knot. Trying to solve that problem, EFUSE offers a resettable electronic fuse, which nicely cuts the power above a certain current, while assuring that below that current it doesn't behave as a "knot" for electric current.

Also, despite the fact that CE regulations can be tough for small companies, Dutch company Sbooster (www.sbooster.com) showed that it does not shy away from such inconveniences by constantly improving its production and its products. A very interesting option is the replacement of the switch-mode power supply of Lumïn products by



Something worth looking at: the Reed 5T "tangentially tracking pivoted tonearm."

an Sbooster connection kit. An upgrade that can be done in 15 minutes.

Promising Newcomers

Under the Newcomers' banner, which is a specific showcase promoted by the High End Society, start-up companies can present themselves under friendly conditions. This year, there was one company that clearly stood out from the rest. Arya Audio Labs (www.arya-audio.com) from the UK, resumed the work originally patented by Christensen Audio, and presented the Airblade transducer. A tweeter constructed with two half rings of magnets on top of each other enclosing a diaphragm stack of foil. By nature of the construction, the dispersion can more or less be determined. Arya did build prototypes with 90°, 180°, and even 360° dispersion. We will definitely hear more from them!

Not exactly a start-up, MSB Technology (www.msbtechnology.com), also made a strong statement in Munich—the Reference DAC, the Reference Transport, and the M204 Monoblock Amplifiers, to mention the best of the best. The future-proof modular hardware approach, the perfect finish. It is also clear that if you have to ask the price... you are probably not in the company's potential customers group.

Turntables and Tubes

Walking around a show like this you will be bombarded with all kinds of visual stimuli. It happens all the time as you quickly pass the umpteenth product of which you have seen variants all day long. In this case, I passed the REED booth (www.reed.lt) and thought I saw a tone arm making a rather unusual movement. Indeed, I was right. The REED 5T is, as they say, a "tangentially tracking pivoted tone arm." With a 1-second groove test disc (where the arm moves in 1 second from outside to inside and back again) this looks weird. However, it is well-thought-out and looks promising. Good that I paused there!

Another turntable worth mentioning is the Technics SL-1200G (www.technics.com). Last in a line of the famous 1200 series. Looking under the hood, you can see the amount of time and technology that went into it to make it a top product. A direct drive motor, motor control, plateau, a tone arm, housing, and isolation. I have to admit that you have to stand close to it because it doesn't look that spectacular. Good to see this icon back. It is a special turntable.

The story goes that Pro-Ject CEO Heinz Lichtenegger once stopped at a gas station and went into the accompanying shop to discover—to his horror—a plastic turntable with a built-in loudspeaker. That was the moment he decided to go into the turntable business to provide decent products at decent prices. His mission was to provide quality musical experiences. Not only for the High End market (which he serves very well) but also for people taking their first

About the Author

Ward Maas is the owner of Pilgham Audio. He studied electronics, marketing, and amplifier design. During his career in consumer electronics, Ward worked in areas ranging from CD standardization to radio and television to personal GPS navigation. Ward has worked on an extreme low-noise magnetic cartridge preamplifier and several special amplifier products. As the CTO of "Witchworld," a theme park near Amsterdam, he also works with animatronics. He lives in Almere, Netherlands, with his wife and son.







It was a pleasure to meet a reenergized Western Electric at the show and see the return of the 300B vacuum tube. Pictured are the *audioXpress* expert team—Norman Weber, Ward Maas, and Gerhard Haas (Haas and Weber have worked on many tube projects, including the original Sennheiser Orpheus project), together with Charles Whitener, president of Western Electric.

steps into the world of LP analog music.

Nowadays, Pro-Ject (http://project-audio.com) is the world largest turntable manufacturer. During the High End 2018 show, Pro-Ject showed the "Jukebox S2" products. A step in a new direction for audiophile turntables, which come completely factory adjusted with built-in amplifier and Bluetooth connectivity. Plug and play.... Now, we won't find these products sold in a gas station, but for those who want to step in the vinyl world, this is a decent product that won't disappoint.

For tube aficionados, the Western Electric brand (www.westernelectric.com) is legendary. The 300B is definitively a highly praised cult tube. It was good news to hear that Charles Whitener is keeping the company alive. Particularly good to hear about the company's expansion plans, with a new and re-imagined electron tube works facility in Rossville, GA, where a new production run of the 300B will be done. According to Whitener, some of the production equipment from an old Philips picture tube factory will be reused. Very interesting also is the fact that there are plans in an advanced stage to start the production of a tube, which was under development when the original Western Electric company closed. This tube seems to be a 300B on steroids capable of driving amplifiers with much more power. More exciting news is expected this year.

Meanwhile in Europe, Thomas Mayer, known from Vinylsavor (http://vinylsavor.blogspot.com/2017/08/deutsche-elektronenrohren-manufaktur.html), is working hard to develop the product program for the Deutsche Elektronenröhren Manufaktur GmbH, which took over the ELROG tube factory in August 2016. This company had its roots in the well-known Telefunken tube activities. We hope to discuss a little more about the activities of this factory in the near future.

Finally, I need to mention Kostas Metaxas, a designer with a reputation for unusual audio designs. Unusual, even strange, but of high quality. *audioXpress* readers will be familiar with the design of one of his first preamps. What Metaxas & Sins presented in Munich was unusual, to say the least. A design that is fascinating, attractive or perhaps a little scary. The photos speak for themselves. Nowadays, Metaxas & Sins headquarters are in Amsterdam. Visit www.metaxas.com for more stunning images and information regarding these unique audio products. You can learn more about the "conventional" products on the legacy pages (www.metaxas.com/pages/masnewfiles).



A different take on the reel-to-reel revival by Kostas Metaxas

CANJAM-Europe

Very close to the MOC building, where the Munich High End show is held, this year there was something else to visit— CANJAM-Europe. Housed in the former "Kohlenbunker," which is on an old factory building where they used to build train coaches, CANJAM-Europe is a small headphone show where a number of (very) high-quality headphones and matching headphone amps could be tried, in a very pleasant environment. Most were established names, while others were completely new to us. Modenaudio (www.modenaudio.it) from Modena, Italy, presented a rather understated headphone/line amp, which was basically able to drive all electrodynamic headphones. Due to some clever switching possibilities, it can be used with a larger

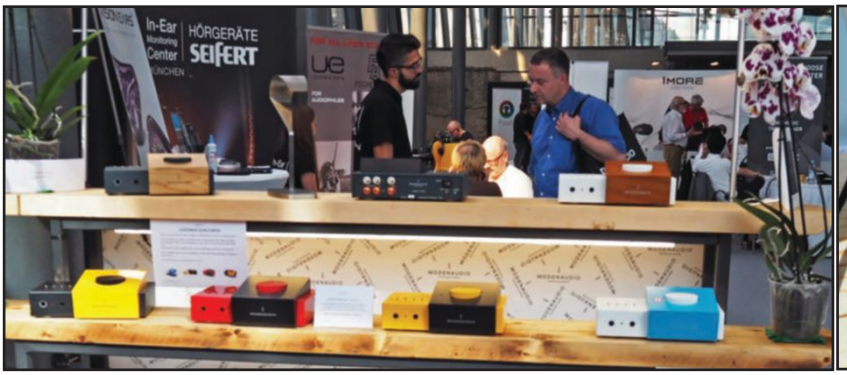
system or standalone. Available in gray or white with a choice of five sorts of wood. Okay, if you're from Modena, then matching red or yellow like the cars, which are made there is also an option.

The Future of Digital

At these shows there is still a lot of "old technology" wanting to prove it is the only way to High-End Nirvana and often also promotes that image of a "service technician included" product, which says something as well. In contrast, are



A partial view of the nice environment at the CANJAM Europe exhibits





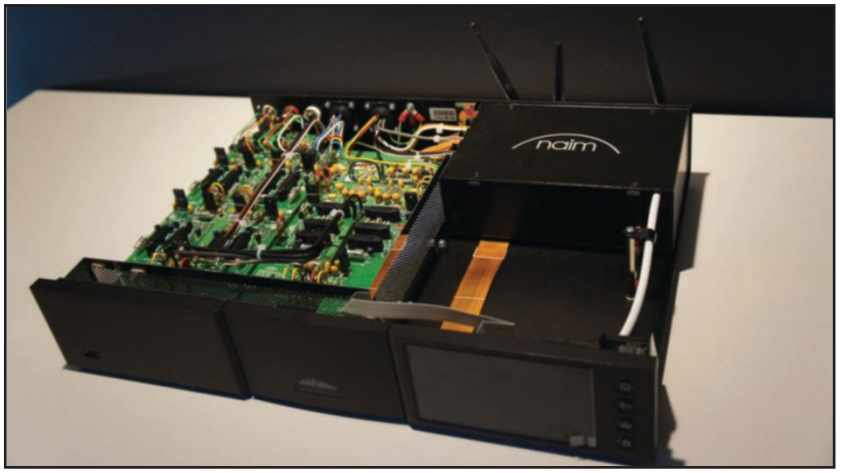
The Lympha headphone amplifier from Modenaudio is a versatile unit and comes in multiple colors and natural wood finishes







Merging Technologies MERGING+NADAC network with an attached DAC and MERGING+PLAYER with an integrated Roon player makes for an extremely elegant high-end system, in this case matched in black finish with the new (and still unnamed) PSI Audio speakers. Swiss perfection.



The reference NAIM ND555 network and streaming player combines state-of-the-art DSP with a rich, analog soul.



These tiny things are the Pre Box S2 Digital DAC, using ESS Sabre converters and supporting up to 32-bit/768 kHz PCM and DSD512, as well as hardware MQA unfolding ($349); and the Stream Box S2 Ultra ($799) a Roon-ready Wi-Fi or Ethernet music streamer.

the companies that demonstrate all-new digital technology with pride and know-how. Merging Technologies (http://nadac.merging.com) proved once more why it can claim absolute top quality. Merging's products, such as the MERGING+PLAYER and the MERGING+NADAC, are there to prove that analog is not the modern concept it used to be.... Connected simply with a Ravenna network and feeding active speakers from PSI Audio, the statement was totally clear.

Grimm Audio (www.grimmaudio.com) was there as well, promoting the Dutch company's flagship LS1be loudspeaker system in a carefully designed and decorated environment. Showing future directions, its new MU1 "streamer" could not be overlooked.

This MU1 is supposed to get a very central role, hence an extended functionality, combined with an unmatched ease of use. It is a complete digital audio platform with room to grow, the ultimate digital source, with support for Tidal, Qobuz, and Primephonic music services, as well as Spotify Connect. Combining top-quality conversion, the MU1 can also be used as a source for third-party DACs. All digital input formats are supported, including AES, as well as internal SSD, external USB and NAS storage options. And there's even an FM and DAB receiver inside, and room for a surround playback option. The MU1 will be available beginning November 2018. Also planned are the MU2 and the MU3, where that central stage role will be expanded once more. By the way, have a look at the Grimm Audio banners at shows. They look awesome!

Pro-Ject (http://project-audio.com) also had a number of non-turntable products on display as well. The large range of products branded "Box Design" are becoming more popular in this digital age. Particularly the Box Design S2 series, created with the intention to make a product as small as possible, but still having a breathtaking sound performance.

Last year's EISA award-winning PreBox S2 Digital now has the ideal companion, the Stream Box S2 a 24-bit/192 kHz streamer and Internet radio. John Westlake, Pro-Ject's digital product designer, did his very best. This little (the size of three CD jewel cases stacked) box is packed with features, options, and possibilities and together with PreBox S2 Digital, they form a formidable digital side of an audio system in itself and not only in price/performance. For the very tech-minded, have a look at specs, the back of the product, and the Internet discussion groups and you will find out that there is a lot more than meets the eye. A cult product is born!



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The Challenges of Live Recordings



Making a high-quality audio recording in a recording studio is one thing, but making a similar recording at a live concert has its own challenges. Neville Roberts flies out to Venice, Italy, to investigate.

By
Neville Roberts
(United Kingdom)

I've been fortunate to be able to attend a number of recording sessions in recent years. The aim at each session has, of course, been to produce audiophile-quality recordings either in high-resolution digital format or in the best that analog has to offer in terms of professional master tapes on quarter-inch, half-track, 15 IPS tapes, and direct-to-disk LPs. Most of these recording sessions have been held at recording studios, where everything to do with the recording, from the acoustic environment through to the recording systems and microphones mixing, can be controlled to produce the final stereo mix. Of course, direct-to-disk recordings can only be undertaken at a studio that has the lathe and the supporting equipment required to cut the master lacquers.

Making a recording at a live venue with a live audience is a completely different process altogether. So, when I was offered the opportunity to attend such a live recording event, arranged by the audiophile record label Chasing The Dragon, I was very keen to attend. Then, when I found out that the event would involve recording a series of live concerts showcasing one of my favorite genres of music, Italian baroque, my excitement increased. To top it all, the concerts were to be held in the world renowned and beautiful San Vidal Church in Venice, Italy, and all I had to do was to get myself there in mid-April 2018 to witness the recording sessions first-hand.

The Performers

The recordings featured the internationally famous Interpreti Veneziani, which debuted on the Italian and international concert stage in 1987. The orchestra quickly gained a reputation for the exuberance and all-Italian brio of its performances. Interpreti Veneziani (www.interpretiveneziani.com/en/index.php) performs every year to more than 60,000 people from all around the world and is the leading chamber orchestra in Venice where they are presently enjoying their 30th concert season. Their important achievements include appearances in the Melbourne, Bayreuth, and the Prague Music Festivals, as well as concerts at Stockholm's Royal Palace, participating in the World Vision Telemarathon at the Kirov Theatre to mark the reinstatement of the name "St. Petersburg," a concert at the Osaka Symphony Hall in a live broadcast for the Japanese Radio, and concerts at the Tokyo Suntory Hall and Kjoi Hall. The orchestra also travels to the United

States, Canada, and Latin America to hold concerts in the most prestigious halls. In 2013, they performed for the first time in India. However, Interpreti Veneziani's home is the magnificent San Vidal Church in Venice.

Interpreti Veneziani recorded their first CD with music publisher Musikstrasse featuring music by Giuseppe Tartini. They also have 19 CDs with InVeNiceSound. In 2014, the orchestra traveled to AIR Studios in London to produce a direct-to-disk LP of Antonio Vivaldi's *The Four Seasons* for Chasing The Dragon.

Uniquely, Interpreti Veneziani doesn't have a conductor or orchestra leader as such. Instead, the musicians all work together as a team—and having heard them firsthand, I can see how well that arrangement works. The members of the orchestra are all soloists in their own right and take turns performing when a solo instrument is needed. For example, with the performance of Vivaldi's *The Four Seasons*, Giovanni Agazzi was the solo violinist for "Spring" and "Summer," and Giuliano Fontanella was the solo violinist for "Autumn" and "Winter." The business side of the orchestra, including the organization of the orchestra's concerts worldwide, is the task of the harpsichordist, Paolo Cognolato.

The orchestra not only performs like a close-knit family, it also has members who have family connections. For example, the harpsichordist Paolo is married to the viola player Sonia Amadio, who also happens to be the sister of the cellist Davide Amadio! The energy and enthusiasm given to each performance during the week was both contagious and captivating. Even the rehearsals, which took place from 6 PM to 8 PM every day, were fresh and exciting.

The Location

The San Vidal is located at one end of the Campo Santo Stefano in the Sestiere of San Marco, Venice. The original church was built at the site in 1084 by Doge Vitale Falier. This church was destroyed in 1105 by a major city fire. Although reconstruction was finished by the 16th century, poor foundations necessitated a further rebuild in 1696 to honor the former Doge Francesco Morosini. The main altar piece features a beautiful painting by Carpaccio of San Vidal on horseback. The church has now been fitted with acoustic carpeting around several of the pillars in order to control reverberations. The result, in my humble opinion, is a venue with absolutely superb acoustics.

The Equipment

Producing audiophile recordings at a remote venue is a massive undertaking. First, there is the logistics of getting all the recording equipment to the venue, and



Aerial view of the San Vidal Church interior in Venice where the live concerts took place.





Mike Valentine starts to set up the microphones (a) and close-up of main microphone assembly (b).



The (AKG) Flea C12 (a), The DPA 4006 (b), and the Neumann U47 FET (c) microphones used in the recordings.

the bulk of that had already arrived by the time we appeared on the scene. A spaced pair of modern Flea C12 microphones set in omnidirectional mode were used for the main stereo feed. These microphones use a 6072 triode fed from a CK12 capsule. Originally designed by AKG and in production between 1953 and 1963, these were the first microphones in the world to allow remote control of the nine polar diagrams (omnidirectional, figure eight, cardioid, and various proportions of these three) through a nine-position switch. This allows for subtle changes to be made without having to dismantle the microphone setup.

A sheepskin-covered Jecklin disk, which enhances the stereo image when used with omnidirectional microphones, was placed between the C12s set in omnidirectional mode. The Neumann KU-100 dummy head was positioned just below the C12s to produce binaural recordings, which sound particularly good when listening on headphones.



The Flea microphone power supplies were used with Nordost power cables.

When concerts are recorded live, there is no opportunity to repeat a performance, so it is important to make sure that all the instruments are recorded correctly. To this end, a couple of additional microphones were set up to capture solo instruments that might require an additional level of detail. The output from each of these microphones was recorded on a separate track, which could be added to the stereo feed later if required.

The Neumann U47 FET is a modern microphone that is widely recognized as a superb microphone for female vocals, cellos, double basses, and kick drums, so it was used as a spot mic on the cello. The cello featured significantly in the various concerts, and Interpreti Veneziani is fortunate to have the internationally acclaimed cellist Davide Amadio as one of its members. There was a concern that the spaced pair of C12s might not pick up all the subtle tones of the harpsichord, so a DPA 4006 (formerly known as a B&K 4006 and used in the BBC Proms transmissions) was initially positioned near the harpsichord. However, during rehearsals, there were no issues with the C12s' ability to pick up the subtleties of the harpsichord. The DPA was then positioned close to the violins so that, if necessary, the balance of the strings could be adjusted at the post-production mixing at AIR Studios in London the following week. Chasing The Dragon Recording Engineer Mike Valentine commented that microphones themselves are like musical instruments, so it is

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In the case is $40,000 worth of cables (a). The cables, along with the Focusrite preamps and two Sennheiser HD-800S headphones (b), the Nagra VI Tascam, and the Studer A810 recorder (c), were used for the live recordings.


a)


b)


c)



Rehearsals with the bassoon

vitally important to choose the right microphone for a particular application.

The feeds from the microphones were connected to a pair of quad Focusrite 1 microphones preamps. Each of these eight preamps (although only six were used) provided three outputs—two balanced transformer splitter outputs and one straight output, which is a parallel feed with the input. This enabled the mics to feed all of the recording devices.

PCM digital recording at 24 bit, 192 kHz was undertaken by a Nagra VI six-channel recorder. Channel 1 and Channel 2 were for the stereo feed from the C12s, Channel 3 and Channel 4 for the KU-100 dummy head, Channel 5 for the U47, and Channel 6 for the DPA mic. For DSD recordings, two Tascam DSD two-channel recorders were used—one for the stereo feed from the C12s and one for the binaural feed from the dummy head.

For all the analogaholics amongst us, the stereo feed from the C12s was fed to a Studer A810 recorder, running at 15 IPS and recording onto ATR master tapes. For many people, using an analog format for mastering seems to be a bit of a backward step that is driven more by nostalgia than sound quality. However, many converts, including myself, can hear significant differences between music played from an analog or digital source.

The analog sound is simply more real, more natural, and less clinical. Within the world of digital audio, there are noticeable differences in the degree of realism produced by high-resolution PCM digital, DSD, MQA, and other digital formats. But what the analog source is able to reproduce is something that seems to be missing from all digital recordings—something that is key to producing the essence, or at least the illusion, of reality.

With all this high-end equipment to record high-end performers in the superb setting of San Vidal church, it would be a shame to let the audio quality be compromised by the use of inferior cables. To connect and power all the equipment, around $40,000 (£30,000) of cables were used. These included Nordost Ax Angel power cables and Heimdall 2 headphone cables, together with ZenSati Zorro balanced interconnect cables. Sennheiser's top-of-the-range HD 800S dynamic headphones were used to monitor the performances.

The Rehearsals

Each morning, the digital files from the previous day were downloaded from the Nagra VI and the two Tascam digital recorders to a hard disk to make space for the day's new recordings. Daily rehearsal sessions were absolutely invaluable. Apart from the obvious benefits of being able to set up the

equipment, adjust the levels, and ensure everything was working correctly, rehearsals allowed any unforeseen issues to be corrected before the actual recordings. For example, on one occasion the weighty U47 microphone started to slide down on the stand during Davide's rehearsal of Vivaldi's "La Follia." Two people immediately tightened the microphone clamp, and the problem never reoccurred. Other issues that had to be addressed related to health and safety, with plenty of Gaffer tape being used to secure cables that ran rather precariously over the walkways.

Rehearsal sessions also gave the musicians the opportunity to hear some of the previous night's recordings. It was evident that the musicians were a little nervous, being very aware that a bad note or inclination, which would pass largely unnoticed at a live concert, would be recorded for posterity. This heightened sense of urgency and tension didn't seem to have any negative effect on the performances. Of course, hearing each morning how good their performance had been the previous night—and believe me, they really were note-perfect—was a real confidence booster for the musicians and helped to allay any fears.

The Recordings

All the concerts began at 9 PM and every night the church was packed and the atmosphere electric. I sat to the right of the orchestra with the panel of equipment in front of me and monitored the performances on my HD 800S headphones. Before the concert started and following the usual request to put all phones on silent, an announcement was made informing the audience that the concert would be recorded and requesting people to refrain from coughing, if at all possible, which probably made things worse. As a precaution, bottles of water were on hand to give to people as needed. Fortunately, the forecasted thunderstorms did not materialize and the audiences were very well behaved.

It took four pairs of hands to set all the equipment to record as Interpreti Veneziani took to the stage to the sound of tumultuous applause. The orchestra commenced with the first movement of Vivaldi's famous *The Four Seasons* and their performance had all the vivaciousness and sparkling excitement that one would normally associate with a new work that was being performed for the first time. I sat in front of the Focusrite mic preamps and could see the six VU meters behaving themselves admirably, confirming that all the set up during rehearsals had been successful.

The first movement concluded and the audience spontaneously burst into a round of enthusiastic applause. Unfortunately, this was not something that



Mike Valentine sets up the cello microphone.



Mike Valentine monitors and makes final adjustments during rehearsals.

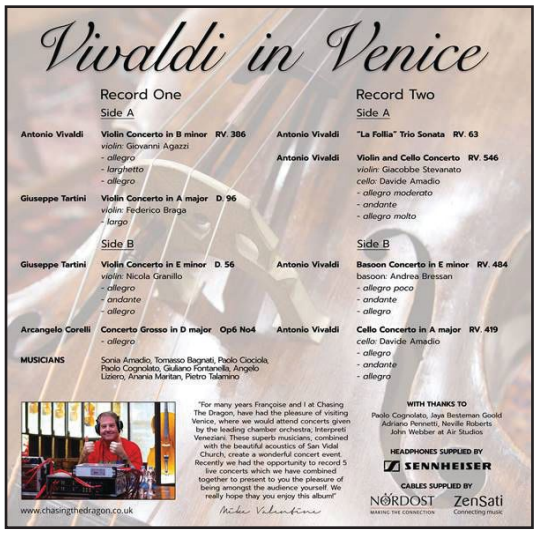


Interpreti Veneziani on the stage

The five recorded concerts from Venice have been released by Chasing the Dragon as the *Vivaldi in Venice* double albums, available as vinyl LP, CD, high-resolution digital downloads, copies of Master Tapes, and even a Binaural download for headphone users.

could be edited out later as doing so would have a detrimental effect on the decay of the final notes of the piece. Aware of this, Davide the cellist thoughtfully raised his hand at the end of the second movement to make sure this was not repeated. The audience remained silent for the rest of "Spring" and only applauded again at the end of "Summer," when the solo violinist swapped with another member of the orchestra who then played the remaining two of the set of four concertos.





It was interesting to notice the subtle differences in acoustics with the audience present, compared to what I heard during rehearsals. The overall sound was clearer and more sophisticated, and the performance somehow had more authority and excitement. Like a good Italian wine, the concerts seemed to improve over time. This was probably because the performers were becoming more accustomed to the presence of the recording equipment and feeling less anxious, knowing that a collection of successful recordings was already building up.

The Aftermath

At the end of each evening's recording session, we were all exhausted but at the same time exhilarated and looking forward to the following day's concert. All the digital files were to be taken to AIR Studios in London the following week for post-production mixing of the digital files (using industry-standard Pro Tools software) and cutting of the master lacquers for the LPs.

It goes without saying that the quality of the recordings is fantastic, but don't just take my word for it. Chasing The Dragon has kindly made available a low-resolution 16 bit/44 kHz digital download of the final Allegro from Corelli Op. 6 No. 4 (an encore played at Tuesday's concert) so you can experience for yourself the ambiance of the church and the quality of the playing. You can download sample recording from my website at rssconsultancy.co.uk/files/ChasingTheDragon/InterpretiVeneziani.wav

The rich variety of the music, primarily by Vivaldi but also featuring another Venetian composer, Giuseppe Tartini, is to be made available in a variety of formats, including double CDs, digital downloads in 24 bit/192 kHz PCM, and single and double DSD of both the stereo and binaural versions. The analog stereo releases will be a double LP and double copy-master tapes. As with the other recordings by Chasing The Dragon, the double LPs, double CDs, and copy master tapes of this recordings can be purchased from the US distributor Elusive Disk at www.elusivedisc.com. In addition, the digital downloads are also available via the website of Chasing The Dragon at www.chasingthedragon.co.uk.

This recording session is the first of a series of recordings that will showcase Venetian music performed by Interpreti Veneziani. If these future recordings turn out to be of the same great quality, which I am sure they will, then music lovers will have access to a growing collection of modern recordings of superbly performed Italian baroque—and in formats to suit all tastes. Recording music at live concerts certainly presents real challenges for the recording engineer, but the end results are well worth it.

About the Author

Neville Roberts, a graduate in Physics, is a Chartered Scientist, Chartered Engineer, Chartered Physicist and a Fellow of the Institution of Engineering and Technology. He is a retired Director of the UK National Health Service (NHS) and has also worked for the UK Ministry of Defence and in the private sector. He is a regular contributor to a UK audio magazine, Hi-Fi Choice, and has a particular interest in tube audio design and is a vinyl and tape devotee. Apart from being an audio enthusiast, he enjoys a wide range of music including classical, especially baroque, light orchestral and jazz. He is a keen photographer and enjoys growing both tropical and hardy orchids at his home near Bournemouth in Dorset. He is also on the Committee of the Bournemouth Orchid Society.

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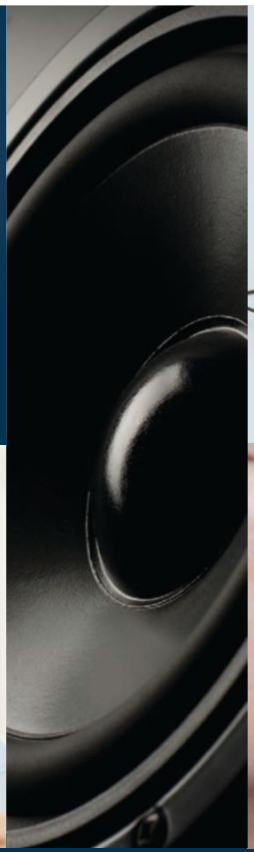
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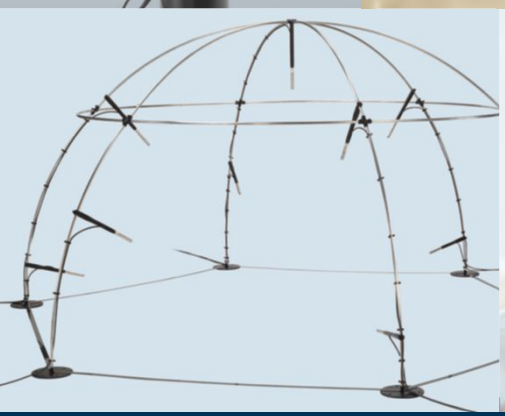
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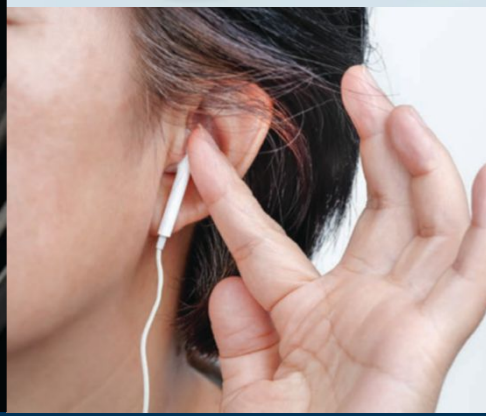












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What Was That Masked Sound Anyway?



In this article, Richard Honeycutt explains the process of sound masking, how it is used, and ways that it can occur unintentionally.

By
Richard Honeycutt
(United States)

What do the following events have in common?

- A family member sitting near a window air conditioner asks you to repeat what you just said.
- You have trouble understanding what your friend across the table in a restaurant said.
- A hearing-challenged person has more trouble understanding speech in a mall.
- People in different open-office areas can have reasonably confidential conversations.

The answer is acoustical masking. Whenever a given sound is rendered inaudible or unobjectionable by other sound, we have a case of masking. Back in the days of vacuum-tube car radios, each radio incorporated a vibrator that chopped the DC from the car's electrical system, producing a square-wave AC that could be fed to a transformer to provide high voltage for the tubes.

Vibrators made a buzzing sound that few people would describe as pleasant. Radio engineers had to know how loud the average noise in the car was, in order to know how quiet the vibrator had to be to avoid annoying the driver and passengers. This is an early example in which knowledge of masking was beneficial from an engineering standpoint.

Masking by the Numbers

Quantitatively, the amount of masking is the number of decibels by which a listener's threshold of audibility is increased by the presence of a masking sound. Let's say that in a quiet room a person is barely able to hear a pure tone at a level of 60 dB. Then a masking sound is introduced, and the tone's level must be raised to 62 dB for the person to be able to hear it. The masking effect of the noise in this case is 2 dB.

Masking noise can have any wave shape: pure tones, random noise, music, speech, and even race cars can effectively mask sound. Pure tones are especially difficult to mask, since the human hearing system is particularly good at picking pure tones

out of noisy backgrounds. But since pure tones are easy to generate, the masking of one pure tone by another sound was the earliest masking combination studied.

If a pure tone is masked by either another pure tone or a narrow band of noise, the closer to the frequency of the pure tone the center frequency of the masking sound is, the greater the masking. In **Figure 1**, the masked tone's frequency is varied. The pure masking tone has a frequency of 400 Hz, and the masking noise is centered at about 410 Hz. For both masking sounds, the level was 80 dBSPL.

Naturally, when the masked tone was centered about 400 Hz, masking was greatest. But also note that when the masking noise is lower in frequency than the masked tone, masking is more effective than if the masking noise is higher in frequency. If a tone is to be heard in the presence of masking noise, the RMS level of the pure tone must slightly exceed the level of the masking noise in a specific range of frequencies known as a critical band.

Early experimenters stated that the width in Hertz of a critical band ranges from between 40 Hz and 100 Hz for frequencies below 1.5 kHz, to 200 Hz (for single-ear listening) or 600 Hz (for double-ear listening) at frequencies between 1.5 kHz and 8 kHz. More recent tests have shown the critical bandwidth to very nearly equal 1/3 octave for frequencies above 400 Hz, and approximately 100 Hz at lower frequencies.

Masking's Cons and Pros

The examples of acoustical masking given at the beginning of this article included mostly negative effects of masking—loss of speech intelligibility due to masking by ambient noise generated by machines or people. Although we all experience this "dark side" of masking, it is particularly troublesome for people who use hearing aids, because when the aids amplify the desired sounds (speech or music), they also amplify ambient noise.

Some hearing aids attempt to compensate for this by reducing the low-frequency response of the aid. This works to a certain extent, since masking is more effective if the masking noise is lower in frequency than the masked sound, and the frequency band most crucial for speech intelligibility is the 2-kHz octave band.

Acoustical masking is often intentionally used to improve privacy in open office areas and other environments in which providing acoustical isolation is impractical. The first applications of acoustical masking used white noise—sound having equal energy per Hertz across the audible range.

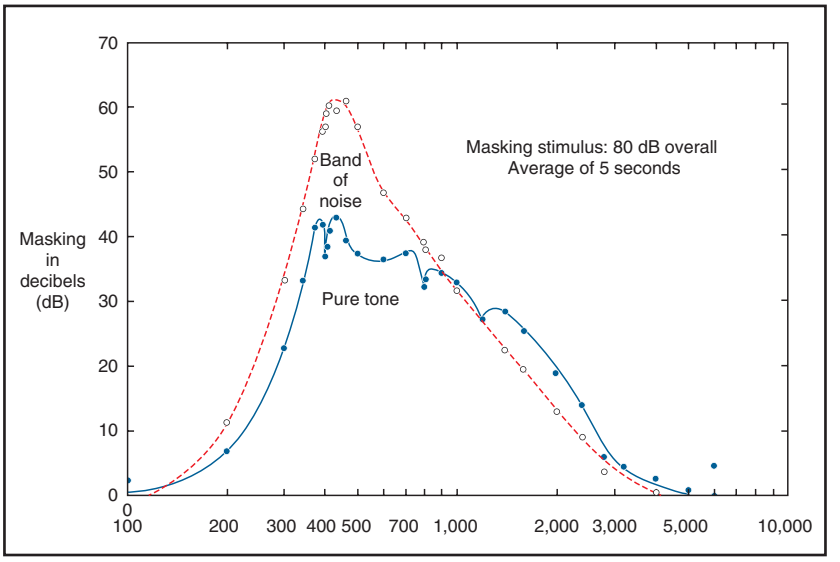


Figure 1: The masking produced by a pure tone and by a narrow-band noise, as a function of the frequency of the pure tone being masked.

An example of white noise is the sound produced by an old (pre-phase-locked-loop) FM radio when it was tuned between stations, or the sound of air from a leaking tire.

White noise is annoying to most people, so different frequency shaping is now often used for noise masking. Typically, masking noise in an open office may be set at a level of 40 to 48 dBSPL, which is somewhere between the level of a quiet library and a typical home. Masking is only used in office situations in which acoustical privacy (the ability to speak normally without unintentionally being heard) is necessary. This includes office areas for businesses and medical practices. **Photo 1** shows a self-contained sound masking system incorporating selectable white or pink noise with adjustable equalization. This unit includes power amplification and speakers.

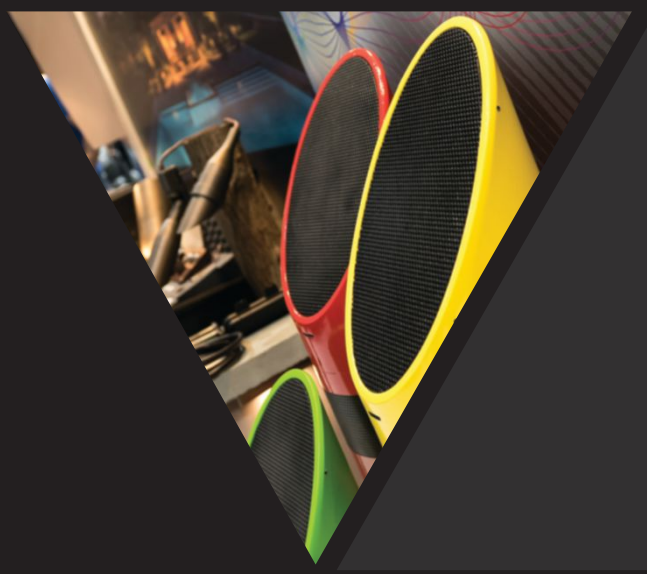


Photo 1: Atlas Sound Self-Contained Sound Masking System Self can be used in a small office or room.

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Intelligibility Issues

Masking systems should not be used in conference rooms, because the conferees' ability to understand each other will be impaired. I once had a commercial client whose complaint was that people had trouble with speech intelligibility in both of their conference rooms. Upon examining the first conference room, I found an old network router with a noisy cooling fan, lying atop one of the acoustical ceiling panels, which was acting as a sounding board for the fan noise. In the other room, I found that the open-office noise masking system also had speakers installed in the conference room ceiling.

Even after turning these masking speakers off, that conference room still suffered from intelligibility problems. But when I listened carefully to the ambient noise in the room, I noticed that transportation noise from the nearby highway was objectionable—measurements showed it to be between 60 and 65 dBA for much of the time. The intelligibility issues were all related to masking: two cases of unintentional masking, and one case of intentional but misused masking.

Another client presented a more challenging problem. She complained that she could not enjoy her living room because of "boom cars." Expecting to find poor noise isolation due to poorly sealed windows or doors, I visited her house. Since it was mid-afternoon, I did not expect to hear the offending cars, and indeed I did not. But I asked whether she had asked the police to investigate, and she said they had done so, but could not hear the cars. Things were getting more interesting. I asked what time she usually heard the cars, and she said, "About 5 PM they begin." It was almost 5 PM, so I waited. After a few minutes, she said that she could hear the cars.

The problem was not their loudness, but that they prevented her from concentrating on her activities. I could not hear the cars, but I turned on my test system, which could detect sound levels as low as 20 dBSPL at frequencies from about 4 Hz to 18 kHz. Sure enough, there was some activity showing. Just then, my client commented, "You know, it's funny: when I'm petting my cat, the noise doesn't bother me!" Remembering that the frequency range of a cat's purr is about 25 Hz to 50 Hz, I focused on that range on my test system, and that's where the noise was, although it was so low in level that most people could not hear it.

Sound isolation becomes more difficult and expensive as the offending noise goes lower in frequency, and I was pretty sure that my client would not be interested in such an extreme solution as would be required to reduce the noise to below her extraordinarily low threshold of hearing. But then the idea of masking struck me. Using a commercial powered subwoofer, I built in a low-frequency pink noise source producing a band of noise about an octave wide, centered at 35 Hz. When I place this device in her living room a little after 5 PM the next day, I was able to turn up the level until she could no longer hear the boom cars, yet she did not notice the low-level, low-frequency masking noise. The human hearing system tends to ignore constant stimuli having no discernible pattern, so the "electronic cat" was a success!

Acoustical Masking vs. Sound Cancellation

Sometimes acoustical masking is confused with sound cancellation, such as is provided by headsets designed for airplane passengers, but they are not the same. Sound cancellation involves generating sound that is out of phase with the offending noise, and injecting that "anti-sound" into the same acoustic space as the noise. The anti-sound cancels the offending noise. This approach can work with spatially well-defined noise sources such as mufflers for internal combustion engines, or with spatially limited regions, such as inside the ear cups of a headset or even the driver's head position in a car. But sound cancellation is not applicable to the office areas for which noise masking is used. In a typical open office, intelligible speech can travel 30′ to 50′, but with a good masking system, this can be reduced to 15′ to 20′. Even speech that is not completely masked can be rendered unintelligible, improving acoustical privacy.

While useful, noise masking is not the first-choice remedy to improve speech privacy. The best approach has been called "the ABC method"—absorb, block, and cover. Once good planning and layout have been included in a project, noise should be absorbed near the source where possible, then blocked by proper sound isolation measures. If problems remain, they can be covered by masking noise.



Acoustic Solutions Round-Up 2018

In the 12 months since *audioXpress* published its Acoustic Solutions Round-Up, the discipline has expanded into different segments and that's what we are trying to highlight in this new edition. Apart from traditional "first-approach" acoustic solutions, which continue to surge and have expanded from home and project-studio environments into listening rooms and conference and collaboration spaces, we are also seeing the industry address very specific acoustic problems with innovative new solutions. Contributing to the segment's expansion is the scope of intense research and new more advanced computer simulation tools, helping to bridge the gap between product design and scientific approaches. But increasingly, acoustics is also about "living spaces" and many companies are complementing their designs with sophisticated finishes, materials and finishes, both conspicuous and invisible solutions. As always, with this solutions' round-up, we hope to highlight innovations and trends, as well as bring light to the companies behind the products. In acoustics, the first rule is to seek expert assistance for an analysis of the room/space and the projected applications. Most of the companies featured in this article can support your projects, providing expert advice and a variety of acoustical solutions for all types of projects and applications.

By
J. Martins
(Editor-in-Chief

Wenger Transform Motorized Acoustical Banner

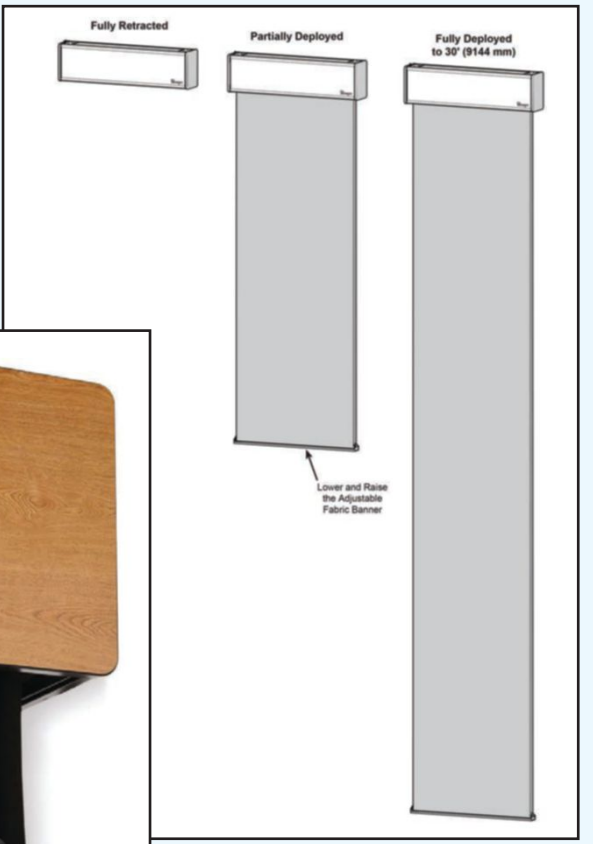
A Powerfully Transformative Solution

For performance spaces that have a need to lower the amount of reverberation, and to make the space more suitable for musical performance and the spoken word, Wenger Corp.'s line of Motorized Acoustical Banners offers the benefits of variable acoustics for stage areas and auditoriums.

With a well-engineered motorized solution, these acoustic curtains have the power to transform any space in seconds. Wenger's variable acoustical banners, can be applied in multiple ways, improving the acoustics for performers and also the audience experience. In just a few seconds, the raisers can transform the acoustics to best suit any performance, helping to control the levels of sound absorption and reflection in the installation space.

The banners are available in standard 3′ to 10′ (0.9 m to 3 m) widths, with 3′ to 5′ (0.9 m to 1.5 m) wide fabrics available in lengths of up to 30′ (9 m) long, and 5′ to 10′ (1.5 m to 3 m) wide fabrics available in lengths of up to 15′ (4.6 m) long in velour or wool only.

Available fabric options (all IFR rated) include Encore Velour, double layer 22 oz (0.6 kg), Wool Serge, double layer 30 oz (0.85 kg), or quilted absorptive material (for increased absorption), using single-layer poly with single-layer wool or single-layer velour. Each banner includes a standard "quick change" provision to allow fabric to be removed for





Designed for performance spaces, including performance halls, theaters, and auditoriums, these motorized acoustical banners are available up to 10′ (3048 mm) wide and up to 30′ (9144 mm) long.

maintenance or change the look (digitally printed, etc.). And each banner includes adjustable spacing options, both for distance away from the wall and between the two fabric layers. Setting the spacing between fabric layers and/or the distance from the wall, allows control over the amount of absorption required for optimal acoustics.

This a solution tested and proven to significantly reduce reverberation time across a broad range of high and low frequencies. And more importantly for auditoriums and rehearsal rooms, banners deploy quietly and quickly—at speeds up to 45′ (13.7 m) per minute—using Scene-Control by J.R. Clancy and other intelligent control systems from Wenger's family of brands. The simple wall mounting brackets can be delivered with different finishes, including laminated wood end enclosures.

www.wengercorp.com/acoustics/transform-motorized-acoustical-banner.php



Wenger Transforms are tubular motor driven systems for use in an operator-controlled or pre-set environment.

IsoAcoustics OREA Series

Isolation for High-Fidelity Audio Components and Turntables

IsoAcoustics, a Canadian company with its head office in Markham, Ontario, has been manufacturing and distributing isolation stands for speakers and studio monitors since January 2012. Based on patented designs, IsoAcoustics markets stands designed for the Pro Audio and home project studio, and the hi-fi and consumer market. In designing all its solutions, IsoAcoustics applies solid knowledge of physics and a scientific approach using the most sophisticated analysis and measurements tools currently available, including laser vibrometer testing in cooperation with the research facilities of the National Research Council (NRC) in Ottawa.

IsoAcoustics' speaker stands prevent the transfer of speaker cabinet energy to surrounding surfaces, effectively isolating the speaker from the desktop, bookshelf, or other surfaces. Most recently, IsoAcoustics applied all its design and manufacturing experience to create the OREA series of isolators, which apply patented IsoAcoustics design principles to provide the highest degree of vibration isolation and energy management for high fidelity audio components.

The OREA isolators remove parasitic vibration when used with amplifiers, monoblock amplifiers, DACs, CD players, speakers, turntables, and more. The upper flange of the OREAs are designed to provide a suction cup-like effect and adhere to the underside of the component, while the lower flange adheres to the supporting surface. All the energy is managed within the core of the OREAs, which are tuned to work within specific weight ranges.



Two OREA models are available in the series. The Indigo, is designed to hold up to 16 lbs (7.2 kg) per unit while the larger Bordeaux can hold up to 32 lbs (14.5 kg). The OREAs are designed to be used in multiples to address weight requirements, with a minimum of three units recommended.

www.isoacoustics.com

Acoustics First Aeolian Diffusers

The Organic Approach

Acoustics First Aeolian Diffusers use a mix of technologies to create an organic quadratic diffuser. They use a unique design process that incorporates "implied symmetry" to defeat lobing problems. The entire diffuser is asymmetric, including the edges. Each edge is designed to vary in height by less than the width of the flange gap, creating an "implied symmetry," without actually being symmetric. This feature eliminates all symmetry from the diffuser, complementing the undulating topology of the geometry and creating a unique aesthetic, while also enhancing the acoustic effectiveness. Created for use in a drop ceiling grid or as a direct wall mount, the Aeolian can be placed where needed to optimize any type of acoustic space.

www.acousticsfirst.com



Acoustics First Aeolian Diffusers may be installed into ceiling grid systems or directly mounted on a wall surface.

Auralex Acoustical Isolation Solutions for Construction

Starting From Scratch

Any acoustician would tell you that the best moment to start working on acoustics is when planning the actual construction. Auralex Acoustics, a leading brand of acoustical treatments, recently updated its full range of acoustical isolation products for construction, providing contractors with comprehensive solutions for room isolation in both new construction and remodeling. Five products in particular are essential tools for contractors to achieve the most solid acoustical isolation performance: Mineral Fiber Insulation, SheetBlok Sound Isolation Barrier, RC-8 Resilient Channel, U-Boat Floor Floaters, and StopGap Acoustical Sealant.

The best way to tighten up a room's sound, and make it more soundproof in the process, is to design it right, build it tight, and make the partitions as dense as possible. That's where Auralex's specialized Mineral Fiber sound insulation comes in. It's far denser than "the pink stuff" and is specifically designed for acoustical usage. It yields much better bass trapping, overall absorption, thermal characteristics, moisture absorption, and sound transmission loss (isolation) than other products on the market.

SheetBlok Sound Isolation Barrier is a dense, limp-mass vinyl material that is an effective barrier for stopping the transmission of sound through walls, ceilings, and floors. It is most effective when used as a component in a multi-layered construction scheme. SheetBlok helps in decoupling (i.e., floating) floors and walls and, while it's not normally considered a finish product per se, it is paintable with latex paint. Size: 4′ × 10′ rolls (40 ft^2) and 4′ × 30′ rolls (120 ft^2) Color: Black (with a smooth low sheen surface finish).

Auralex's RC-8 Resilient Channel is a specially-formed, sturdy metal device that, when used to hang drywall (instead of just attaching the drywall to the studs or joists), greatly improves the sound transmission characteristics of the wall or ceiling system. RC-8 should be installed horizontally at the bottom and top of the wall, then every 2′ or less in between. Up to two layers of 5/8″ drywall and a layer of SheetBlok may be hung on RC-8, even on a ceiling. RC-8 is available in handy 8′ lengths that are ground shippable. There are (24) 8′ pieces per box.

Made of a specially-formulated rubber compound, U-Boat Floor Floaters are Auralex's proprietary U-shaped channels used to support framing members and float (isolate and decouple) them from the surrounding structure. With the help of U-Boats, a floated room features greatly improved transmission loss (isolation) and low-frequency definition. (Translated: a tight, floated room will always sound better!) U-Boats are the industry's most affordable floating solution and are easier to use.

StopGap Acoustical Sealant is a highly elastic, water-based sealant used for reducing sound transmission at the perimeter of wall systems or around electrical box cutouts to achieve specified STC/MTC values. The surface dries to a tough film but the sealant remains flexible and stays pliable for up to five times longer than alternative brands. And unlike other products, StopGap has no residual run-off. StopGap is paintable and nonstaining, and meets ASTM C-834 standards for flame retardancy. StopGap comes in a large, 29 oz. caulking tube and requires a larger than normal-sized caulking gun, which can be purchased at any local hardware store. Coverage per 29 oz. cartridge is approximately 89′ with a 3/8″ bead applied.

www.auralex.com



Auralex Mineral Fiber sound insulation

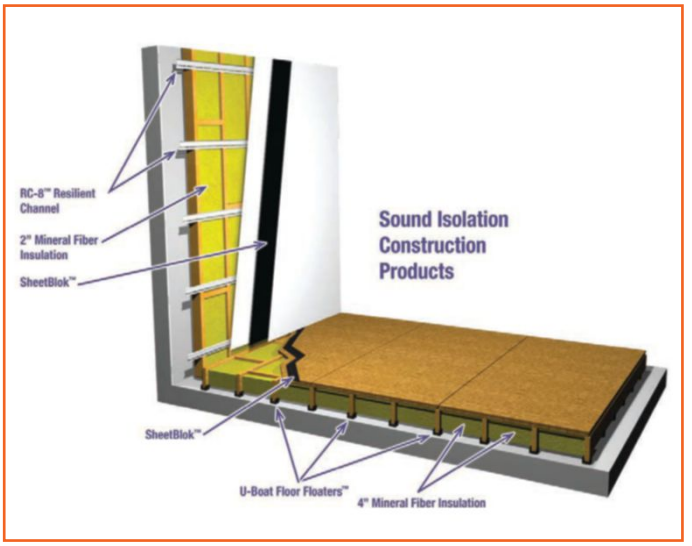


Diagram of Auralex acoustical treatment solutions for construction



Auralex U-Boat Floor Floaters



Auralex StopGap Acoustical Sealant



GIK Acoustics Portable Isolation Booth (PIB)

This Portable Isolation Booth solution from GIK Acoustics stands out as a versatile solution for any recording environment. The PIB is made of 2″ thick rigid fiberglass, effectively absorbing down to 150 Hz, yielding improved clarity. The size of the PIB makes it more effective than existing portable vocal shields. Unfolded, the PIB acts as a vocal booth; and unlike a mic shield, the PIB extends to 6′6″ tall × 3′7″ wide. Folded, the PIB is a 4′ tall × 3′7″ wide isolation screen that can be used around acoustic instruments. The most unique feature of the PIB is its thin wooden face whose stylish cutouts improve absorption while providing some diffusion. The wood side is ideal for recording and tracking environments as it will diffuse high-frequency content across the room, in turn improving sonic imaging. The PIB is available in 9 standard fabric options, or additional Guilford of Maine fabric choices, and is a Class A fire-rated solution.

www.gikacoustics.com

Acoustics Science Corp. IsoThermal TubeTrap

Isothermally Conditioned Performance

The IsoThermal TubeTrap from Acoustics Science Corp. (ASC) is an isothermal class RC time-constant bass trap combined with the adjustable treble diffusion of the company's own Standard TubeTrap. This solution, which we first highlighted in our Acoustics Focus edition back in 2016, is now a familiar sight in many listening rooms. The IsoThermal TubeTrap delivers twice the deep bass absorbing power as its predecessor. That's why the IsoThermal model is identified with two chrome dots along the rim instead of the traditional one dot. As Art Noxon, president of ASC and inventor of the TubeTrap likes to say, "Two dots means twice the power!"

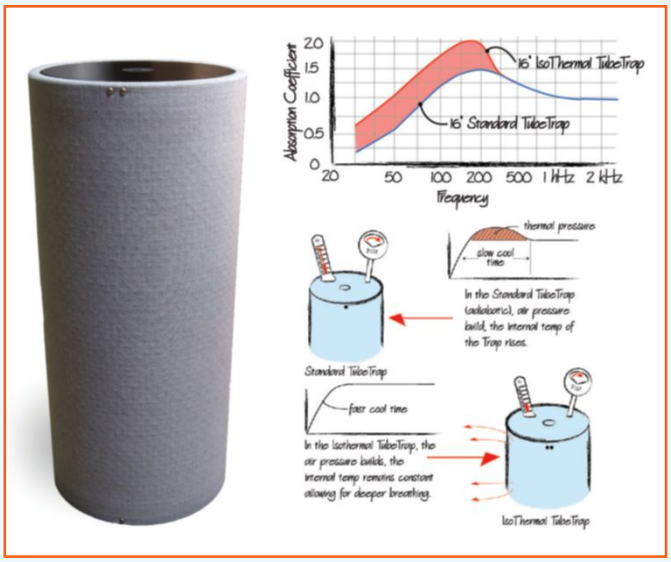
By eliminating the thermal component of acoustic pressure, the IsoThermal TubeTrap has an effectively 40% larger internal volume below 70 Hz, which literally doubles its absorbing power while retaining all of the other familiar sonic characteristics of the existing TubeTraps.

As Noxon explains, "in any listening environment, the rear wall bounce creates the most bothersome of low-frequency problems, the phase-cancel effect, in hi-fi it is also known as a 'suck-out'. If someone has an 8 dB hi-fi suck-out at 49 Hz, they usually get tired of hearing the same note drop out of double bass riffs. We developed the IsoThermal TubeTrap to be well suited for aggressively managing typical hi-fi rear wall bounce phase-cancel effects."

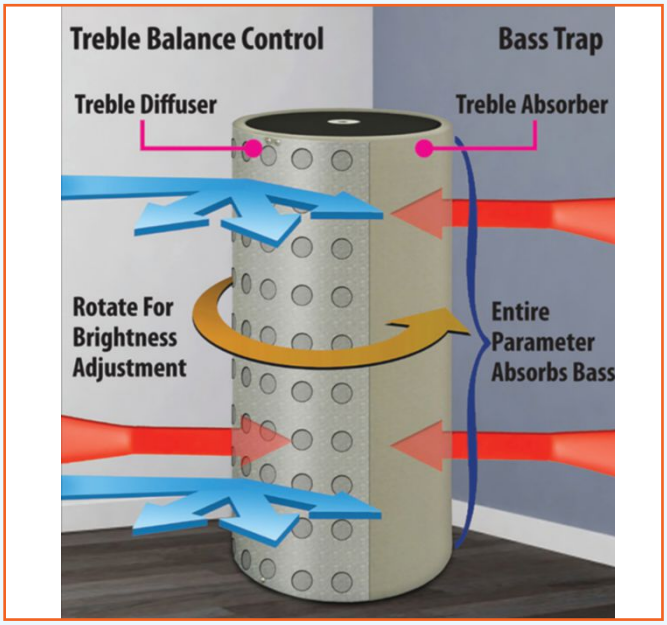
Compared to ASC's new 16″ × 4′ Isothermal TubeTrap, the low-frequency absorption coefficients of the standard 16″ × 4′ TubeTrap is doubled, while retaining all the familiar absorption and adjustable diffusion properties of the Standard TubeTrap. The 16″ × 4′ IsoThermal TubeTrap controls rear walls out to 8′. Larger traps go to lower frequencies.

How does it work? ASC laces the entire air cavity inside the TubeTrap with micron-sized fibers, which are only a few dozen microns apart. With solid surfaces so close to the gas molecules, the gas returns to room temperature very quickly, well within the amount of compression or expansion time of low-frequency sound waves. The separation of fine fibers is controlled to provide the proper isothermal efficiency crossover such that deep low-frequency absorption is enhanced without changing the familiar upper range acoustics of the TubeTrap.

www.acousticsciences.com



The ASC IsoThermal TubeTrap is available in multiple sizes and colors.



An isothermal corner loaded bass trap extends bass absorption from 30 Hz to 250 Hz with adjustable midrange diffusion above 400 Hz.

Acoustical Solutions Update

Primacoustic Hercules

Impact-Resistant Acoustic Panels

Primacoustic Hercules panels are absorption panels that are designed to withstand the abuse of environments such as gymnasiums, offices, classrooms, and other public occupancy spaces. While most acoustic panels are delicate and susceptible to damage, Hercules will help to reduce reverberation while looking like new after repeated impact.

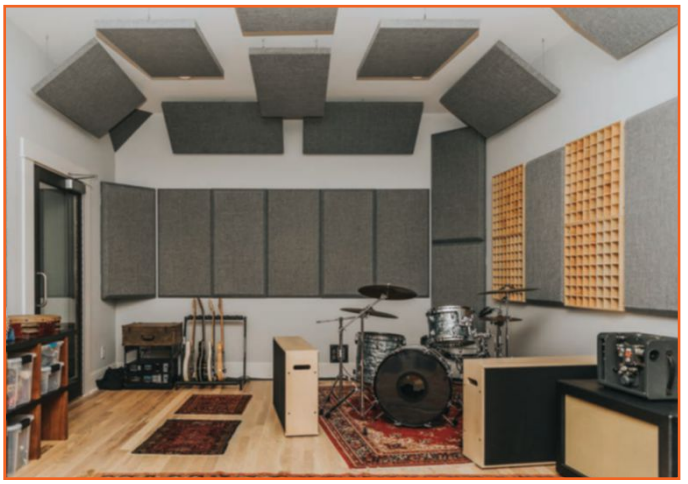
Hercules Impact-Resistant panels are manufactured from the same high-density (6 pcf/96 kg/m^3) glass wool core as Primacoustic Broadway panels, but the face of the panel is covered with a rigid fiberglass layer. This facing enables the panel to effectively absorb most frequencies, but the protection will prevent damage occurring from accidental contact. Whether someone bumps into the panel or a basketball bounces off it in a gymnasium, the Hercules panel will remain looking fresh and intact. This durable facing even allows for the use of thumbtacks in the panel, making it ideal for office spaces or schools.

Hercules panels measure 24"× 48" × 2" (60 cm × 120 cm × 5 cm) with square edges, and are available in beige or gray acoustically transparent fabric. MAP: $699.99 USD per box of six.

www.primacoustic.com/hercules



Primacoustic Hercules panels are available in beige or gray.



Producer Jacquire King's studio was recently outfitted with Primacoustic solutions, including the new Hercules panels.

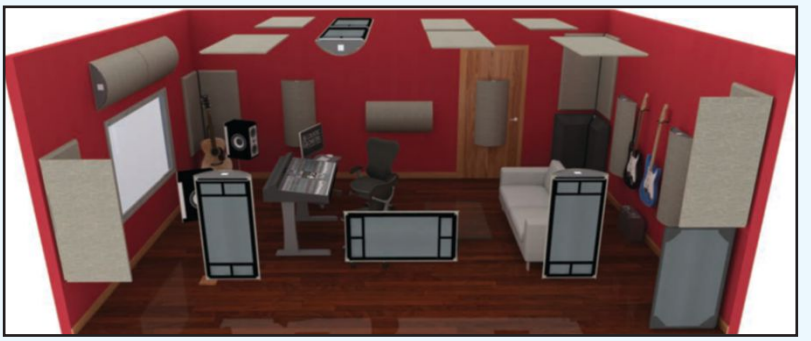


Acoustic panels are more in demand than ever, but not all rooms are as kind as recording studios! In places such as gymnasiums, schools, and offices, delicate absorption panels are easily damaged and can quickly become unsightly.

Acoustic Geometry Pro Room Packs

Curve Diffusor & CornerSorber Combination

Acoustic Geometry recently expanded its offer of first-approach acoustical solutions with a money-saving combination of the company's Curve Diffusors and CornerSorbers available for the first time in a new set of room treatment packages called "Pro Room Packs." The packs allow proper acoustical-treatment balancing in critical-listening environments, with phase-coherent diffusion and, for the first time in any combo package, full-bandwidth absorption featuring both mid-high frequency Fabric-Wrapped Panels and low-frequency membrane absorbers (Curve Diffusors combined with CornerSorbers).



Acoustic Geometry Pro Room Pack 12 is shown in a recording studio configuration.

The AcoGeo Pro Room Packs are differentiated by the inclusion of accurate, lab-proven low-frequency absorption (see John Calder's "New Research on Low-Frequency Absorption Using Membranes" article in *audioXpress*, August 2017). Pro Room Packs also feature constant-radius cylindrical diffusion for much smoother imaging and sound-energy spread than quadratic-residue-type diffusors. Starting at $3,461.76 for the Pro Room Pack 6 and ranging to $6,415.22 for the Pro Room Pack 12, these packages offer an 18% savings over individually purchased treatment products and are available for online purchase on the AcoGeo website. For easily understood videos on acoustics, visit the Acoustic Geometry YouTube Channel.

www.acousticgeometry.com

ISOACOUSTICS
Sound quality by design

GAIA

ISOLATION FOR SPEAKERS & SUBWOOFERS

The most astounding improvement For your hi-fi system



GAIA I
SPEAKER WEIGHT
MAX: 220LBS (100KG)

GAIA II
SPEAKER WEIGHT
MAX: 120LBS (54KG)

GAIA III
SPEAKER WEIGHT
MAX: 70LBS (32KG)

isoacoustics.com

Product Design

GAIA isolation are precision machined to provide a high degree of speaker isolation. The IsoAcoustics patented design resists lateral movement and oscillation to maintain alignment with the listening position and provide greater clarity and focus. Internal reflections are attenuated resulting in a sound stage that blossoms open.

"…the biggest improvement in sound I have ever heard on the system. Better than changing cables or electronics, the noise floor was lower, the focus was better, and the bass went deeper."
Harry Weisfeld, Founder, VPI Industries

"…imaging is even more amazingly precise. Truly incredible!"
Nick Squire, Chief Sound Engineer, Boston Symphony Orchestra

"… I expect to see more of these packages become a "required option" for larger floor mount monitors."
John Storyk, Founding Partner of Walters-Storyk Design Group

Primacoustic Paintables Turn an Acoustic Panel into an Art Piece

Unhappy with the acoustics in his winery's "Viewing Room," Sandhill Winemaker Howard Soon went in search of a solution and in the fall of 2017, the space he uses for private tastings was transformed from a highly reverberant but stylish room to a room with the acoustics very much under control. This was artfully accomplished using the Primacoustic Paintables line of products including a unique application where the panels were printed to serve not only as acoustic attenuators, but also as artwork perfectly suited to the venue.



Primacoustic Paintables acoustic panels control acoustic reflections on the glass wall portion of the viewing room.

Primacoustic Cloud Paintables and Multi-Shape Panels

The Primacoustic range of Paintables absorptive acoustic panels now includes a collection of exciting shapes that can be used as ceiling clouds, baffles, or wall panels to reduce reverberation and improve intelligibility. The unique shapes and sizes open the door for architects, designers and installers to present clients with creative solutions that are bold, yet architecturally pleasing.

Cloud Paintables include the circular Cirrus, the hexagonal Hexus, and the square Altos. Each style is available in 36" (91 cm) and 48" (120 cm) versions, and are 1.5" (3.8 cm) thick. All panels are made from high-density glass 6 lbs per cubic foot (96 kg/m^3) glass wool for exceptionally even absorption, and each is completely encapsulated in micro mesh with resin hardened edges. Cloud Paintables are factory finished in Absolute White latex on all sides and are ready for mounting. Unique to their design is a proprietary textured surface that enables users to lightly spray paint the panels onsite to match the décor without affecting acoustical performance.

The included Helix cloud anchors and SlipNot suspension cables allow the cloud panels to be suspended horizontally from the ceiling. Cloud Paintables can also be wall mounted using Surface Impalers or installed as vertically hanging baffles using Corkscrew anchors (sold separately). As with all Primacoustic panels, Cloud Paintables are laboratory tested for acoustic performance as well as fire safety, achieving the most stringent Class-A/1 rating in North America and Class-B in Europe, making them safe for use in all types of installations. Each Cloud Paintables kit includes two high-density glass wool panels and all hardware required for mounting as horizontal suspended clouds. Additional hardware for wall/baffle mounting sold separately.

www.primacoustic.com/cloud-paintables







Cloud Paintables look great when used on the ceiling of restaurants, lobbies, gymnasiums, or any other space where unobtrusive acoustic treatment is desired.

Acoustics Vendor and Services Directory

Acoustic Fields
www.acousticfields.com

Acoustics First
www.acousticsfirst.com

Acoustics in a Box
www.acousticsinabox.com

Acoustic Manufacture
www.acousticmanufacture.com.pl

Acoustic Systems
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Acoustic Sciences
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www.proaudioacoustics.com

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www.acousticgeometry.com

Artnovion Acoustics
www.artnovion.com

Aural-Aid
www.auralaid.com

Auralex
www.auralex.com

ClearSonic
www.clearsonic.com

Delta H Design
deltahdesign.com

Eccentricity of Wood
www.woodblocks.design

GIK Acoustics
www.gikacoustics.com

Golden Acoustics
www.goldenacoustics.com

G&S Acoustics
(Golterman & Sabo)
www.gsacoustics.com

International Cellulose Corp. (ICC)
www.spray-on.com

Jocavi Acoustic Panels
jocavi.net

Listen Audio Systems
www.listen-speakers.com

Media Specialty Resources
www.msr-inc.com

Overtone Acoustics
www.overtoneacoustics.com

Pinta Acoustic
www.pinta-acoustic.com

Primacoustic
www.primacoustic.com

Prosocoustic
www.prosocoustic.com

Ready Acoustics
www.readyacoustics.com

RealTraps
realtraps.com

Resolution Acoustics
www.resolutionacoustics.com

RPG Acoustical Systems
http://rpgacoustics.com

Silent Source
www.silentsource.com

Sonex
www.sonex-online.com

Vibrant Technology
www.vibetech.com

Vicoustic
www.vicoustic.com
www.audioplusservices.com

VocalBooth
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COMSOL Multiphysics Acoustics Module

Making the Case for Acoustic Modeling and Simulation Apps

Acoustic phenomena are multiphysics in nature. When building a model, engineers must account for several types of physics and their coupling at different scales and frequencies.

By
Mads J. Herring Jensen
(COMSOL)

With increasingly complex systems and tighter project deadlines, acoustical engineers are turning to numerical simulation software to get the job done. By using computational tools, design tasks can be accelerated and the need for costly and time-consuming physical prototypes can be reduced. Acoustics simulation also increases the understanding of a design, leading to better informed decisions and higher-quality products.

To reap the benefits, what capabilities are important in acoustics simulation? Applications often include the reproduction, propagation, and reception of sound signals under diverse conditions. This includes not only the interaction of the sound signal with structures, porous materials, and flow, but also modeling the transducers involved in the generation and detection of the sound signals. All these are multiphysics problems by nature that acousticians have to consider for the efficient development of new products and technologies. This places a critical requirement on the modeling software in terms of the ability to couple physics effects relevant to the full system.

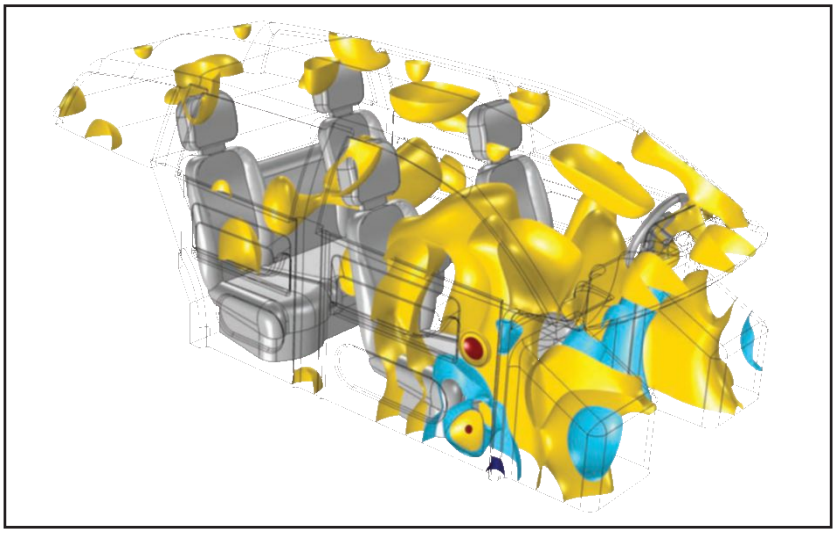


Figure 1: This acoustics simulation of a sedan interior includes sound sources at the typical loudspeaker locations. Results show the total acoustic pressure field inside the cabin.

Current Technological Challenges in Acoustics

Sound quality is a trending topic in many industries. This concerns the reproduction of sound inside—for example, car cabins (see **Figure 1**)—or the output from the exhaust and muffler systems. Other examples include the performance and optimization of headphones and loudspeakers or the speaker system of mobile devices. In all of these cases, a detailed understanding of both sound propagation and transducer behavior is necessary to optimize the systems. Clever digital signal processing is no longer enough to make systems behave and "sound good." For example, to improve the performance of hearing aids using adaptive feedback canceling, a coupling of a miniature loudspeaker vibroacoustics model with an acoustic and solid mechanics finite element (FE) model is needed for producing accurate simulation results.

In the loudspeaker industry, a standard driver design has reached the limit of where improvements can be done by simple trial-and-error testing (see **Figure 2**). Optimization requires detailed numerical analysis. Miniature loudspeaker systems are now driven at such high sound pressure levels (SPLs) that distortion and attenuation due to nonlinearities are introduced. The same nonlinearities also play a significant role in liners in aerospace applications. Another example

involving a multiphysics coupling—electrostatics, structural membranes, and thermoviscous acoustics—is the modeling of condenser microphones. The physics are tightly coupled and necessary for a correct prediction of the microphone sensitivity.

The Acoustics Module

The Acoustics Module, an add-on product to COMSOL Multiphysics, is ideally suited for modeling the many frequencies involved in acoustics, ranging from infrasound to ultrasound, as well as the multiscale nature of acoustics when dealing with, among others, thermoviscous loss mechanisms or aeroacoustics. The latest release of COMSOL Multiphysics software version 5.3a introduced the boundary element method (BEM) in the Acoustics Module as a physics interface. This new functionality was designed to enable engineers to model large acoustical radiation problems, such as exterior sound reproduction for electric cars. Acoustic engineers will get unprecedented modeling power by being able to analyze the full range of acoustic frequencies from the lowest bass notes to ultrasound (see **Figure 3**), in addition to all of the possible multiphysics couplings available between the different physics, which are set up seamlessly in the same modeling environment.

Acoustics Simulation Apps

To tackle the acoustic challenges faced by many in the industry, users without previous simulation software experience can run apps specifically tailored for them and their needs with predefined inputs and desired outputs. This is possible using apps created with the Application Builder available in COMSOL Multiphysics. Simulation apps are multiphysics models wrapped in a custom user interface. With this tool, specialists can package a complex simulation and allow users to change design parameters and analyze results autonomously with respect to industry standards and customer requirements (see **Figure 4**).

Thanks to a local installation of COMSOL Server, apps can be easily deployed to colleagues and customers. Users can connect via the COMSOL Client or a major web browser. It has never been easier for simulation specialists to model acoustic devices with such high fidelity and let their colleagues benefit from their work.

About the Author

Mads Jakob Herring Jensen works at COMSOL as Technical Product Manager for the Acoustics Module of COMSOL Multiphysics, based out of Copenhagen, Denmark. He is responsible for specifications, planning, documentation, and the general development and management of the acoustic capabilities of COMSOL Multiphysics simulations software.

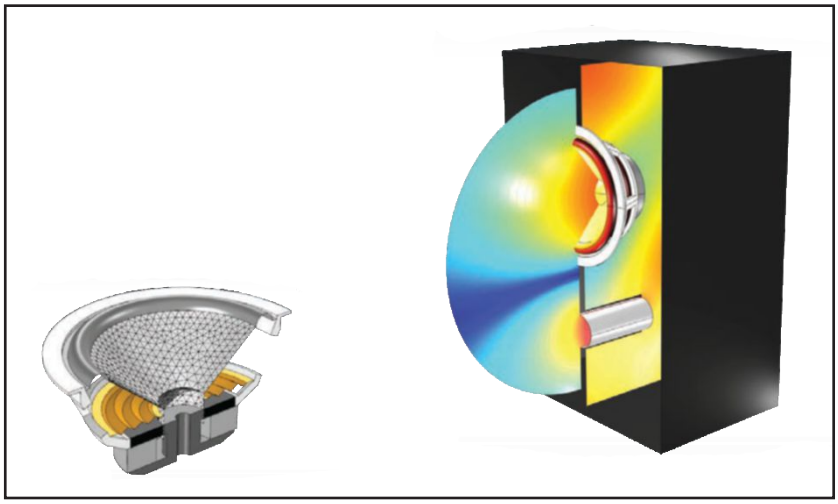


Figure 2: The simulation results show the sound pressure distribution in a loudspeaker driver enclosure.

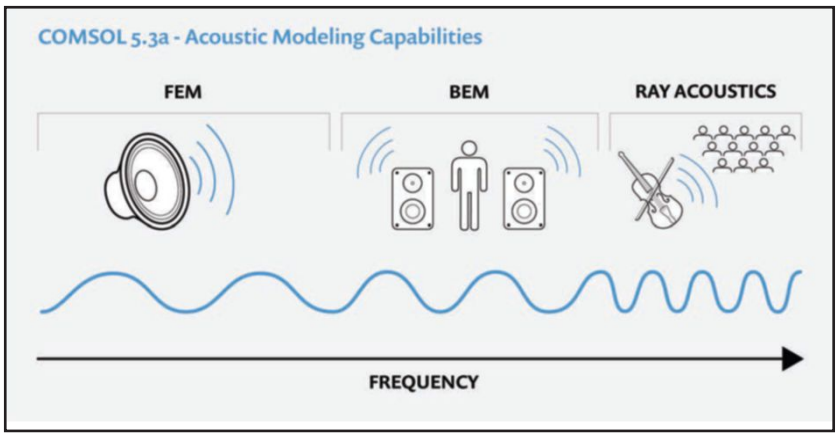


Figure 3: By combining boundary element, finite element, and ray acoustics analysis in a multiphysics environment, COMSOL Multiphysics delivers unprecedented acoustic modeling capabilities.

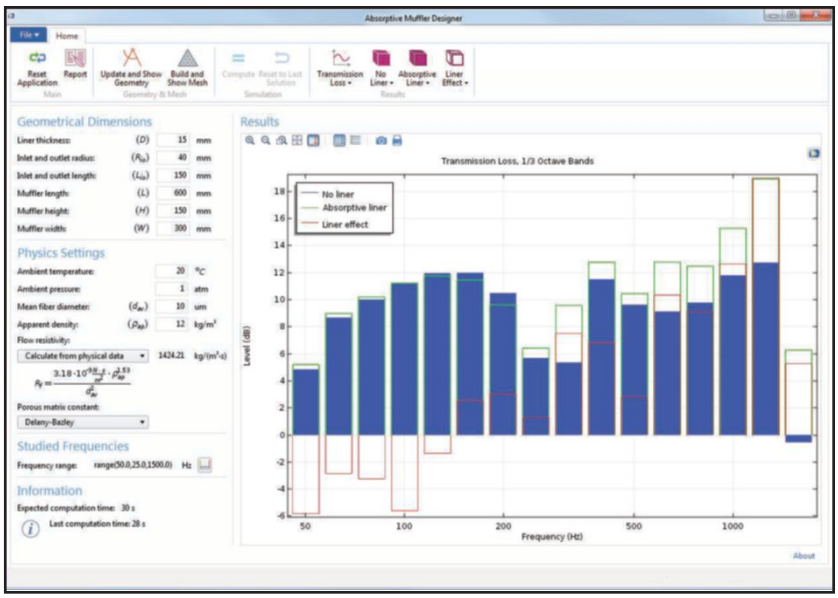


Figure 4: This example app is based on a COMSOL Multiphysics model of an absorptive muffler. The user may change the geometric design of the muffler, the ambient temperature and pressure, and material properties in order to evaluate the resulting acoustic behavior.

Curing the Open Office Noise Epidemic...With Nature?

Noise distraction is the number one complaint for open plan office workers according to a recent study from Oxford Economics. After experiencing its own challenges with noise in an open office, Plantronics tested and developed its own solution to noise and distraction using biophilic acoustics or acoustics-based on elements from nature. The article details how Plantronics arrived at its Habitat Soundscaping solution.

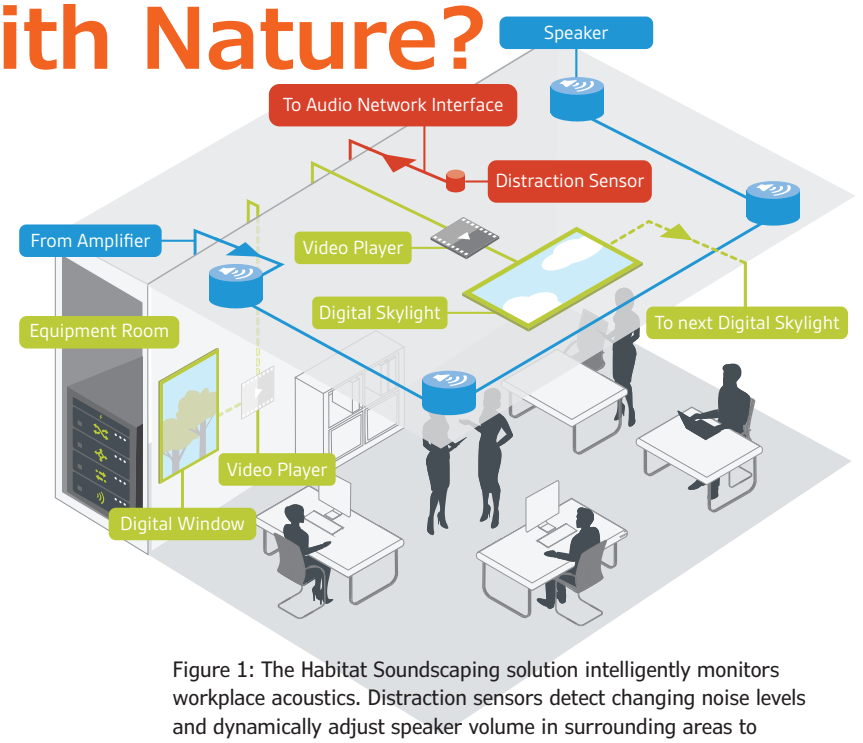


Figure 1: The Habitat Soundscaping solution intelligently monitors workplace acoustics. Distraction sensors detect changing noise levels and dynamically adjust speaker volume in surrounding areas to support collaboration without disrupting nearby focused work.

By
Robert Manassero
(Global Product Marketing Manager, Plantronics)

Open offices with beautiful designs are making headlines around the globe. However, the human experience within those open offices can often be overlooked, especially when it comes to how those offices sound. Plantronics moved its Santa Cruz, CA, headquarters to an open plan office a few years back, and employees immediately began feeling distracted by noise.

In our quest to solve the noise problem in the open office for ourselves, we developed a unique, industry-first solution called Habitat Soundscaping. Habitat Soundscaping, which Plantronics announced in July of 2017, combines nature-inspired audio and visual elements to engage and immerse people with both sight and sound. Coupled with room-based, intelligent software, it actively works to mitigate distracting noise (see **Figure 1**).

On Nature's Path

The journey to get there was not what we expected. As technologists, we arrive at solutions to our customers' problems through rounds of research, big sky thinking, prototyping, and testing until we get it right. After years of developing services for enterprise communications, we believe in what we call a Smarter Working approach that puts people first by recognizing, accommodating, and nurturing their various work styles—whenever and wherever they work. Smarter Working not only informs the products and services we create but is a philosophy that we practice in our own work environment.

We knew our background in acoustic technology and Smarter Working would help us address distracting noise in our office. But what we didn't know is that we'd be traveling down nature's path to get there.

A Case for Biophilic Design

Our research continued to show that access to nature can help sharpen thinking, restore energy, and help reduce stress. As human beings, we have an innate desire to connect with nature, yet we're inside most of the time. According to the US Environmental Protection Agency (EPA), the average American spends 87% of the day indoors and nearly as much time in the car (6%) as outside (7%). It became clear that nature in some form was going to be an integral part of our solution.

We found that a key ingredient for curing open office noise is related to biophilia. I have no doubt when you read the words "biophilia" or "biophilic science" you might think I've just shared a bit too much

about my medical history! But despite its somewhat medical-sounding name, biophilic design is a design principle that incorporates the benefits between nature and humans into the built environment. According to a report by Human Spaces, introducing biophilic elements to an office environment can help induce powerful benefits, such as:

- 15% increase in well-being
- Up to 6% increase in work productivity
- 15% increase in creativity

Research has proven natural water sounds are the most effective means to reduce the impact of intelligible speech while also creating a comfortable, rejuvenating environment for work. Imagine the sounds of a gentle waterfall or a babbling brook. Research also shows that bringing visual connections to nature into the office can improve cognitive functioning and increase well-being. Think images of birds, creeks, waterfalls in digital form or in-office water walls. In our Habitat Soundscaping solution, nature-based visuals complement the nature-inspired audio to create a harmonious, multisensory experience (see **Photo 1**).

Introducing Intelligent Software to the Solution

By combining biophilic elements, specifically the sights and sounds of nature, with highly sophisticated, adaptive, and data-driven acoustic software, you end up with a solution that ensures distracting intelligible speech is dramatically reduced. This results in a peaceful, welcoming environment where focus work and collaboration can happen in the same place, at the same time and where employees can thrive (see **Photo 2**).

We've incorporated software into our Habitat Soundscaping solution using complex algorithms to continuously respond to changing levels of distracting conversation. We recently added real-time visualizations of office acoustics to show noise levels, distracting conversation levels, and how the Habitat Soundscaping system is responding to distracting levels of conversation. It's a huge difference when compared to filtered noise, which are known to trigger health issues such as headaches and fatigue.

Although I'm sure we'd all agree that companies that create open work spaces do so with the best of intentions, but visually stunning designs with open floor plans and challenging acoustics can very often lead to unhappy, and even absent employees. According to a Gensler workplace survey, over half the employees working in open offices are disrupted by others when trying to focus. And for the employees who do come into the office, a study by the University of California, Irvine shows each interruption costs employees an average of 23 minutes to get back on track.

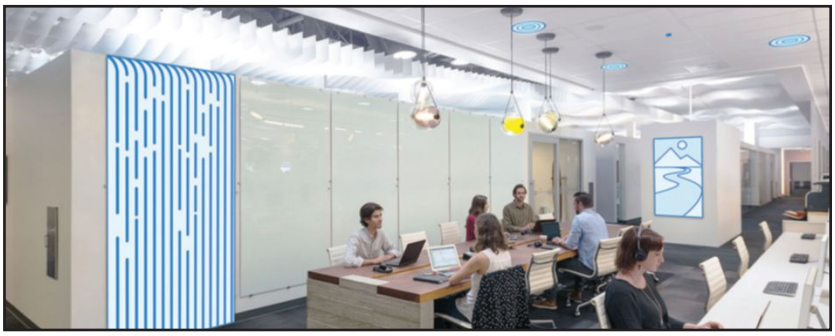


Photo 1: Habitat Soundscaping elements work together for a holistic approach to managing the noisy distraction within the office. Natural elements or biophilia combined with intelligent software help create an atmosphere that encourages wellness and productivity.

A noisy work environment can even impact the war for talent. A 2018 Plantronics study with Oxford Economics found that millennials are particularly annoyed by office noise and that people in the noisiest offices are most likely to say they'll leave their job in the next six months. This is vital data for Human Resources departments since according to a report by Human Spaces, one-third of office workers say the design of an office impacts their decision to work at a company.

We're excited to be the first company with an offering that ensures organizations no longer have to choose between having a beautiful, modern open office or a peaceful, calm and collaborative workplace. Our hope is that this solution will help companies create productive environments that also support employee wellness. In October 2017, Frost & Sullivan honored Plantronics with the 2017 North American Augmented Reality Technology Innovation Award for Habitat Soundscaping. In the related award report, Frost & Sullivan notes, "Plantronics has identified a significant business problem and then applied cutting-edge technology to address that problem. Its Habitat Soundscaping

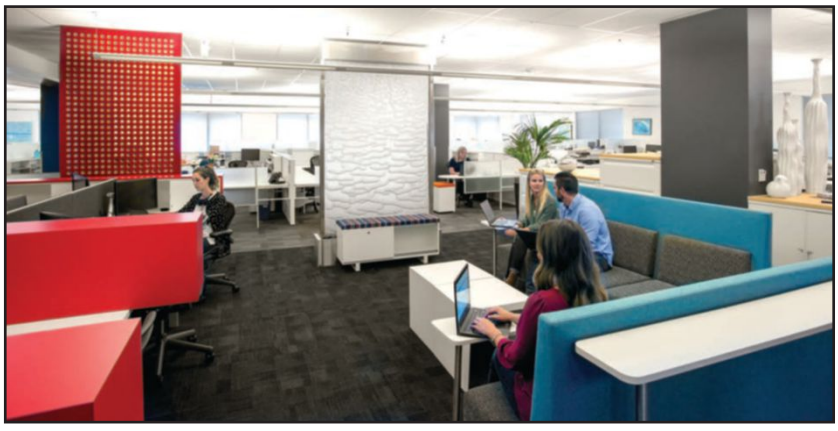


Photo 2: Habitat Soundscaping provides a more productive environment.

About the Author
Robert Manassero is the Global Product Marketing Manager at Plantronics and currently leads the innovations marketing team. In this role, Manassero is responsible for bringing new business models to market and all associated marketing efforts. Manassero is a workplace technology and Unified Communications and Collaboration subject matter expert with a deep understanding in biophilic design and office acoustics. Manassero is a graduate from the University of Pacific and holds a Masters degree from San Jose State University.

service allows businesses to reap the rewards of open office spaces while minimizing the problems associated with them."

Habitat Soundscaping was also recently recognized at Integrated Systems Europe (ISE), winning the 2018 Top New Technology Award in the Audio Distribution Systems category. The service was also recently named a finalist for the Best of Enterprise Connect Award 2018.

A Deeper Look at the Habitat Soundscaping System

Now it's time to get under the hood of our system. Habitat Soundscaping consists of core elements that make up our holistic approach to managing noisy distractions within the office. Those elements include sights, sounds, intelligence, and a cloud-based subscription.

Sounds

Soothing sounds of nature create a harmonious environment that boost overall wellness and leave people feeling creative and energized. We built our audio from the ground up by heading into the outdoors and recording water sounds at various running streams. To make sure we had the audio right, we then compared the water sounds we recorded to a sound template we tested as being effective at reducing intelligible speech.

We tuned the water sounds to provide more energy in the frequencies important to speech intelligibility (1 kHz to 4 kHz), then ran the sound over high-pass filters to cut out low-frequency artefacts. Our audio tracks are precisely engineered to match the visual displays of the system so that it delivers a realistic natural experience.

This audio is carried through in-ceiling speakers that dynamically adjust to changing conversation levels, which are strategically placed throughout a workspace. In-ceiling sensors detect distracting speech, quantify its distractive power, and dynamically adjust speaker volume in surrounding areas.

Although acoustic treatment can help—we've designed the system to work in any variety of open office environments with varying acoustic parameters. These can include open ceilings, different acoustic presence of acoustic wall panels, type of carpet, and varying cubicle partition heights.

Sights

Real water features complement the nature-inspired audio tracks and provide the innately soothing presence of running water. We offer three unique waterfall designs to augment a work space based on needs (see **Photo 3**). Each waterfall comes in three different dimensions (8′ × 4′, 9′ × 5′ and 10′ × 5′) and includes a two-year limited warranty.

An open mesh waterfall allows water to flow over a woven stainless-steel mesh surface. The result is remarkable cascading wave patterns, which makes this a beautiful addition to any space. This offering includes a ultraviolet (UV) sterilizer and reverse osmosis filter. For a relatively easy water feature to install and maintain, we also offer an enclosed waterfall. In this feature, water flows between two sheets of acrylic, adding movement and energy to a static space. Our third option is an open waterfall where water flows down a single, clear sheet of glass and creates a visually stunning water feature. This last version also comes with a UV sterilizer and reverse osmosis filter.

If a water feature is not the right fit, the Habitat Soundscaping solution also has the option to use



Photo 3: Open mesh, enclosed, or open flow waterfall designs offer options to augment a work space based on needs.



Photo 4: The Habitat Soundscaping solution includes virtual displays designed to look like conventional windows.

virtual displays of serene landscapes including babbling brooks in wooded settings. These displays integrate with natural sounds to deliver a rich, multisensory experience. Displays include ceiling-mounted flat panel monitors that show natural scenes and flat panel displays designed to look like conventional windows. Each window provides a natural vista, bringing light and an inspiring view of nature into the office (see **Photo 4**).

Intelligence

Intelligent software acts as the central hub for the Habitat Soundscaping service. Complex algorithms continuously monitor distraction sensors and adjust speakers to seamlessly maintain perfect, natural harmony within the space. The software's user interface gives users the ability to change a theme, toggle Adaptive Soundscaping Intelligence, control or mute the volume, view audio zones overlaid on floor plans, adjust scheduling and set the health check feature (see **Photo 5**). And the recently released acoustic heat map capabilities deliver real-time insights into speech distraction levels in the office and visualizes how the system responds to changing noise and conversation levels in the open office, adjusting volume to support collaboration without disrupting nearby focused work (see **Photo 6**).



Cloud-Based Subscription Service

Plantronics offers a cloud-based subscription service, at one-year, three-year and five-year increments. The service provides access to Adaptive Soundscaping intelligence to dynamically optimize the open office environment by adjusting to changing conversation levels. Access to data insights using acoustic heat maps and automatic health checks, which ensure Habitat Soundscaping is operating efficiently. Subscribers also receive a library of Habitat Soundscaping audio content and a selection of digital window and digital skylight scenes.

Putting Habitat Soundscaping to the Test

In August and September 2017, we tested the Habitat Soundscaping solution at Plantronics offices in Hoofddorp, near Amsterdam, Netherlands. Our goal was to understand the system's efficacy in reducing speech intelligibility in an open office and how employees would accept it. The biophilic soundscape was tested against a control (no distributed audio) and against two other sound files using a distributed audio system, both of which were filtered noise sound files.

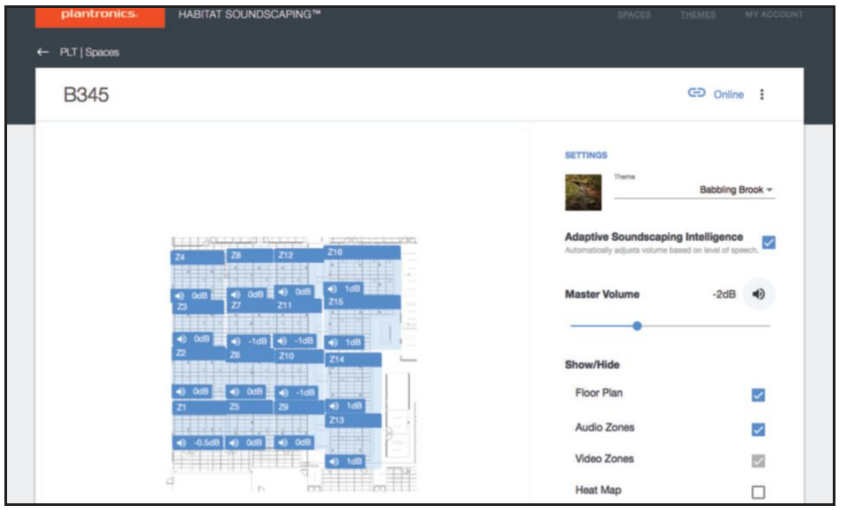


Photo 5: The Adaptive Soundscaping intelligence dynamically optimizes the open office environment by adjusting to changing conversation levels.

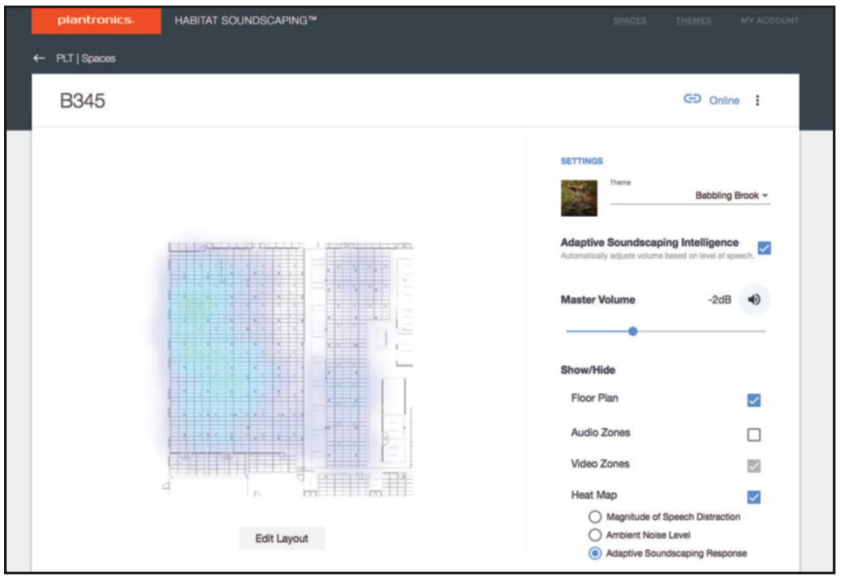


Photo 6: A heat map shows noise levels within the office setting and how the Adaptive Soundscaping system is responding.

What we were looking for was a reduction in Speech Transmission Index (STI) at workstations in the space and deriving the distraction distance from the results. We surveyed employees that used the space before and after installation of the system to understand their levels of acceptance.

The testing showed that Habitat Soundscaping was the most effective at reducing STI when compared to the filter noise sound files. Significant improvements to the distraction distance (rD), were measured when a biophilic soundscape was deployed. Our findings concluded that rD reduced from 59' with biophilic sound off to 23' with biophilic sound turned on.

Going further, analysis of the employee responses shows that the biophilic soundscape system was rated (as an average) between 0.7 and 1.0 points better (on a scale of 1 to 5) than no distributed audio for three key acoustic environment and performance questions.

We took this employee satisfaction research a step further and engaged Leesman, a global firm that examines how workplaces affect employee and

Resources

"2017 North American Augmented Reality Technology Innovation Award," https://habitat.plantronics.com/wp-content/uploads/2018/05/frost-sullivan-habitat-soundscaping.pdf, 2017.

Habitat Soundscaping, Plantronics, https://habitat.plantronics.com

"The Global Impact of Biophilic Design in the Workplace," Human Spaces, 2015, www.humanspaces.com.

"U.S. Workplace Survey 2016," Gensler, 2016. www.gensler.com/uploads/document/442/file/gensler_us_wps_2016.pdf.

"The Unmet Promises of the Open-Plan Office," Oxford Economics, 2018, http://blog.oxfordeconomics.com/unmet-promises-of-the-open-plan-office.

"The Cost of Interrupted Work: More Speed and Stress," Gloria Mark, University of California, Irvine; Daniela Gudith and Ulrich Klocke, Humboldt University, Berlin, Germany, https:// www.ics.uci.edu/~gmark/chi08-mark.pdf, 2008.



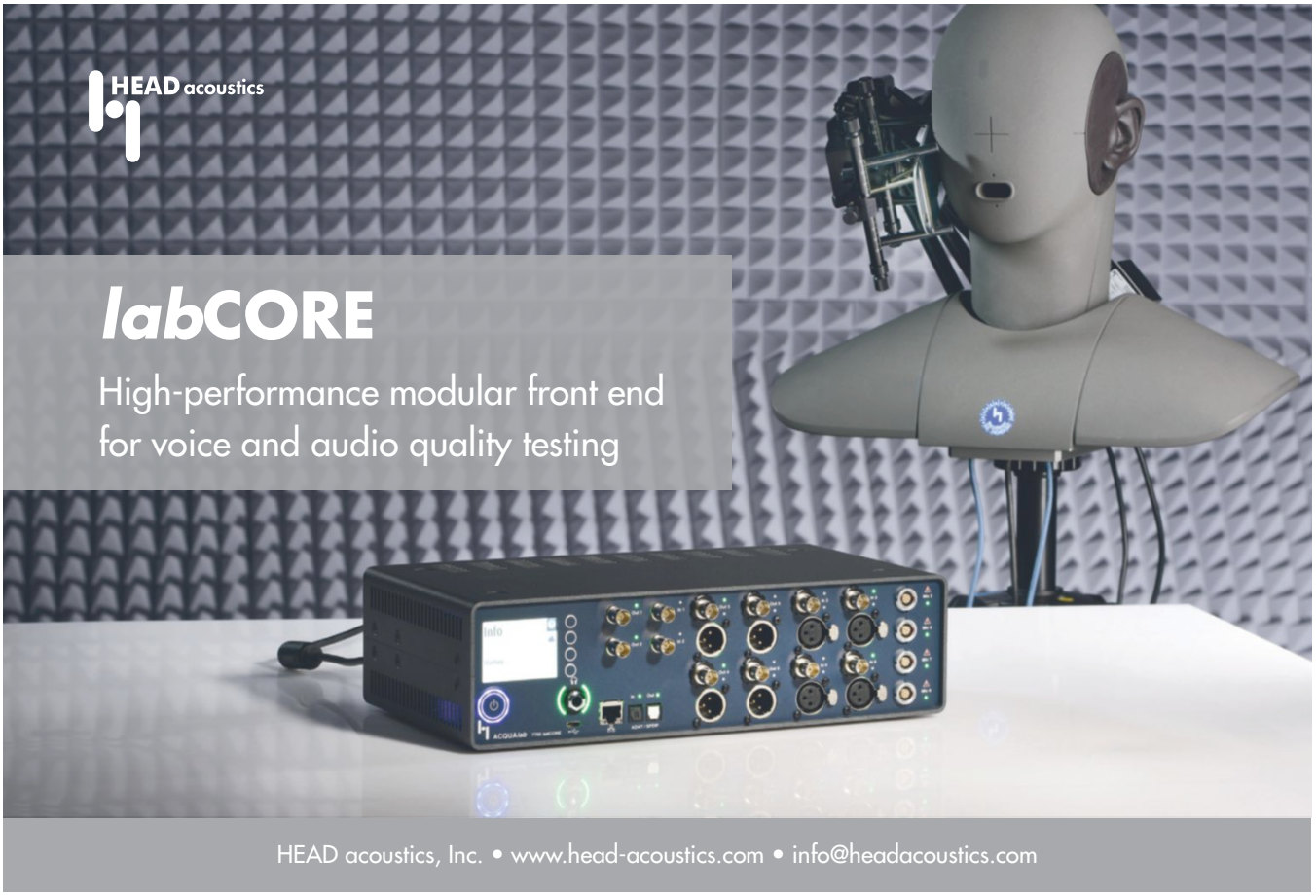
Photo 7: The solution connects with nature and transforms the open office space into a holistic, natural experience.

organizational performance. Leesman conducted research within our Hoofddorp office where the Habitat Soundscaping service is a core part of the office experience.

Leesman determines ranking using its Leesman Index, a global business intelligence tool which captures employee feedback on how effectively the workplace supports them. The Leesman Index measures a variety of factors including employee satisfaction around environmental features such as acoustics, temperature and natural light in the office (see **Photo 7**). The survey revealed that Plantronics employees agree that the ability to reduce noise is vital to creating an effective workplace, ranking it fourth among the most important office features.

Plantronics Hoofddorp office garnered the highest level ranking possible by Leesman, known as Leesman+, which is awarded to an elite group of workplaces that score 70 Lmi* or above on Leesman's standardized performance scale. The score places the office in the top three Leesman+ buildings in Europe and makes it the highest scoring Leesman+ building in the Netherlands. Only 6% of all workplaces measured by Leesman have received the Leesman+ certification.

To learn more about biophilic science and Habitat Soundscaping, visit the company's website at www.habitat.plantronics.com.



Measuring the Performance of Diaphragm Absorbers

This article aims to describe empirical and finite element analysis prediction of the performance of weighted diaphragmatic membrane absorbers, the predominant acoustic core employed in Artnovion low-frequency absorbers, such as the "Corner Bass Trap" and the "Sub Trap" range.

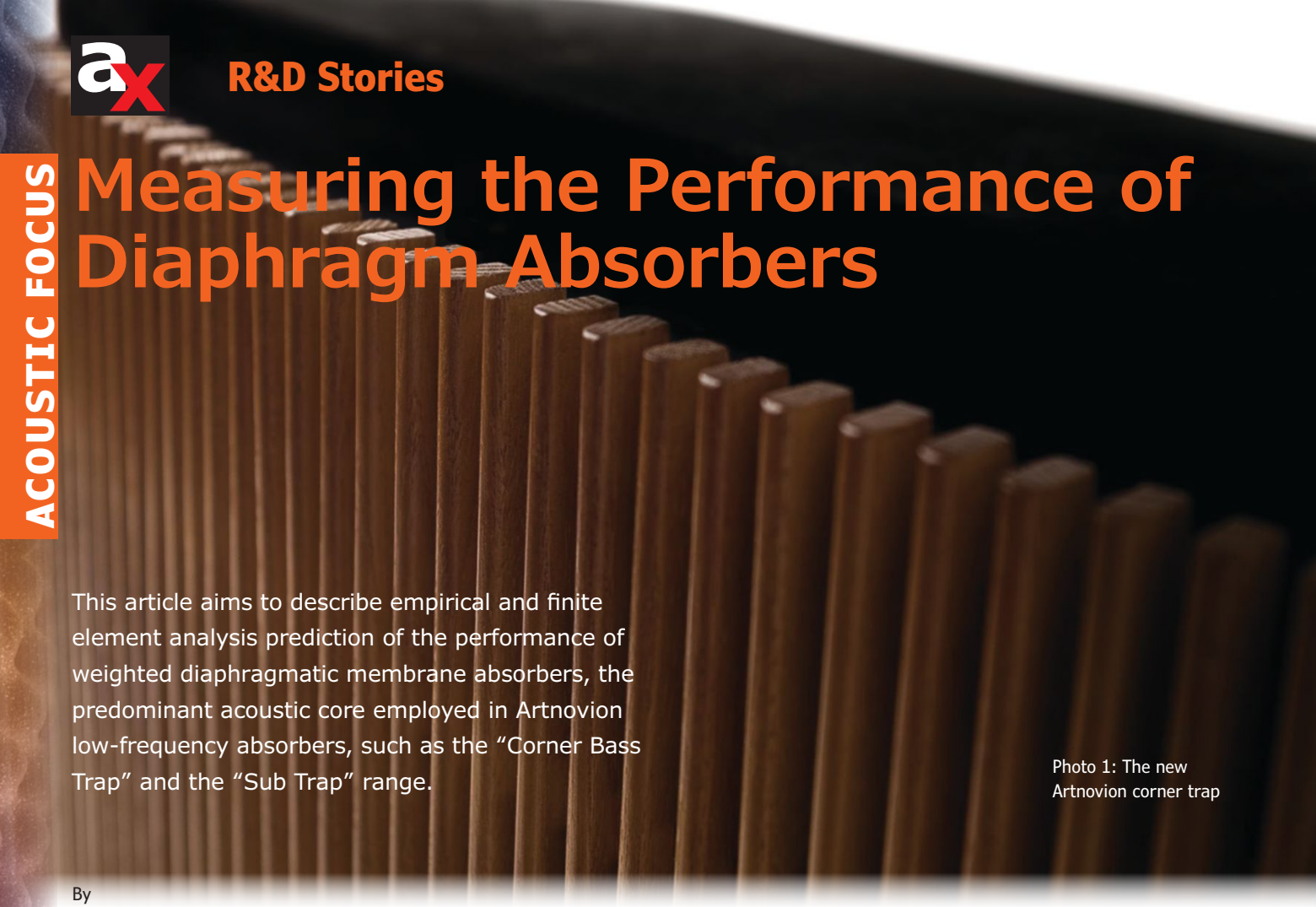


Photo 1: The new Artnovion corner trap

By
Jorge Castro, Nathaniel Bailey
(Artnovion)
Bruno Fazenda, Kelvin Griffiths
(Sense Research, Electroacoustic Design)

This technology was developed to address the market demand for precise low-frequency absorption. Mid-high frequency absorption is easily achieved through broadband porous absorbers and is readily available on the market—solutions for frequencies from 60 Hz to 120 Hz are considerably rarer and require distinct absorption techniques. Treating even lower frequencies usually involve either vast custom-built solutions or structural alterations to the space.

The goal was to create a high-performance absorber with a tunable resonant frequency that could effectively work down to 40 Hz. The tunable element is essential, so a mass-produced product could be adapted to target each rooms' characteristic resonant frequencies.

A New Design Approach

To tackle these criteria, a novel approach to low-frequency absorber design was required (see **Photo 1**). Porous absorbers utilize a very large volume to reach a comparable performance and have an uneven absorption range—enough material to be effective at low frequencies will in turn have an exaggerated effect at high frequencies. Resonance absorbers (e.g., Helmholtz resonators) are more suited to actuate on acoustic pressure, although tuning Helmholtz resonators to be effective at low frequencies requires large depth perforations. The former requires massive volume, while the latter leads to a low ratio of effective exchange area over surface area, resulting in large areas of treatment to be effective. These factors can make this technique impractical outside of large room treatment.

Diaphragm (or membrane) absorbers are a common tool utilized to attenuate standing waves as part of an acoustic treatment. They achieve a high absorption coefficient utilizing a relatively small volume. When paired with a porous acoustic core they present a wider, more linear absorption coefficient, making them a preferred option.

Diaphragm absorbers work on a simple principal: They are velocity transducers, transforming incident

pressure into particle velocity that is then easily dissipated through porous absorption typically placed in the enclosed volume behind the membrane.

Traditional diaphragm absorbers employ a solid material, located over a sealed volume containing an absorptive acoustic core. Incident pressure causes the membrane to move, generating work and movement—displacing air in the interior of the enclosed volume and conveying velocity to the air particles—that are then "forced" to interact with the porous acoustic core.

Porous materials react efficiently to air particle velocity—the air particles travel a convoluted path through the inner matrix of the porous material, dissipating energy through viscothermal losses.

Diaphragm absorbers also have their limitations—seeing as the performance range has a narrower bandwidth at lower frequencies—it is important to build the absorber to resonate at precisely the desired frequency. This is not easily accomplished.

Diaphragm absorbers are essentially mass-spring systems. The peak absorption range of a diaphragm absorber is determined by multiple factors. Mainly the stiffness and mass of the membrane, and also the stiffness of the air inside the enclosed volume.

While the resonant frequency of the cavity and core can be precisely determined,the intrinsic properties of the membrane—the Young modulus and the density—vary greatly in common membrane materials, which can be problematic for precise calibration.

In simple terms, the resonant frequency is determined by the following criteria:

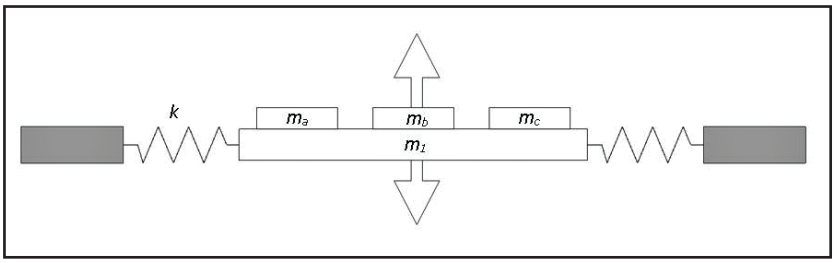


Figure 1: The diaphragm schematics show Total mass = $m_1 + m_a + \ldots + m_i$

Enclosed volume characteristics

- Overall dimensions and volume
- Quantity and absorption coefficient of acoustic core
- Stiffness of enclosed air—determined by volume and depth of the enclosed cavity

Membrane characteristics

- Area—As the area of the membrane increases, the weight also increases, contributing toward lowering the resonant frequency.
- Mass—Determined by the density of the component material and thickness—as the weight increases, the resonant frequency decreases.
- Stiffness—Determined by the Young modulus of the membrane. As the stiffness decreases, the resonant frequency decreases.

The overall absorption coefficient is determined by the amount of energy lost to the porous acoustic core and through flexural losses in the membrane

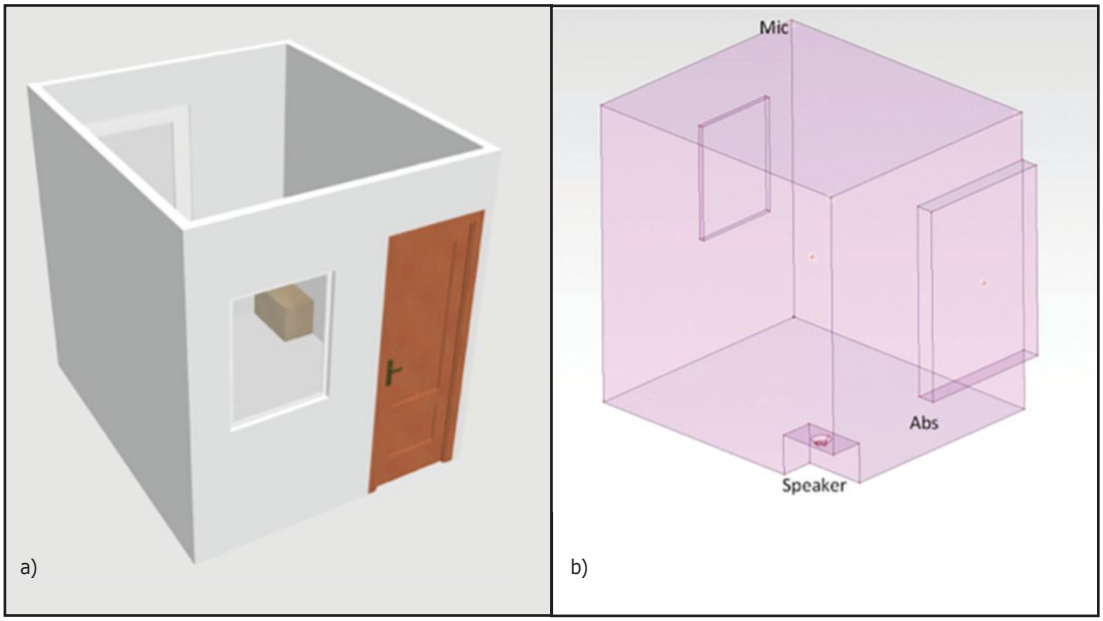


Figure 2: A small room with easily determined room modes was used for testing (a). A speaker was placed in one corner and a microphone was placed in the opposite corner (b).

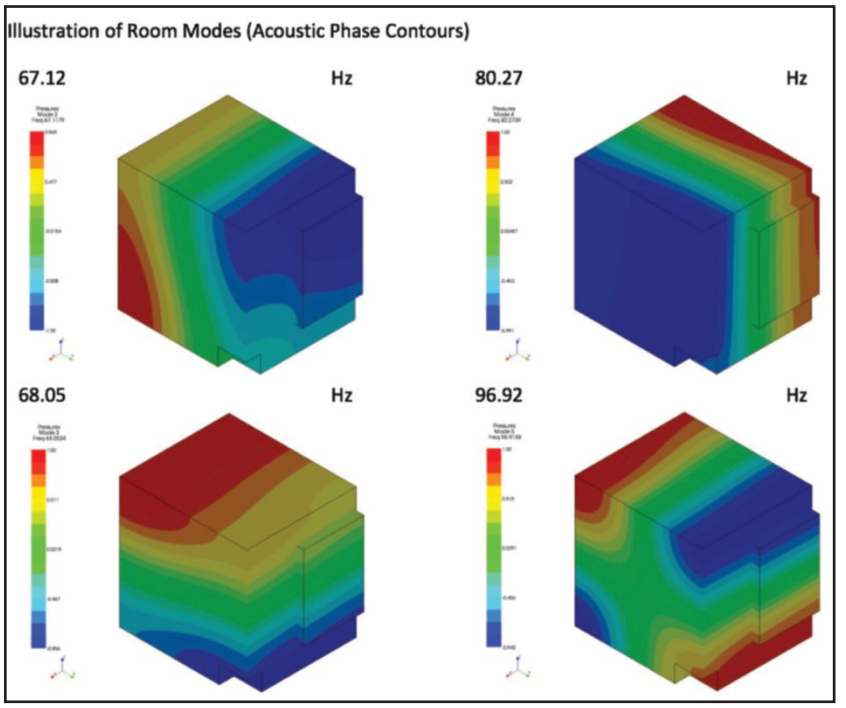


Figure 3: This shows the acoustic phase contours of the room modes.

material. The larger the area and the lower the stiffness of the membrane, a higher quantity of air is displaced in the internal cavity, for a higher absorption coefficient.

This leads us to the main issue—larger membrane areas and lower stiffness decrease the resonant frequency, but this has a limit. To lower the resonant frequency, the membrane weight can be increased by using a thicker membrane or denser material, but these factors also influence the stiffness and make the device heavier and more difficult to activate.

Creating an Efficient, High-Performance Diaphragmatic Absorber

So, how do we create a high-performance diaphragmatic absorber that is efficient at low frequencies? This question brought about a new membrane design—the weighted diaphragm. It employs the same technique as a passive bass radiator—this membrane is composed by a mass suspended on an elastic membrane. This method guarantees range of movement (activation) and true pistonic movement, without forcing a compromise on mass (see **Figure 1**).

Separating the mass from the elastic component also allows for accurate prediction of the absorber's performance, as the mass and the stiffness are separately controlled, making the system behave as a predictable mass-spring system.

A further advantage is that the mass of the membrane can be easily modified, allowing to calibrate the resonant frequency of the system through altering the mass of the membrane without much effect on the stiffness. This results in a tunable low-frequency absorber that presents a high absorption coefficient using a relatively small membrane.

Measurements

To validate the theory, empirical testing was carried out using a small room with easily determined room modes. The room was excited at the corner, and a microphone was placed in the opposite corner (see **Figure 2**). The absorber panel was placed on the edge corresponding to maximum pressure of the axial and tangential mode at 68 Hz. The resonant frequency of the bass absorber itself was tuned to 62 Hz.

Room Conditions Used for Measurements

The room modes are shown at 67.12 Hz, 68.05 Hz, 80.27 Hz, and 96.92 Hz (see **Figure 3**). A small bass absorber—"Corner Bass Trap"—was used. **Figure 4** shows the performance results for the

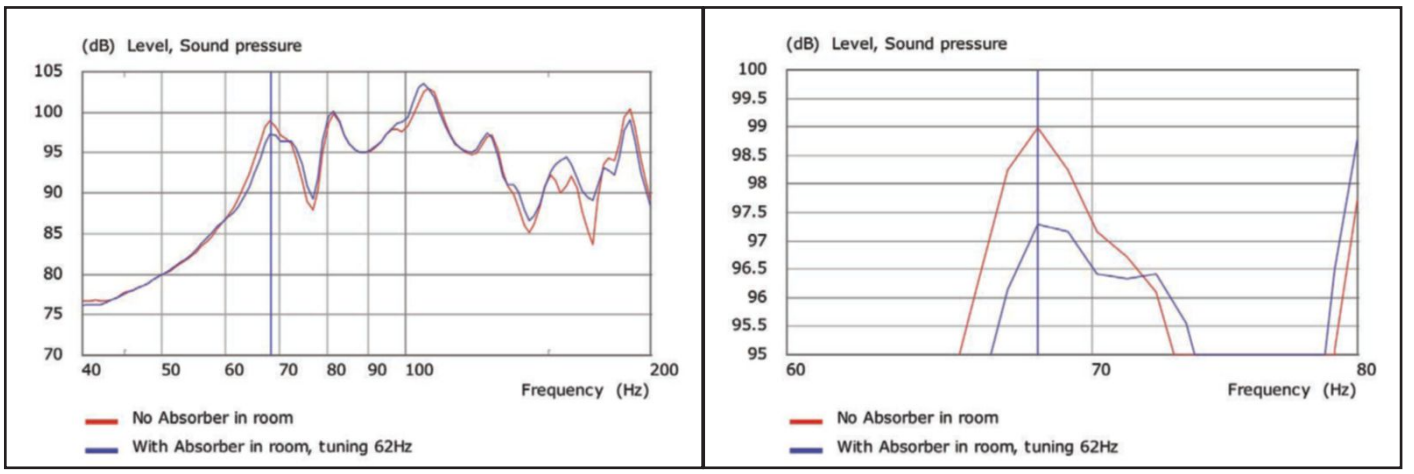


Figure 4: These are the performance results for the single unit in the room.

single unit in the room. Although the absorber is not perfectly tuned to the modal frequency of the room, it still offers significant absorption. A single 0.35 m^2 unit offers an approximate 2 dB reduction at the resonant frequency of room—more units will obviously offer better performance.

Tuning the Membrane's Resonant Frequency

Measurements of mobility with the Laser Doppler Velocimeter (LDV) show the tuning of the absorber (see **Figure 5**). The added masses are indicated in the legend. The original diaphragm (with no additional mass) is approximately 2 kg.

For this absorber, with only the diaphragm, the resonant frequency was 62 Hz. With 0.135 kg mass added to the diaphragm, the resonant frequency was 56 Hz. With 0.5 kg of mass added to the diaphragm, the resonant frequency was 48 Hz.

Mobility decreases as frequency decreases away from the original tuning frequency, possibly due to the panel becoming heavier and thus less mobile or because the excitation from the room mode is less effective—most likely it is a combination of these two factors. This concludes that there are limitations to

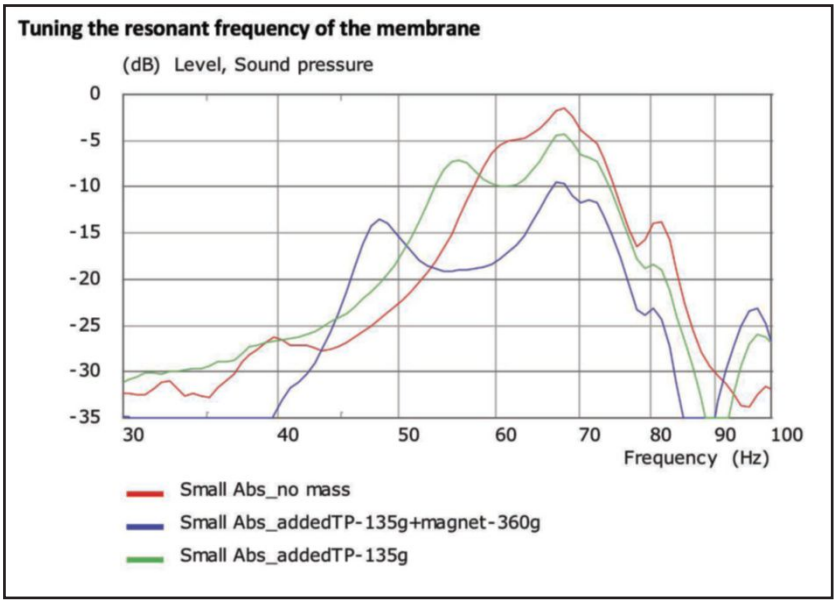


Figure 5: Measurements of mobility with the Laser Doppler Velocimeter (LDV) show the tuning of the absorber.

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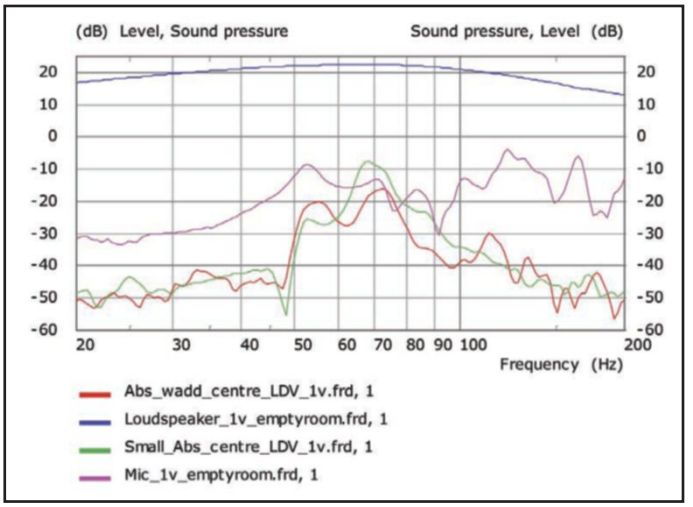


Figure 6: The small absorber is more sensitive to activation from the room modes because it is lighter.

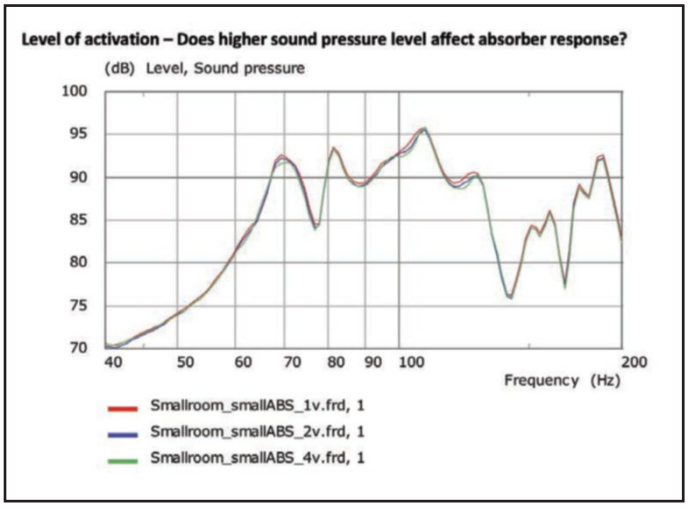


Figure 7: Exciting the room with louder signals appears to activate the absorber so that the Q of the mode is reduced further.

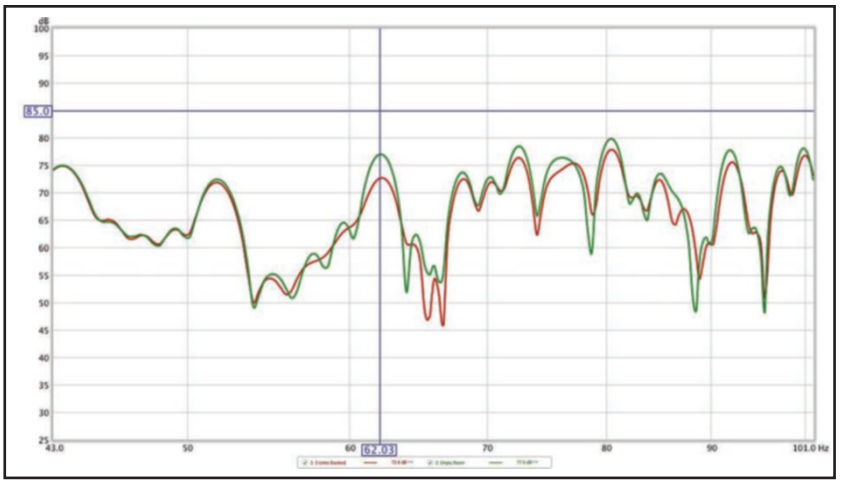


Figure 8: The green line represents the initial conditions. In red, is the room response with three units in place. A significant reduction is visible at around 62 Hz—a total reduction of 4.3 dB. An important phenomenon to note is the range of effective absorption—the diaphragm is tuned to 62 Hz and yet presents effective absorption from 55 Hz up to the 100 Hz range.

the tunable frequency range of the membrane, which is to be expected.

Diaphragm Movement

How much does the diaphragm move? Our laser measurements show the mobility—or range of movement—of the diaphragm, compared to the speaker driving the room. They compare the diaphragm's activation when exposed to the sound field.

The diaphragm's activation is obtained from the acoustic sound field, which means that the room's modal response is superimposed over the measured responses (i.e., it is the modal excitation of the room that makes the diaphragm move). Absorbers that move considerably with the pressure wave result in good absorbers, which means that accurate tuning of an absorber to modal frequency results in optimal absorption.

Activations at the level of the loudspeaker would

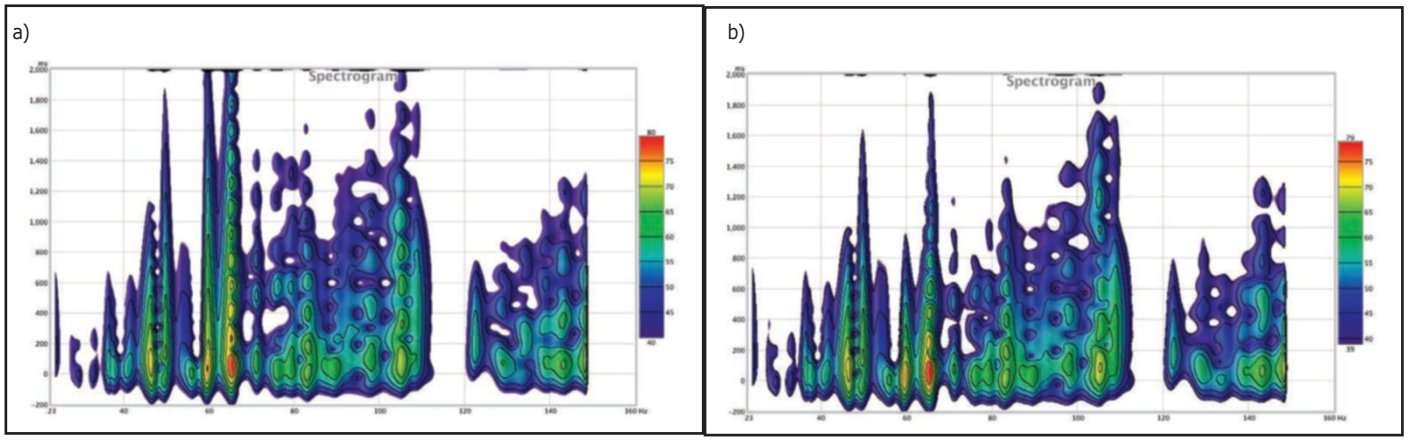


Figure 9: The reverberation time (RT) for an empty room (a) and the RT with three bass trap units (b). We can see a significant difference at 60 Hz—the RT drops from over 2.4 seconds to just under 1 second. There is significant reduction up to 140 Hz. The overall peak energy level also decreases over the frequency range.

result in very efficient levels of absorption. Absorber membranes move considerably less than loudspeaker and that's why they need to have much larger areas to be effective. **Figure 6** shows that Mic_1v_emptyroom is the acoustic room response. The small absorber is more sensitive to activation from the room modes because it is lighter.

Activation Level

Does higher sound pressure level affect absorber response? **Figure 7** shows the acoustic measurements. Exciting the room with louder signals appears to activate the absorber so that the Q of the mode is reduced further. Note that we cannot totally discount the room effects as our measurement room is not perfectly rigid at low frequencies (i.e., the room becomes more absorbent as we drive it harder).

Performance for Three Units in a Medium-Sized Resonant Chamber

Further empirical testing was carried out using a medium-sized reverberation chamber (85 m^3) and three "Corner Bass Trap" units (see **Figure 8**). The same room setup was used, with the speaker placed in the opposite corner to the treatment, and the microphone placed in the opposite corner, in the resulting high-pressure zone. The three units were tuned to one of the modal frequencies in the room—62 Hz. **Figure 9** displays the alteration in decay time of the low-frequency range in a spectrogram.

Large Bass Absorber—"Sub Trap"

So far in this study all theoretical and empirical results have been shown for a small absorber unit—the "Corner Bass Trap." All factors indicate that this style of diaphragm construction can be configured to be effective for even lower frequency. While the effective absorption range of the "Corner Bass Trap" is designed from 60 Hz to 80 Hz, we propose a configuration with effective absorption from 40 Hz to 60 Hz. This absorber—the "Sub Trap"—is divided into three independent closed volumes, in a prismatic configuration that exposes each diaphragm to the maximum pressure zones of each room mode. Further tests are being undertaken to completely characterize the behavior of these larger scale diaphragms.

Understanding the Behavior and Performance of Diaphragm Absorbers in a Room

A parametric model has been created where performance of the small, single-membrane absorber can be assessed (see **Figure 10**). All studies in this section use the small form factor working absorber that was identified in the measurements.

Results for simulations with a varying number of absorbers have been carried out. Impedances of room walls have been set at 5 × Rho × c to simulate the natural losses of the real room. The first problematic mode in this room is at around 68 Hz (see **Figure 11**). A single absorber gives a reduction of 1.66 dB at the modal frequency, which is similar to what was measured.

The location of the device is important. Stacking them up vertically (and entering a zone less involved in the mode shape to be absorbed) resulted in lower returns, which makes sense. Placing them side by side along the wall gives better performance. To illustrate, **Figure 12** shows the results when we consider three absorbers, stacked vertically vs. side by side.

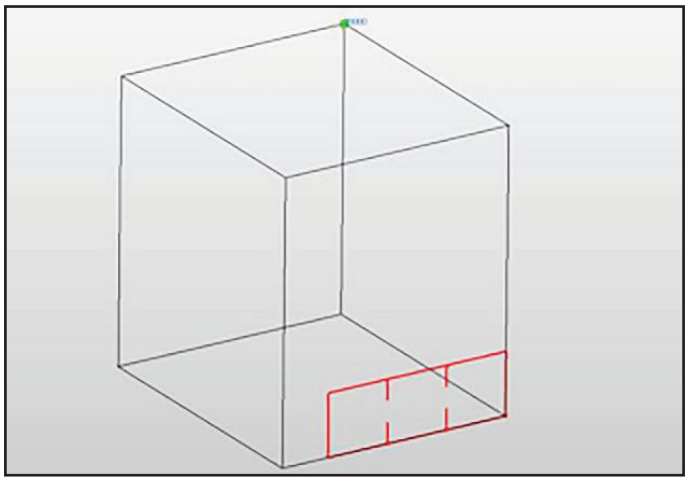


Figure 10: A parametric model has been created where performance of the small, single-membrane absorber can be assessed.

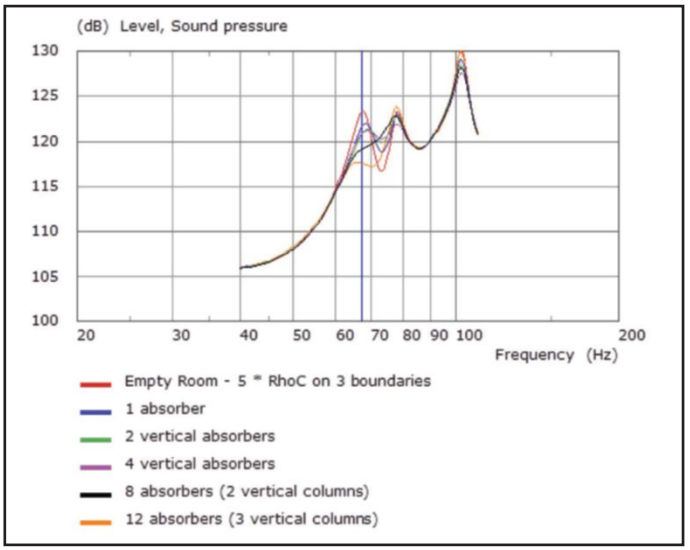


Figure 11: A single absorber gives a reduction of 1.66 dB at the modal frequency, which is similar to what was measured.

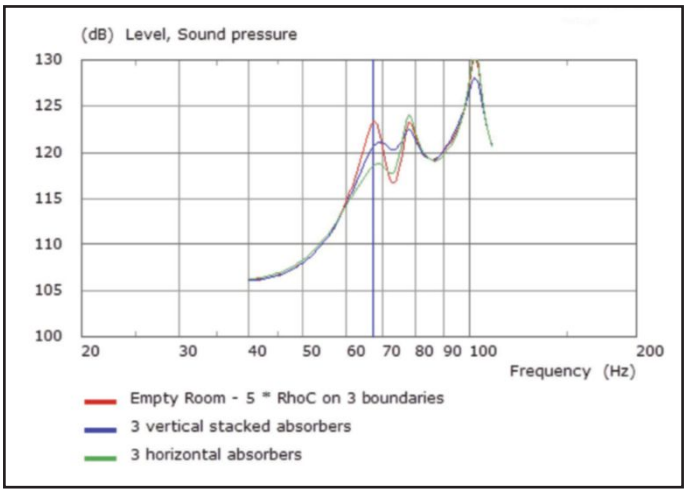


Figure 12: Optimal positioning of the device does improve its performance.

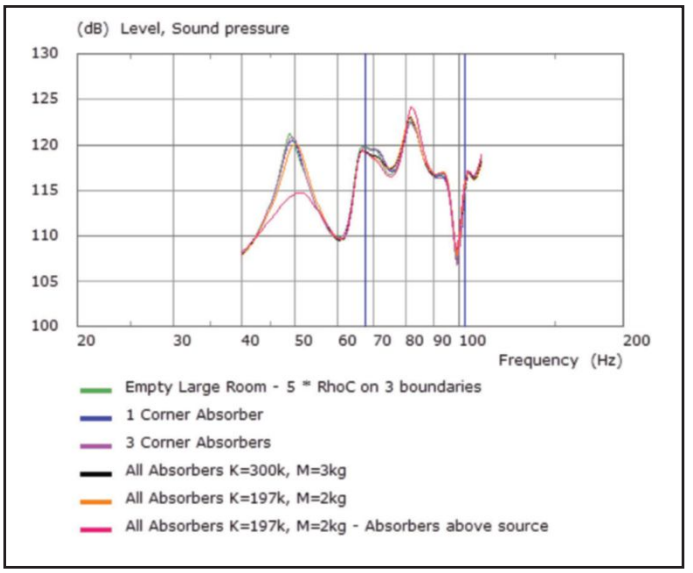


Figure 13: Simulation of room response with varying quantities, tuning and location.

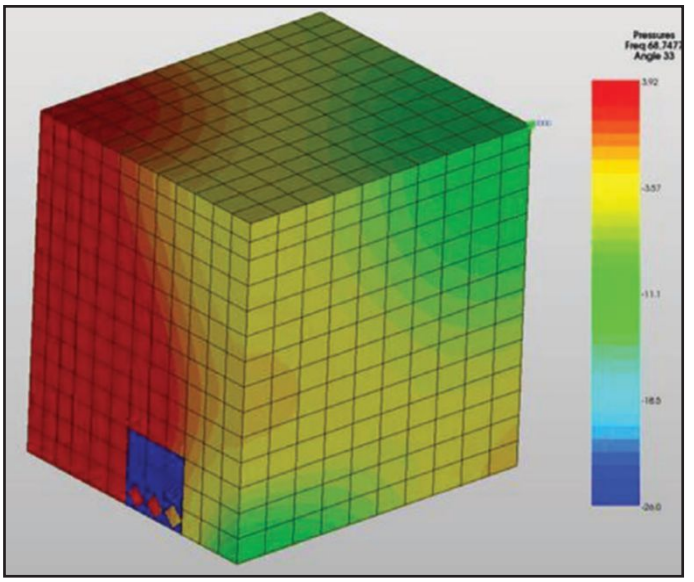


Figure 14: Pressure map of the test room. Placing bass traps in high pressure zones is critical.

Decibel Loss at Modal Frequencies	Room Dimensions		
	Small	Medium	Large
X(m)	2.15	4.3	6.45
Y(m)	2.56	5.12	7.68
Z(m)	2.5	5	7.5
V(m^3)	14	110	272

Table 1: The room dimensions for the first example.

As can be seen (and as we saw in practice during measurements), optimal positioning of the device does improve its performance. The green curves displayed in Figure 12 shows that with only three absorbers lined horizontally on the floor of the room, a decent room treatment can be obtained. Proof of this should be sought with measurements in a real room with various absorbers.

Further simulations were carried out using the same setup, over a range of room sizes and number of units, with the objective to determine a correlation between location, quantity, and effect as shown in **Figure 13**.

Quantifying Absorption Performance

Here we quantify the effectiveness of the small absorber in removing modal energy according to various simulation scenarios. Further simulations were run to ascertain the amount of absorption in terms of decibel drop at the modal frequency when:

- The same percentage of the room's surface area is occupied in three different sized rooms (the tuning of the absorbers was changed to follow the change in modal frequency at mode 1). This corresponds to creating simulations for three different rooms with increasing volume and covering an increasing amount of wall area with absorbers as the room size increases. However, to provide a relative comparison between the three rooms, the percentage area covered by the absorbers remains the same.
- The number of absorbers is increased in the same (small) room. In this study, a single room size is used but three difference simulations are run where the number of absorbers in the room increases.

Absorbers are always placed at the maximum pressure zones of the modes they are designed to control (see **Figure 14**). We have two examples of parametric rooms with absorbers.

In Example 1, the three different room dimensions are shown in **Table 1**. And the number of absorber units are one, four, and nine, respectively. **Figure 15** shows the results. As expected, for the same proportion of absorption area, the same loss at the modal frequency is obtained. This shows that, as the room becomes larger (and the surface area increases) more absorption units are required to achieve the same degree of modal control. In small volume rooms (less than 20 m^3) a single unit is capable of providing demonstrable (but not sufficient!) absorption. For large volume rooms (greater than 300 m^3), a total of nine units are required to achieve around 2 dB reduction at the modal frequency.

In Example 2, the room size is kept constant, but the number of absorber units in the room is increased to assess the amount of modal reduction. The amount of absorption is provided as percentage of wall that has been covered by absorber units. The number of units required may then be determined from the room dimensions and absorber dimensions. The simulations shown in **Figure 16** correspond to a square of one, four, and nine elements. The square grows from the single absorption position outward, to try to minimize position dependence that we have already identified as a complex interaction affected by the room

aspect ratio/shape, source number and position, and so forth.

It can be seen that the amount of loss at modal frequency grows linearly with absorption area covered by absorption units, described by the following expression:

$$Loss_{dB} = 0.5\frac{Absorption\ Area\ of\ Absorber}{Surface\ Area\ of\ Room}\times 100$$

Using this expression, the number of required absorber units, at any of its tuning frequencies, may be prescribed for any room. A very simple model can be designed to calculate the amount of modal reduction (Loss) possible for a N number of absorbers in a Sw × Sd × Sh room:

$$Loss_{dB} = 50\times\frac{N\times width_{abs}\times height_{abs}}{2\times(width_r\times height_r + depth_r\times height_r + width_r\times depth_r)}$$

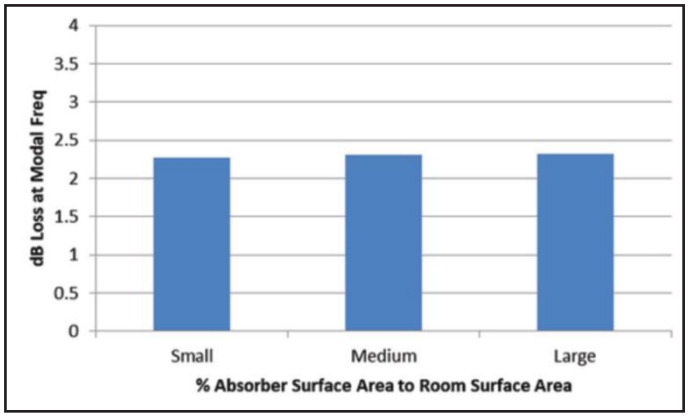


Figure 15: The absorber surface area to room surface area is compared to the decibel loss at modal frequency.

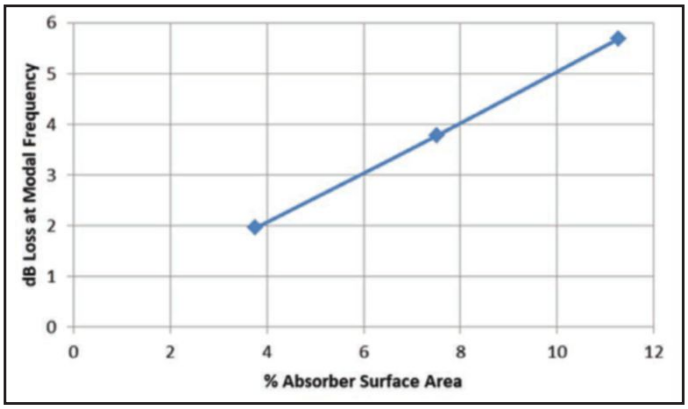


Figure 16: The square grows from the single absorption position outward, to try to minimize position dependence that we have already identified as a complex interaction affected by the room aspect ratio/shape, source number, position, and so forth.





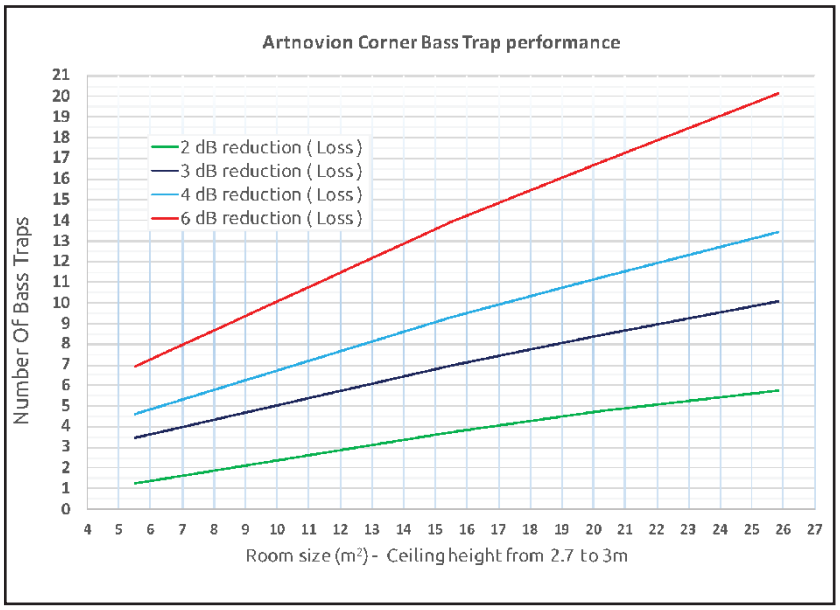


Figure 17: We can conclude a direct ratio between the number of absorber units, volume, and the decibel reduction in each room.



Photo 2: This is the new Bass Trap tuning system and membrane technology.

Note this expression holds for the size and tuning of the absorption measured and simulated. Other configurations might not present the same trend exactly.

Conclusion

Standard representation of acoustic absorber products is based on the absorption coefficient, which is directly linked to reverberation time calculation. While reverberation time is a useful scale to estimate frequency response, especially in the mid-high frequency range, it gives no indication to the modal behavior of a space. A more practical—and simple—approach is to present the attenuation value for a certain quantity of units in a certain sized room. A simple room measurement can then determine the required quantity of absorbers.

From the earlier examples, we can conclude a direct ratio between the number of absorber units, volume, and the decibel reduction in each room (see **Figure 17**). To use the chart, simply cross reference the room size (in m^2) with the desired decibel reduction to get the necessary quantity of Bass Trap panels (see **Photo 2**).

As shown in both the simulations and the measurements, the positioning of absorber units relative to the mode shapes is extremely important. All simulations have placed absorber units in the maximum modal pressure zones. For more information, visit www.artnovion.com.

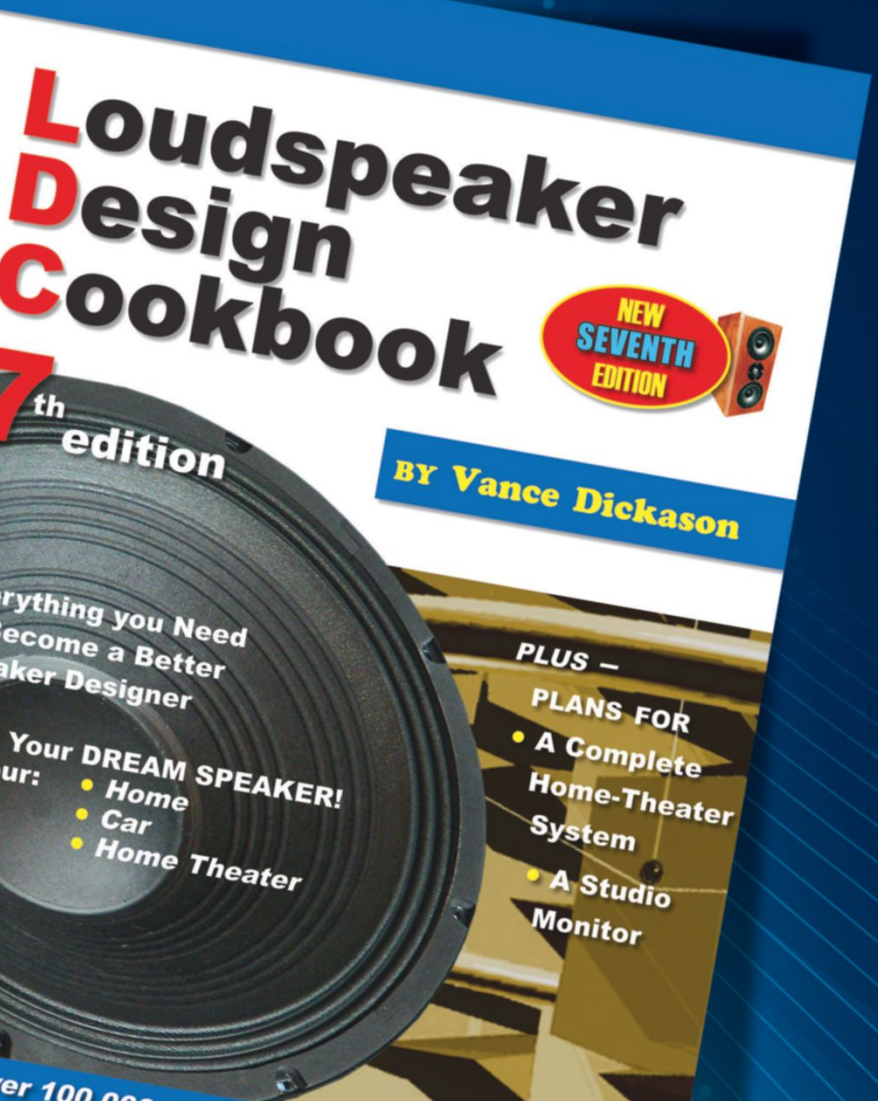
About the Authors

Dr. Bruno Fazenda and Kelvin Griffiths are researchers working together with Artnovion to develop innovative ways of facing acoustic problems. Fazenda has been working in acoustics since 2000 and is a specialist in room acoustics and psychoacoustic, in particular, the perception of modal problems in rooms. Griffiths is a specialist in numerical design and optimization, having vast experience in the design of loudspeakers and related products.

Working directly for Artnovion on this project were Jorge Castro and Nathaniel Bailey. Castro is Artnovion's CEO and founder. He attended Salford University to complete a PG/Msc in Audio Acoustics. Nathaniel Bailey is Artnovion's R&D engineer and product development leader. All tests were developed at Electroacoustic Design, Ltd., in Porthcawl, South Wales.

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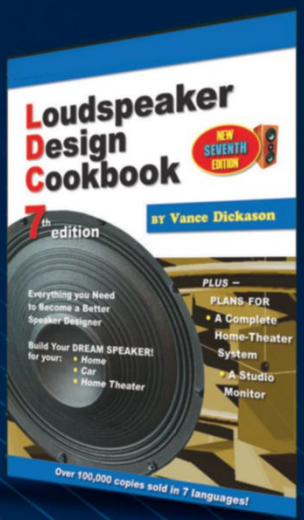
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Current Feedback: Fake News or the Real Deal? (Part 2)

A Detailed Examination of Current Feedback Amplifiers

In Part 1 of this article series, we designed, analyzed, simulated, built, and tested simple versions of voltage and current feedback amplifiers and demonstrated some differences in their characteristics. In Part 2, we will take a detailed look at some commercially available current feedback amplifiers and voltage feedback amplifiers.

By
Christopher Paul
(United States)

In the first part of this article series, we addressed the question asked by the article's title. We subjected several circuits to bench tests, simulations, and derivations of equations that described their properties. We also examined the history of the term "current feedback." From all of this, it seems that we might have been asking the wrong question: Not does it exist, but how do we define it? Consider the following possibilities:

1. We can sense either the voltage across the output load, or the current through it. That current can produce a voltage across a small ground-referenced resistor in series with it. Sensing that voltage is referred to as series derived feedback (see Part 1, Figure 7) and sometimes simply as current feedback.
2. Figure 7 from the first part of the article series also addresses shunt applied feedback. In that arrangement an amplifier's non-inverting input is grounded, and its typically high open loop gain drives the voltage difference between the inputs to very small levels. Since the inverting input is now at "virtual ground," the feedback signal looks like a current flowing from the output through the feedback network to ground. This is another claim for current feedback.
3. Using the tools created by R.A. Middlebrook, we discovered in Part 1 that regardless of amplifier input stage structure, the relationship between the impedances of the amplifier's feedback input Z_{FB} and that seen looking from that input into the feedback network (R_{EQ}) determines whether voltage or current feedback predominates.
4. Consider a single transistor input stage whose feedback point is its emitter. Its output is its collector current, and its emitter is biased by a DC current source, which by definition cannot source AC signal current. Its base current is much smaller than its collector's. Therefore, the AC current signal must pass through the feedback network and derive from the amplifier's output. (To see this clearly, replace these elements with their Thevenin equivalent: an attenuated voltage source in series with a single resistor.) Here is yet another claim for current feedback.

It is foolish at this late stage to attempt to restrict the meaning of current feedback to one single thing. The best we can do is remain aware of the context in which the term is used. Let's look at some designs that qualify for definition 4.

A Commercially Available Op-Amp Using Current Feedback

There is at least one op-amp that uses the simple e(mitter)-fed stage introduced in Part 1. In fact, it uses two of them. The part is Analog Device's SSM2019.

This device is a differential to single-ended amplifier (see **Figure 1**). The signal input is applied between IN+ and IN-. A single resistor R_g is connected between the points RG_1 and RG_2 to set gain. The circuit is bilaterally symmetric, and so if the input signal is balanced, the voltage at the center of resistor R_g is at AC ground. In this case, you can think of the 1+ and 1- stages as each operating with an $R_g/2$ resistor to AC ground. Stage 1 drives Stage 2. Stage 2 drives Stage 3. Together, they form a "folded cascode." Through sets of 5 kΩ resistors, Stage 3's ×1 buffered outputs supply differential feedback to Stage 1 and inputs to the Stage 4 differential to a single-ended converter.

Typical distortion at and below gains of 100 is better than 0.01%, and noise is 1nV/√Hz above 100 Hz. To determine the Stage 1 emitter impedances, we can plug this noise level into an equation for thermal noise, R (ohms) = (volts/√Hz)^2/(1.62 × 10^{-20} Watt-sec). We must consider that the two input transistors contribute to R in an RMS fashion, and so we can calculate that the resistance at the emitter of each is about 44 Ω. And sure enough, the datasheet graphs are consistent with bandwidth remaining constant as $R_g/2$ rises above that value. At values of 44 Ω, the impedances of the emitters, Z_{FB}, and those seen by the emitters, R_{eq}, are approximately equal. We discussed this type of behavior in Part 1. Let's call it the "constant bandwidth effect."

There is another important effect, independent of the relationship between Z_{FB} and R_{eq}, that arises from this circuit topology. The current available to charge and discharge the capacitor in the signal paths of all op-amps limits their slew rates. Some of that current can come from the DC current sources supplying bias for Stage 1. Portions of these currents can be steered through Stages 1+ and 1- by the voltages at IN+ and IN-. This is the only source of charging current for capacitors in standard voltage feedback amplifier (VFA) op-amps, where feedback is applied to a transistor base rather than the emitter. But when feedback is applied to an emitter through the feedback network, it flows through the entire stage, boosting the charging current as needed. This current only flows when needed and so does not contribute to constant power dissipation. We can call this effect "slew rate enhancement."

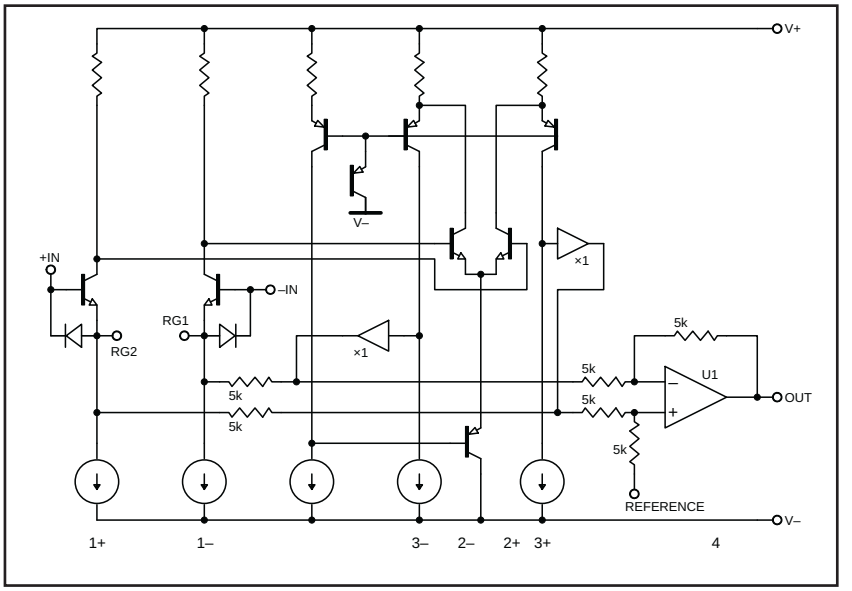


Figure 1: This is the schematic of the SSM2019. Each of its stages is numbered for discussion. (Original image courtesy of Analog Devices)

The CFA

Although we can argue (in accordance with current feedback definitions 3 and 4) that the SSM2019 is a current feedback amplifier, it is not what the industry has come to call a Current Feedback Amplifier (CFA). Such a device has the specific basic structure shown in **Figure 2**.

This design contains two current mirrors, Q5/Q7 and Q6/Q8, and four separate Faux-Darlingtons: Q1/Q3, Q2/Q4, Q9/Q11, and Q10/Q12. (I regret that I have found no better name for this configuration and shall henceforth refer to them as Darlingtons.)

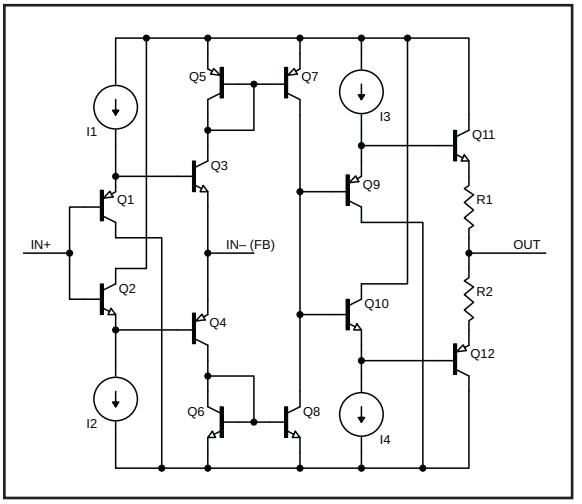


Figure 2: Here is the basic structure of what the industry refers to as a Current Feedback Amplifier (CFA).

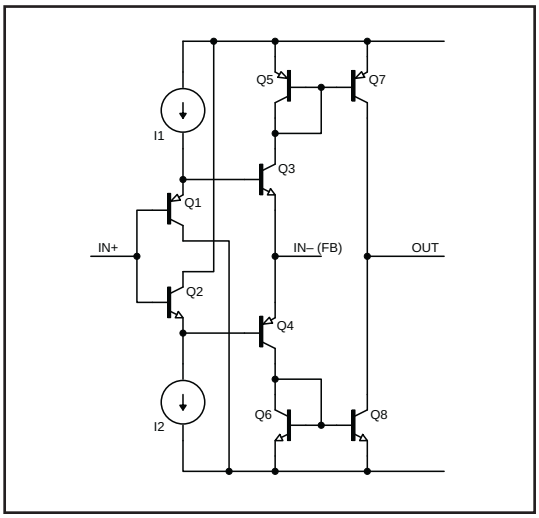


Figure 3: The assumed structure of Texas Instruments' "ideal transistor" Operational Transconductance Amplifier, the OPA861.

Each Darlington contains one NPN and one PNP transistor, allowing its input and output to be at the same DC voltage, simplifying the overall design. The input transistor of each pair receives an emitter bias current from a DC current source as shown. The matching capability inherent to ICs enables the bias current of the pair's output transistor to be set as desired (this topology would not work well with discrete transistors). The signal current (I_{FB}) flowing into IN- is routed through Q3 and Q4. Their outputs are added together by the current mirrors, whose outputs have a very high resistance Z. This resistance is only slightly lowered by the two output Darlingtons. Typically, this is the point at which capacitance is controlled to ensure stability when the feedback loop is closed. The transfer function of this device is Z = OUT/I_{FB}. Each of the two input Darlingtons functions like the simple e-fed input stage described in Part 1. Since the same DC current I_{DC} runs through these seriesed devices, the impedance Z_{FB} looking into IN- at room temperature is approximately (26 mV/I_{DC})/2. This design exhibits both the constant bandwidth and the slew rate enhancement effects already discussed.

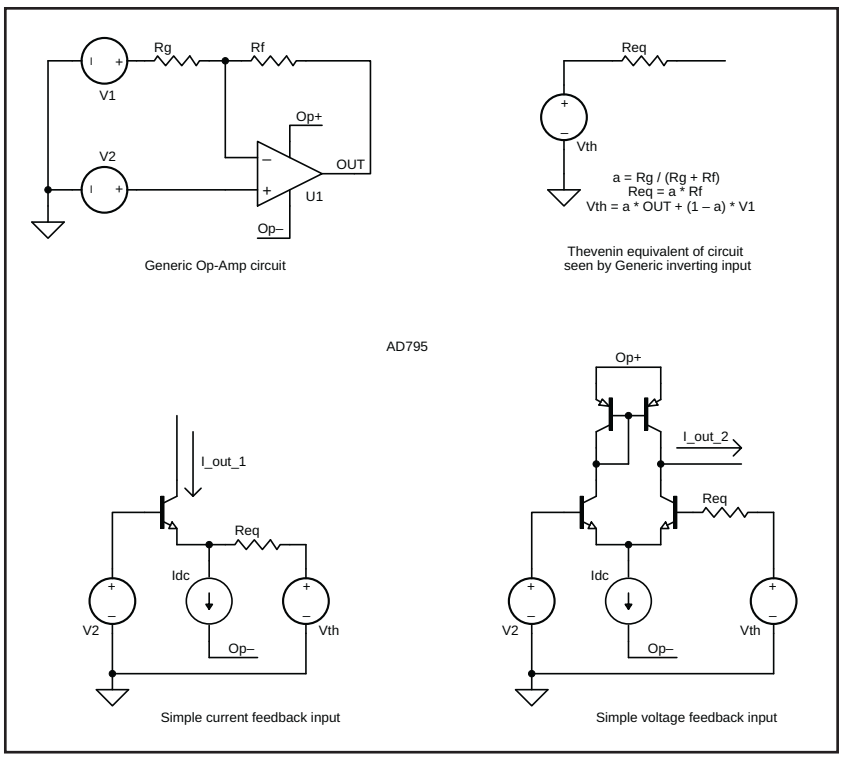


Figure 4: This is a generic op-amp with simplified current and voltage feedback type inputs.

A CFA Enhancement

The constant bandwidth effect is dependent on R_{eq} being larger than Z_{FB}. To support this effect, we make Z_{FB} as small as possible so that even at high gains, when the feedback shunt resistor that contributes to R_{eq} is very small, this effect is still present. A brute force method is to increase the bias current of the input stage, which of course increases dissipation. But another approach is possible—enclose it in a unity-gain feedback loop. This is what Texas Instruments (TI) did with the OPA683, the OPA684, and the OPA691. It would be nice to be able to describe this circuitry, but TI's datasheets do not provide any schematics. Consider, however, that the OPA683 affords bandwidths of 100 MHz or more and a slew rate of 450V/µS at gains of +5 or less at a supply current of under 1 mA!

Solomon's Partial CFA

To paraphrase from King Solomon's teachings: "That is my CFA, your majesty," shouted Rebecca. "No, it's mine," cried Miriam as she scowled murderously at Rebecca. The wise king considered for a moment. "Since you cannot agree, you both shall have a part of it!" And he proceeded to slice the microchip into two pieces with an extremely precise laser.

History does not record which portion went to whom, let alone how the packaging engineers managed to make useful surface mount components out of them. But one of the pieces proved very useful, and it is shown in **Figure 3**.

Clearly, this is the input of the Figure 2 circuit. But it's also an Operational Transconductance Amplifier (OTA). It accepts a voltage difference between its two "IN" pins and supplies a bidirectional current from its OUT pin. Most likely, something very much like this has been implemented in TI's OPA861. (I can't be certain about this, but some folks I know with respectable credentials claim this to be so.) Not shown in Figure 3 is an enhancement to the basic design which allows input stage bias current to be set

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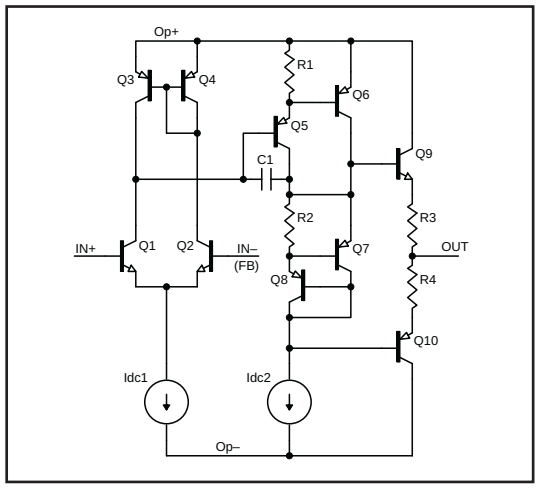


Figure 5: Here is the basic structure of what the industry refers to as a Voltage Feedback Amplifier (VFA).

to trade off quiescent current, gain and bandwidth. This part is marketed as an "ideal transistor," whose IN+ is the "base," IN- the "emitter," and OUT the "collector." Its ideal aspect comes from its zero DC offset "base-emitter" voltage, and the fact that its inputs and outputs support both polarities of voltages and currents. Transconductances from 40 mA/V to more than 100 mA/V can be attained with bandwidths from 50 MHz to 100 MHz. Even if some of the datasheet's applications might not be optimal for audio, it's fun to stretch your analog brain and look at things from the point of view of a device whose relative output and inverting input impedances are the opposite of an "ideal" op-amp's.

Yes Virginia, there is a VFA

Comlinear Corp. was first to offer the CFA commercially back in the 1980s. Before that, there was a topology that didn't need a name because it was pretty much the only game in town. But with the advent of the CFA, it has come to be referred to as a VFA. **Figure 4** helps us see how these names evolved. The generic op-amp, either a VFA or a CFA, works in both inverting and non-inverting configurations. Converting the resistive network into a Thevenin equivalent aids in our understanding of these variants. Simplified examples of both input stages are provided. The output of each is a current. In the VFA, that comes almost exclusively from I_{dc}, which is steered by the difference between voltages V2 and Vth. The output is controlled by this voltage difference, one part of which is fed back from the voltage OUT, hence the term VFA. In the CFA, the DC source cannot be steered; its output passes through the transistor collector save for a small portion that passes through the transistor base. The dynamic (signal) portion of the feedback current, therefore, must come from somewhere other than I_{dc}: OUT, through R_{eq}. Accordingly, this type of op-amp is referred to as a CFA. All CFAs and VFAs are elaborations of these simple circuits, which are distinguished by the type of transistor pin to which feedback is applied.

The basic structure of a complete VFA can be seen in **Figure 5**. DC current Idc1 supplies Q1 and Q2. The voltage difference between IN+ and IN- steers Idc1 between these two transistors. The current mirror consisting of Q3 and Q4 recombines the currents, which supplies the input of the Darlington consisting of Q5 and Q6. C1 provides frequency compensation to ensure stability. The Darlington's output and Idc2 supply current to the voltage reference consisting of Q7, Q8 and R2. This reference, along with R3 and R4, establishes the bias current for the output stage comprising R3, R4, Q9, and Q10.

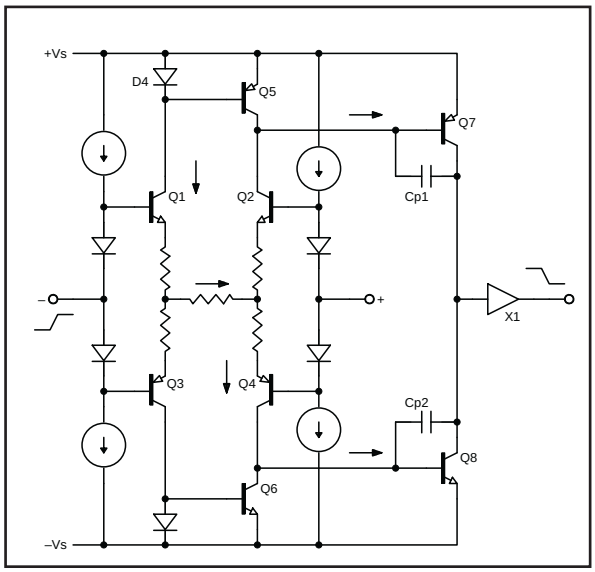


Figure 6: This is the basic schematic of Analog Device's "Quad Core" input stage. (Original image courtesy of Analog Devices)

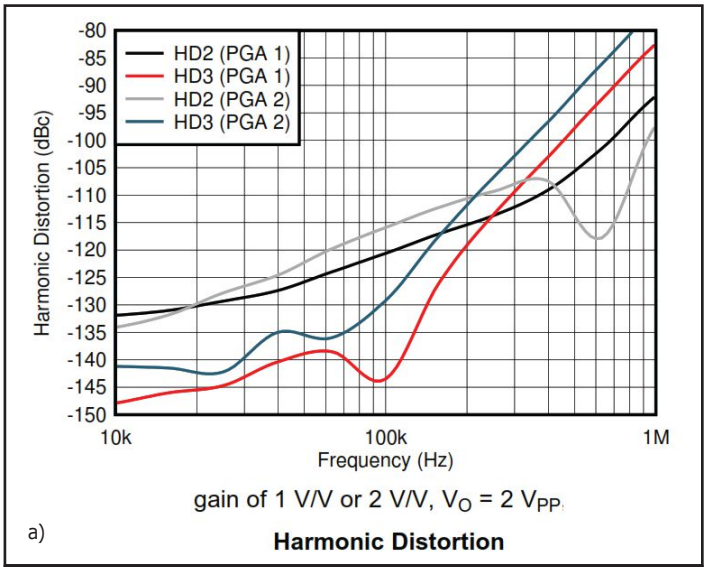
VFA Enhancements

Several techniques have been applied to VFAs to enhance slew rate without increasing quiescent power dissipation. Analog Devices employs what it calls a "Quad Core" design, which can be seen in **Figure 6**. The "central" resistor in the middle of and equidistant to Q1, Q2, Q3, and Q4 is the key to its operation. (The complete bias circuitry is not shown.) When the voltages at the + and – inputs are equal, no current flows through this resistor. But if the voltage difference increases because of inadequate slewing, the input stage becomes increasingly unbalanced. If the voltage at the – pin were to exceed that at the + input current through the components and paths indicated by the arrows would increase. The increase could much more than double the currents supplied by only constant current sources (CCS). These "unbalance" currents would charge the capacitors, increasing slew rate.

One of the op-amps employing this technique is the AD8038. With a supply current of only 1.5 mA per amplifier, it achieves a gain bandwidth product of 350 MHz and a slew rate of 425/µs! Another approach to enhancing slew rates is to sense the difference in input voltages and increase the input stage "DC" bias current as this difference increases. Neither scheme adds to quiescent dissipation. Manufacturers use these and other approaches to achieve similar results.

VFA-CFA Shootout

I won't claim that the devices I've chosen to compare are the very best of their kind. I'm not sure that "bests" exist. Certain parameters are superior in other parts. But I do believe that like Ralph's Pretty Good Grocery Store in Lake Wobegone (a fictional town that served as the setting for stories from longtime radio broadcast *A Prairie*



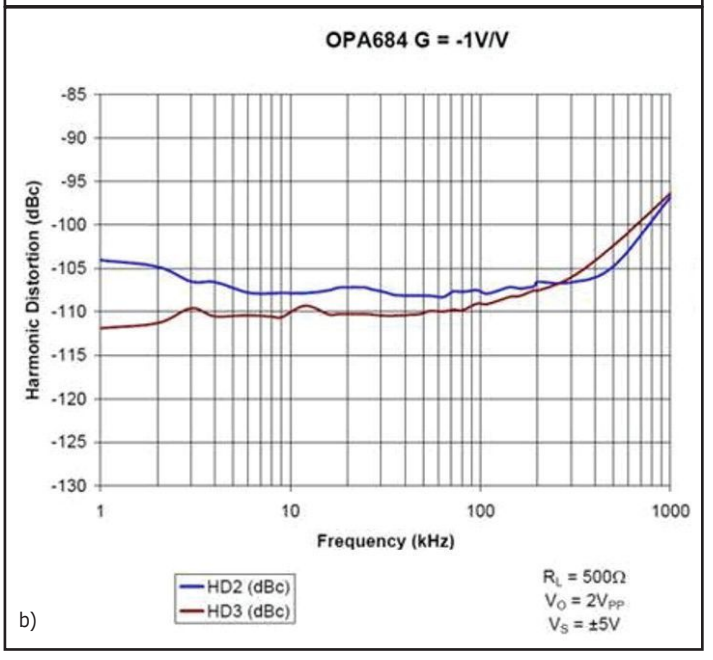


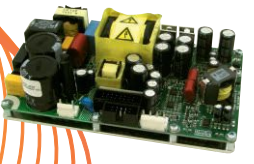
Figure 7a: Harmonic distortion of the VFA OPA837; Figure 7b: Harmonic distortion of the CFA OPA684 (Image courtesy of Texas Instruments)

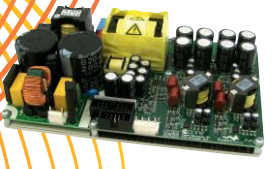
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overview

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Highlights

- High efficiency
- Universal mains operation
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- Low output impedance
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Features

- One or two channel amplifier
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- Advanced over current protection
- External controlled operation
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Applications

- Monitor loudspeakers for recording and mastering studios
- Audiophile power amplifiers for professional and consumer use.
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NC502MP

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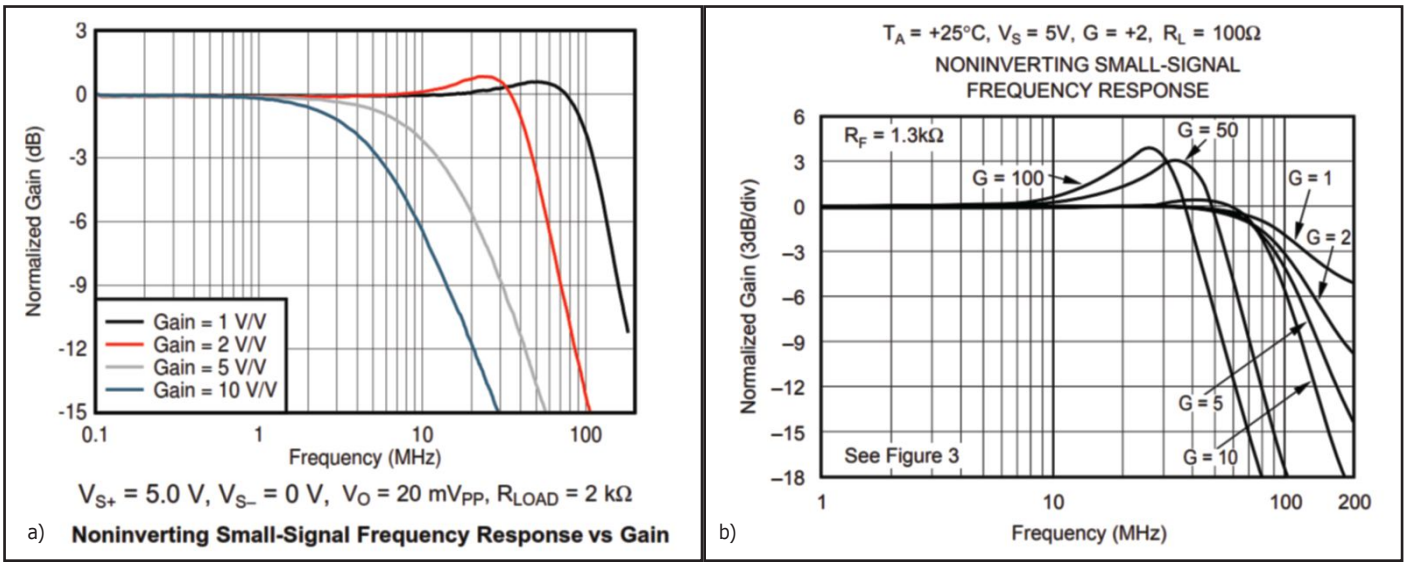


Figure 8a: Non-inverting small-signal frequency response of the VFA OPA837; Figure 8b: Non-inverting small-signal frequency response of the CFA OPA684 (Image courtesy of Texas Instruments)

Home Companion), these parts are pretty good—maybe even pretty darn good.

We are fortunate to have several graphs that display characteristics of interest to audiophiles for the OPA387 VFA and the OPA684 CFA. Unless explicitly stated otherwise, I'll restrict my comments to the audio frequency range. The graphs shown in **Figures 7-12** are courtesy of TI.

Traditionally, one of the first things we look at is distortion performance. With a closed loop gain (CLG) of unity, we see total harmonic distortion (THD) better than -100 dB (0.001%), although this VFA at audio frequencies is perhaps 30 dB better. We expect the relative distortions at higher CLGs to move toward favoring the CFA because of its much slower fall in open loop gain (OLG), as shown in Figure 5 from Part 1, but that data is unavailable. (Note: The CFA distortion graph generated by TI was never updated in the datasheet.) Next, we note the VFA's inverse relationship between bandwidth and CLG in comparison to the CFA's bandwidth change of a mere factor of three as CLG varies by a factor of 100. Moving on, the CLG output impedance and power supply rejection ratio (PSRR) of both units and the CFA's common-mode rejection ratio (CMRR) closely track the shapes of their OLGs. Interestingly, the VFA's CMRR has a much wider bandwidth than its OLG. Regarding audio frequency output impedance, the VFA out-performs the CFA because of its greater loop gain. As for speed, the CFA is about 10 times

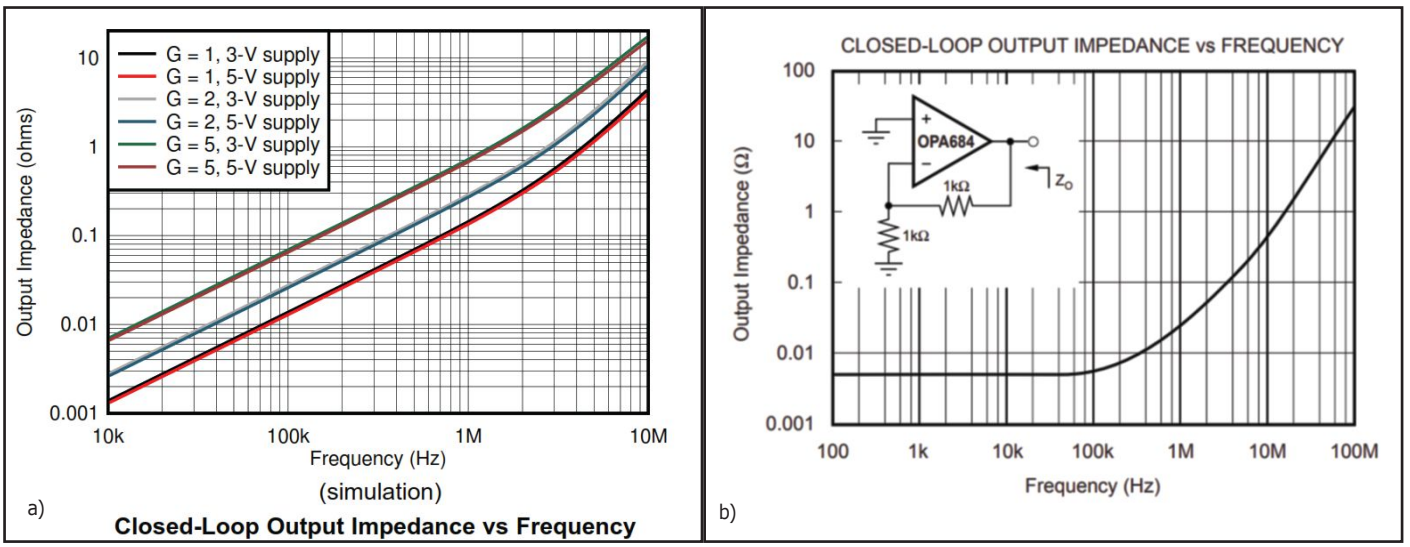


Figure 9a: Closed-loop output impedance of the VFA OPA837; Figure 9b: Closed-loop output impedance of the CFA OPA684 (Image courtesy of Texas Instruments)

faster at a gain of 2 than the VFA (note the difference in time scales.) As with bandwidth, this difference would be even more extreme at higher CLGs. Note also the comment about the need to limit the slew rate of the signal applied to the VFA to preclude slewing-induced distortion. This is of little concern at audio frequencies for this device. Finally, the voltage noise of the two are about equal, with an edge given to the CFA. But VFA current noise is better by a factor of 2 to 4.

Additional Comparisons

This VFA has a maximum offset voltage of 200 µV vs. the CFA's 4.3 mV. When it comes to bias current, the VFA's maximum is 718 nA vs. the CFA's 12 µA at its non-inverting input and 18.5 µA at its inverting one. The CFA's offset current isn't even specified. Its non-inverting single-ended input impedance is typically 50 kΩ || 2 pF, while the VFA's differential impedance is typically 180 kΩ || 0.5 pF. Our VFA draws 865 µA from a supply of up to 5 V, while the CFA

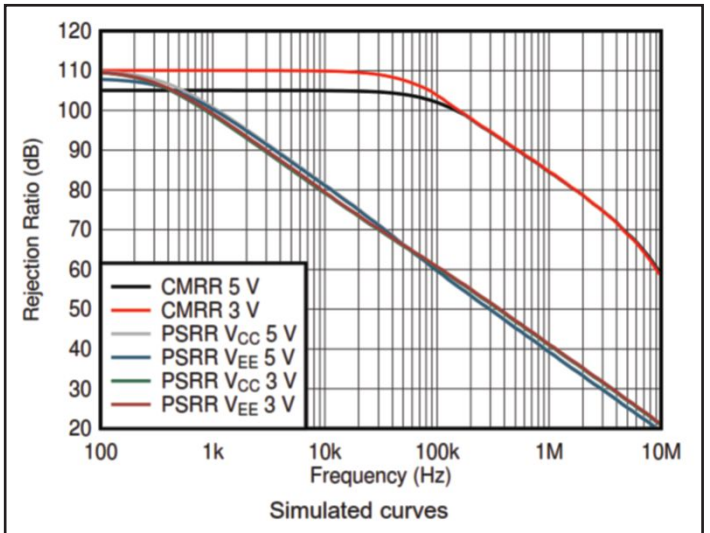


Figure 41. CMRR and PSRR vs Frequency

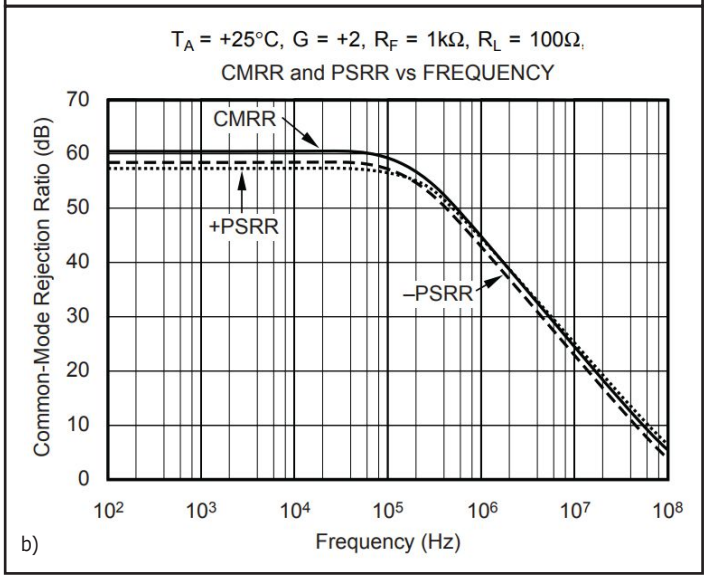


Figure 10a: CMRR and PSRR of the VFA OPA837; Figure 10b: CMRR and PSRR of the CFA OPA684 (Image courtesy of Texas Instruments)

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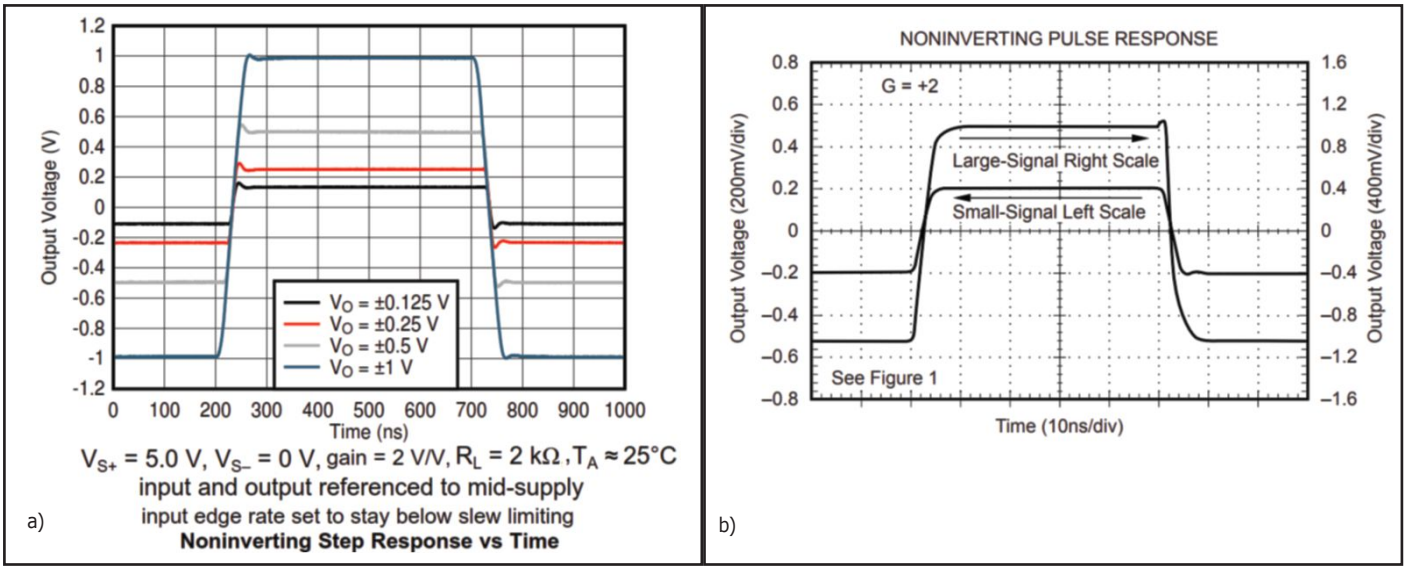


Figure 11a: Non-inverting step response of the VFA OPA837; Figure 11b: Non-inverting step response of the CFA OPA684 (Image courtesy of Texas Instruments)

consumes 1.85 mA from ±6 V rails.

A CFA's bandwidth and slew rate are highly dependent on the value of R_f (see Figure 4). The constraint on R_f's value forces a commensurate restraint on R_g for a given gain, which limits the usefulness of CFAs in inverting modes of operation. There are no such limitations on a VFA.

Summing Up

Current feedback is real. Unfortunately, it has multiple definitions. We must be aware of the context in which the term is used. The industry has taken to calling a specific type of amplifier a Current Feedback Amplifier (see Figure 2), and the reason for this name has been discussed.

In Part 1, problems were noted with the traditional associations of voltage feedback with series applied feedback and current feedback with shunt applied feedback. It is this author's opinion that we may retain series/shunt derived/applied to describe signal routing. But we must avoid associating the signal routing type with exclusively voltage or current feedback. This is because "Applied feedback" is "applied" to both CFA and VFA amplifiers.

The feedback type accepted by an amplifier can be distinguished by the source of the output current of its input stage. If none of this comes from the amplifier output, voltage feedback is the obvious choice. If all or much of it derives from the amplifier output, then we have current feedback.

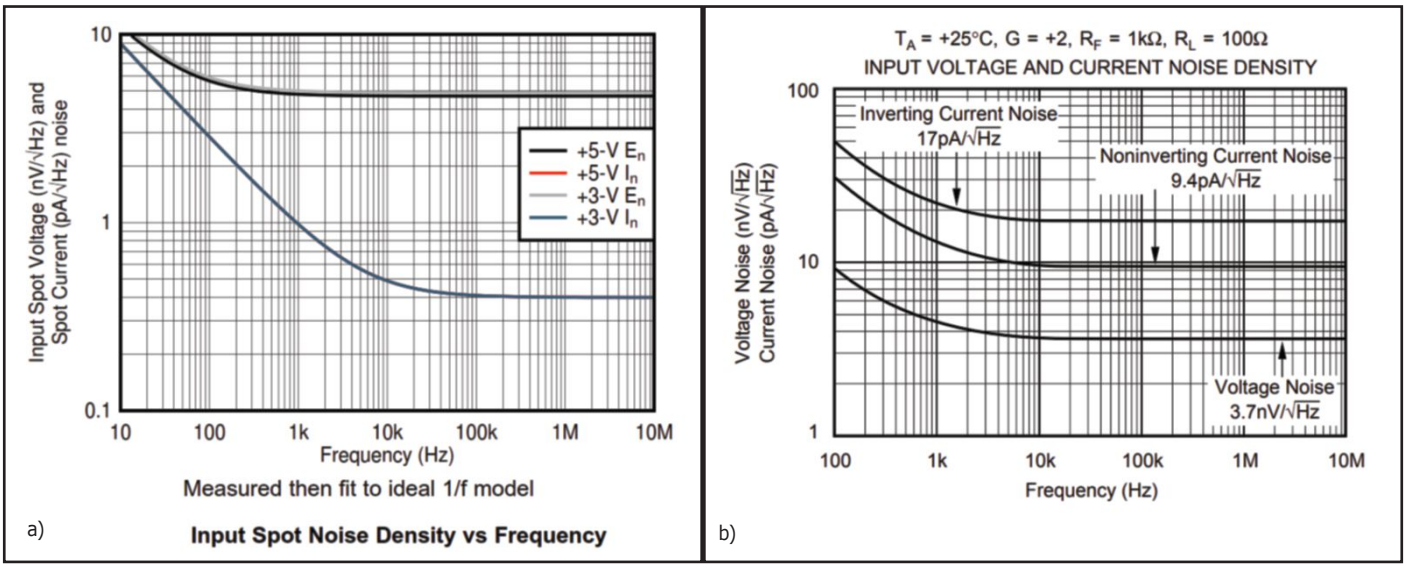


Figure 12a: Input noise density of the VFA OPA837; Figure 12b: Input noise density of the CFA OPA684 (Image courtesy of Texas Instruments)

An exemplary CFA and an exemplary VFA have been considered. These devices are generally representative of members of these two amplifier classes. Certain performance characteristics of interest to audiophiles have been presented.

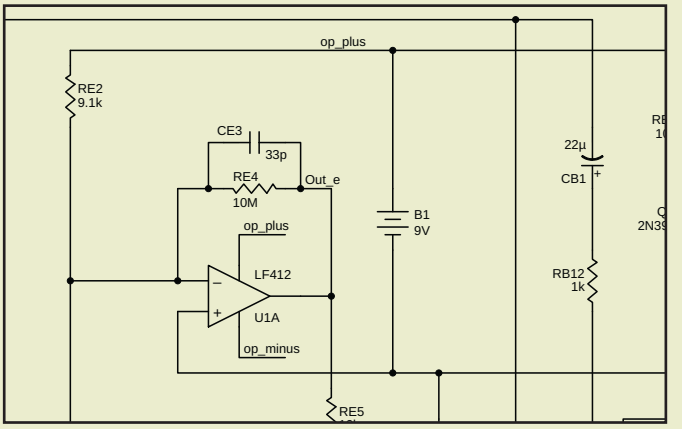
Amplifiers that employ current feedback can readily exhibit constant bandwidth with varying closed loop gain. All that is needed is for the network impedance R_{eq} seen by the inverting input to be kept larger than that input's impedance Z_{FB}. This effect can also be exhibited in voltage feedback amplifiers by adding a suitable resistor between their inputs (see Part 1, Figure 6). However, there is no good reason to do this, as it reduces bandwidth. Regardless of these relative impedances, current feedback amplifiers inherently benefit from slew rate enhancement, putting the current flowing from the output into their inverting inputs to good use. Slew rates of voltage feedback amplifiers can be increased by additional specialized circuitry within the amplifier.

Voltage feedback amplifiers' generally higher audio frequency loop gains usually afford them lower distortion and higher CMRRs and PSRRs. Their input structures typically allow them to benefit from lower input noise, bias currents, and better DC characteristics. They also lack the inverting mode limitations of current feedback amplifiers. The slew rates and bandwidths of modern units, though perhaps not as great as those of current feedback alternatives, may not be barriers to use in audio designs. Readers can draw their own conclusions as to which class is better suited to a particular audio application.

About the Author

Christopher Paul was born in the US the year before the Nobel prize was awarded for the invention of the transistor, the development of which he claims no credit for. Chris has worked for companies whose products are in the fields of communications, electronic article surveillance, and enterprise hand-held computers. He holds two M.Sc. degrees in electrical engineering from Brooklyn Polytechnic (now part of New York) University.

Figure Correction for Part 1 of "Current Feedback: Fake News or the Real Deal?"



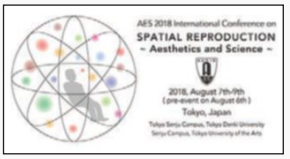
Part 1 of the "Current Feedback: Fake News or the Real Deal?" article, which ran in the July 2018 issue of *audioXpress*, contained an incomplete figure. The label "Out_e" was left off of Figure 2. It should have been associated with the output of op-amp U1a. Here is the portion of Figure 2 that was missing.



Industry Calendar

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

August 7-9, 2018
2018 AES International Conference on Spatial Reproduction—Aesthetics and Science
Tokyo, Japan
www.aes.org/conferences/2018/spatial



Many practical technologies have been developed in the area of spatial audio since the last international conference in Tokyo, Audio Engineering Society (AES) International Conference on Spatial Audio—Sense the Sound of Space—was held in 2010. The industry and all research institutions globally have managed to handle the technology and used it to create countless works since then. In the light of such technology-driven content creation, it may be the time to address the aesthetics in audio production.

The AES committee for this conference is expecting discussions in the following topics:

- How can the Spatial Audio technology contribute to music and other artistic productions?
- What technologies and scientific knowledge are expected from musicians and the content creators?
- What are the keys for connecting Science and Aesthetics in this area?

August 10–12, 2018
2018 Hong Kong High End Audio Visual Show
Hong Kong Convention & Exhibition Centre (HKCEC), Hong Kong
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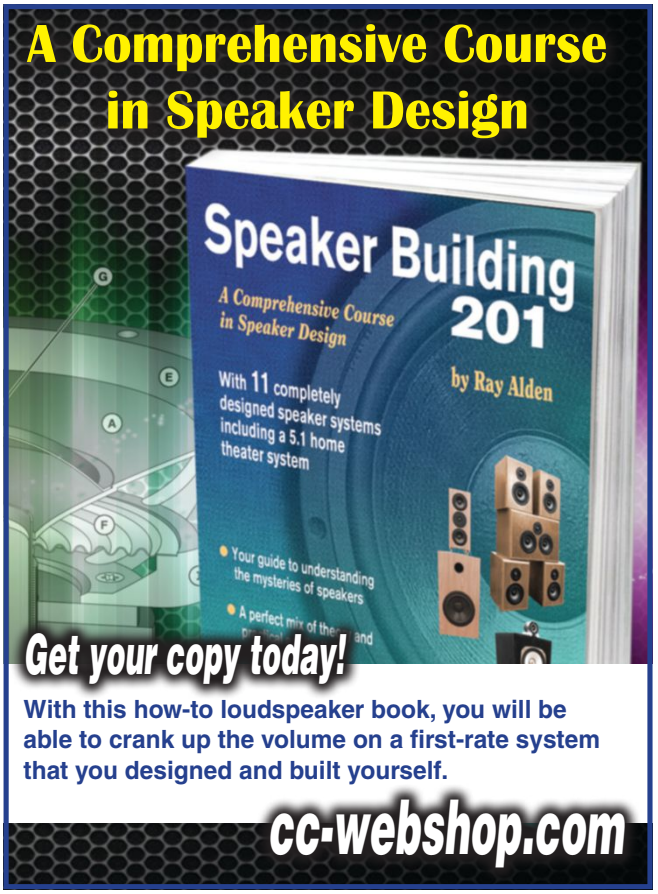
This three-day event, sponsored by Far Eastern Music & Audio Enterprises, Ltd., provides a good opportunity in Asia to showcase all types of audio products in a vibrant market, find new suppliers, and do business, The event features plenty of displays in personal audio, headphones, speakers, cables, accessories, and all audio components.

August 20-22, 2018
2018 AES International Conference on Audio for Virtual and Augmented Reality
DigiPen Institute of Technology
Redmond, WA
www.aes.org/conferences/2018/avar



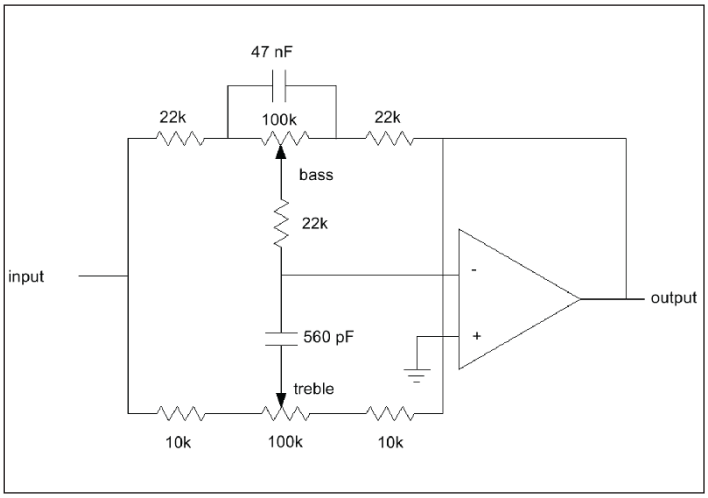
Featured speakers at the 2018 Audio Engineering Society (AES) International Conference on Audio for Virtual and Augmented Reality include Jean-Marc Jot, Distinguished Fellow, Magic Leap; Ivan Tashev, Microsoft Research Labs - Redmond; and Ravish Mehra, Lead Research Scientist, Facebook Reality Labs.

At the conference, Jot will disuss "Unleashing Wearable 3D Audio," focusing on the effects of acoustic reverberation, reflectors, and obstacles in the virtual and physical environments, driven by combinations of pre-computed and real-time propagation models. Tashev plans to focus on the "Capture, Representation, and Rendering of 3D Audio for Virtual and Augmented Reality. Finally, Mehra will discuss "Sound Propagation for Virtual and Augmented Reality."



Article Update

Our June 2018 *audioXpress* issue featured an Hollow-State Electronics article by Richard Honeycutt, titled "The Evolution of Tone Controls in Tube Amplifiers" Unfortunately, the article contained an incorrect schematic, Figure 9 shown on p. 64. The correct image is provided here.





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