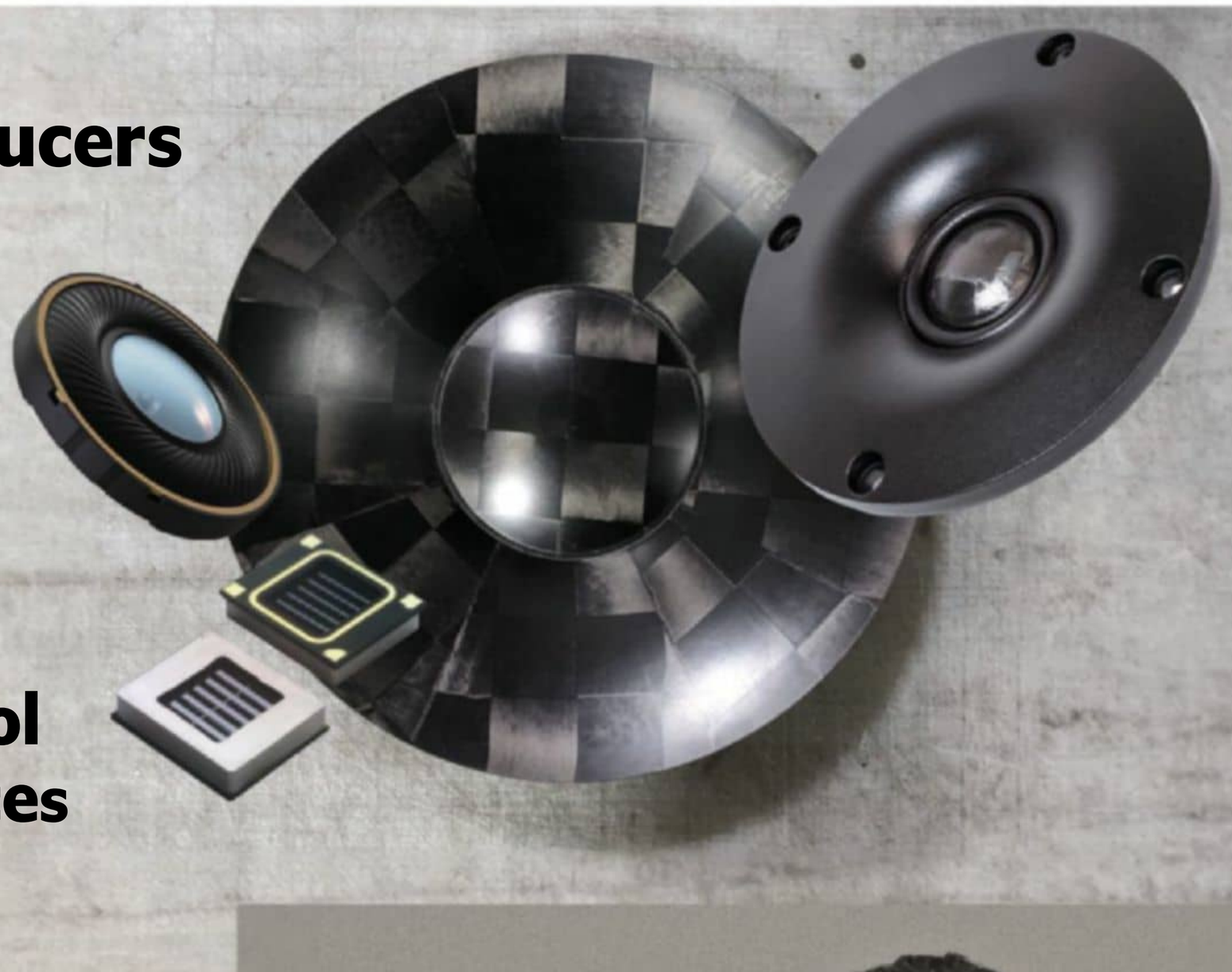
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The Things That Stand Out



As we start a new year for this publication, it's time to offer some perspective about the things that inspire us for 2024.

Following my previous reference to the Urbanista Malibu, the world's first self-charging wireless speaker using an integrated Powerfoyle membrane that charges the battery whenever exposed to indoor or outdoor light, Exeger, the developer of the Powerfoyle technology, is now top of mind for 2024. We will attend Exeger's press conference, selected to be part of the Oficial CES 2024 Media Days, which is in itself a significant honor and a sign of the impact this Swedish company is having on the technology world.

Exeger has also announced a strategic partnership with Tymphany, one of the largest OEM/ODMs of speakers and audio systems in the world, to produce a self-powered speaker that will be unveiled at CES 2024. Tymphany's Hybrid-life Green Speaker reference design will be powered by Exeger's patented solar cell technology and is certain to cause significant impact in the audio and consumer electronics industry.

Exeger's Powerfoyle technology can significantly contribute to drastically cut down the need for disposable, single-use batteries in common objects such as remote controls, alarm clocks, bedroom lights, portable radios, speakers, and much more. A significant contribution to reduce dangerous waste and toward sustainability.

Next on our list is—once again—the evolution of wireless power technologies in general. Complementing energy harvesting technologies from light/solar sources, we are now seeing an important evolution in the technologies that will continously power our devices.

The new and very exciting Qi2 standard, officially launched with products making its debut at CES 2024 is one of the most significant developments in the mobile and consumer spaces, boosting convenience, interoperability, safety, and sustainability.

Also exciting is the use of projected and long-range power distribution, which will be key for the smart home—and will soon break the last frontier for wireless in the office. Leading the efforts in RF wireless charging are a series of companies with actual working solutions. NuCurrent, Atmosic, Ossia, Wi-Charge, Powercast, and Energous are among the names that deserve attention.

For audio development, combined with the general availability of USB Power Delivery up to 240W and 48V, these new exciting technologies will allow us to rethink the very foundation of product design.

Meanwhile, while the first Bluetooth LE Audio and Auracast products are finally reaching the market, we are seeing manufacturers push for a formal adoption of LC3plus, as a higher audio quality LE Audio profile. But given the pace of things on the Bluetooth front, I bet we will be seeing quicker adoption of higher bandwidth alternatives to support high bit-rate, lossless audio streams. We have just seen an announcement from Qualcomm, with the use of micro-power Wi-Fi to extend wireless audio range and support for up to 24-bit/192kHz lossless in true wireless form factors. Something that could likely see Apple's support, given that its AirPlay communication protocol already leverages Wi-Fi Direct (Wi-Fi Peer-to-Peer) to, among other things, stream and broadcast audio (like Auracast does). And we know that Apple will be making a key announcement in 2024 (likely to extend two or combine all these technology options).

Also looming in this space is ultra-wideband (UWB), a technology that has been adopted somewhat stealthily and is now built into billions of devices, used for Find Someone/Something, indoor navigation, and lock/unlock features. The FiRa Consortium just updated its 2.0 Technical Specifications for UWB Technology introducing applications such as "Point and Trigger" that lets users control UWB-enabled connected systems, such as TVs or smart speakers.

And now, efforts supported by the UWB Alliance and the Audio Engineering Society (AES) are targeting the development of UWB standards for ultra high-quality wireless audio applications. The AES has formally created the SC-02-02-B Task Group to develop a new standard for UWB high-resolution, low-latency audio interface, which will build upon and provide extensions to the IEEE 802.15.4ab draft standard. Industry support is pouring in, and we are hoping to see a working demonstration of soon-to-be-available headphones—from Lenbrook's PSB brand—at CES 2024.

Of course, CES is just the start of what is expected to be a strong year for audio shows, with the Audio Expo North America (AXPONA, April 12-14) and High End Munich 2024 (May 9-12) among the shows we are eagerly anticipating.

I'll close with our renewed commitment to expand on our trajectory initiated in 2013, when we embraced our mission to follow audio innovation, true to our motto, Advancing the Evolution of Audio Technology. We will continue to expand on our mission to offer the leading Audio Development and Design Publication.

J. Martins
Editor-in-Chief

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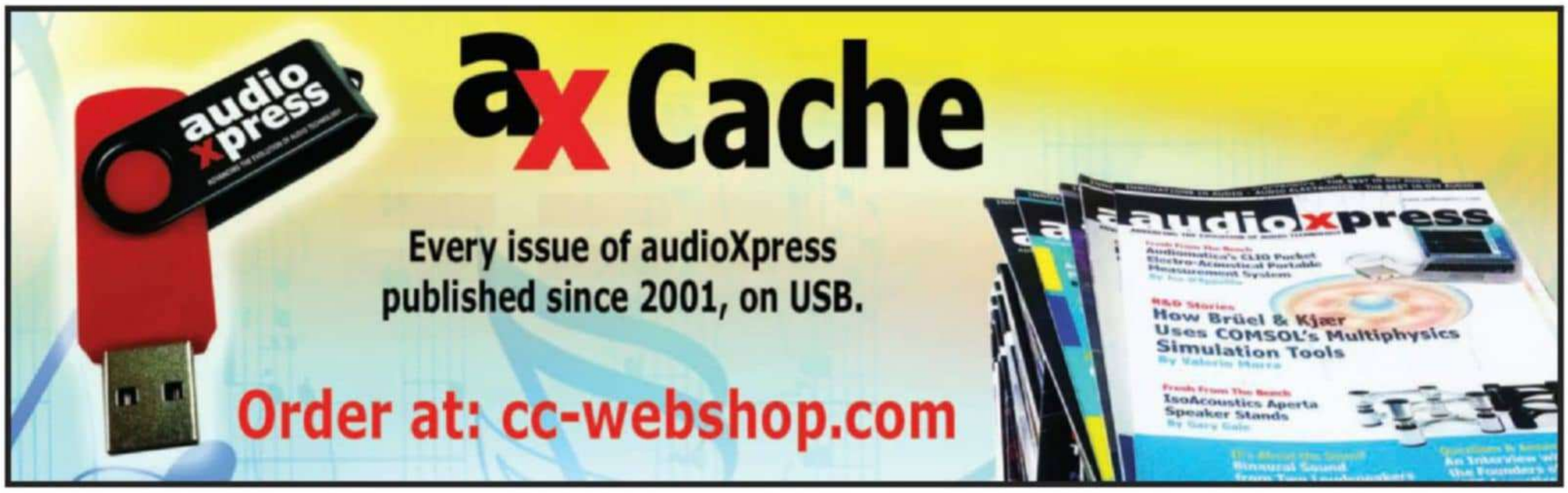
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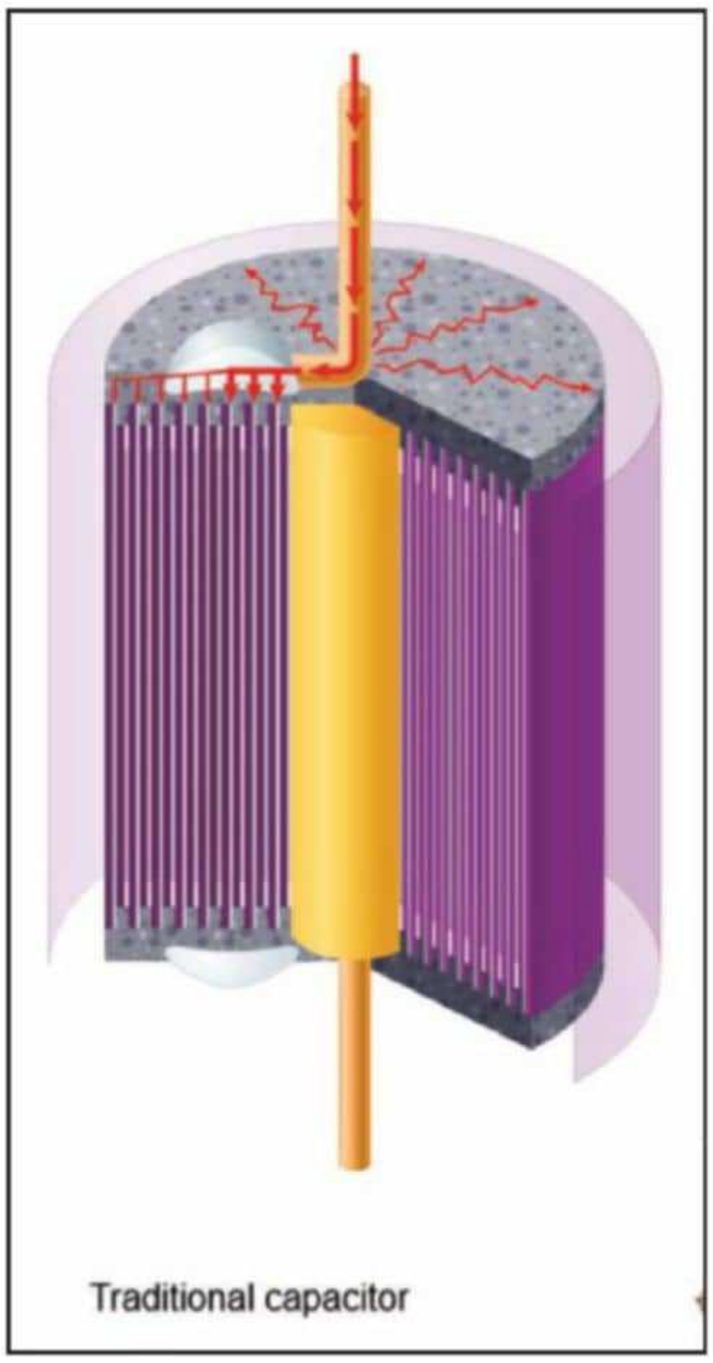
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Engineering to Control Diaphragm Resonances

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Thin-ply carbon diaphragm (TPCD) technology, pioneered by Composite Sound, tries to control the negative effects of resonances rather than trying to reduce them. Now, the company is entering the over-ear headphone market with the TPCD diaphragm technology, enabling headphones with better sound quality, improved bass performance, and higher sensitivity.

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What do motional feedback, silicon wafer steppers, voice coils, and Class-D amplifiers have in common? Everything, as far as Prof. Ir. Robert Munnig Schmidt is concerned. *audioXpress* Technical Editor Jan Didden decided to find out why and interview the man known as “mister motional feedback.

46 The Role of Resonance in Audio Clarity

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Years of research have shown that resonance from the body of a capacitor in a loudspeaker crossover circuit has an audible impact on sound quality. This article details the extensive testing, measurements, and conclusions, which allowed perfecting a manufacturing process minimizing resonances in capacitors.

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By Paul Marchese

This article presents a fairly in-depth analysis of composite amplifiers, including the effects of local as well as global feedback. In his exploration, the author looks at stability in detail and debunks a few rules of thumb, before applying clever measurement techniques to measure the very low distortions.

58 Is Digital Jitter Really a Problem?

By Ethan Winer

Since the dawn of the digital era decades ago, the timing error known as “jitter” has been accused of harming audio fidelity in A/D/A converters. In this article Ethan Winer explains what jitter is, what causes it, and—most important—lets you hear jitter in controlled amounts so you can decide for yourself how damaging it really is.

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Market Update—Drivers and Transducers

8 New Options for Improved Designs

By J. Martins

From home speakers to high-end speakers, from soundbars to subwoofers, the world of transducers continues to evolve mostly outside of the spotlight, with crucial innovations occasionally breaking through. *audioXpress* highlights those that stand out.

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Learning about hollow-state electronics used to be as easy as DIY trial-and-error, getting information from a family member or friend, or reading technical books and magazines such as *Popular Electronics*. Even today with all the information available online, finding a knowledgeable and patient mentor and lots of practice still seem to be key factors.

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New Options for Improved Designs



By
J. Martins

From home speakers to high-end speakers, from soundbars to subwoofers, the world of transducers continues to evolve mostly outside of the spotlight, with crucial innovations occasionally breaking through. *audioXpress* highlights those that stand out.

Two years ago, in the January issue of *audioXpress* we covered an extensive number of technology and material innovations for drivers and transducers for all types of applications. That remains our favorite approach to the topic, given how this very specific discipline remains a solutions-oriented segment of the audio industry.

In most cases, manufacturers and brands of finished loudspeakers specify their own custom drivers, designed to their project requirements, even if essentially based on the technologies and specific base models that driver manufacturers have on catalog. For the headphone and in-ear product segment that's even more the case, since there isn't even such a thing as a "product catalog," and manufacturing designs and choices are rarely discussed in public, unless you are a large volume OEM customer.

As we are the publishing house for *audioXpress*, *Voice Coil*, and the *Loudspeaker Industry Sourcebook* directory, we know too well how speaker driver manufacturers remain engineering- and production-oriented, and very rarely engage in conventional public relations, or even any promotional efforts. With a few rare exceptions, it's always a reason to celebrate when we see a press release about new speaker drivers. And even then, the information made available about new products and developments is extremely "dry" and requires a lot of "digging," or the vast experience of our own Vance Dickason and his Test Bench characterizations, published for more than three decades in *Voice Coil*, to fully uncover the details and the actual product story.

For this Market Update we are trying hard to cross that "curtain of silence" to dive into the drivers and the transducer product world and highlight the things that stand out in the current market, regardless of the application field.

Selected products are some of the notable examples of a combination of technology, design, and innovation that deserves to be recognized and closely followed for new product implementation ideas.

Greetings from Denmark

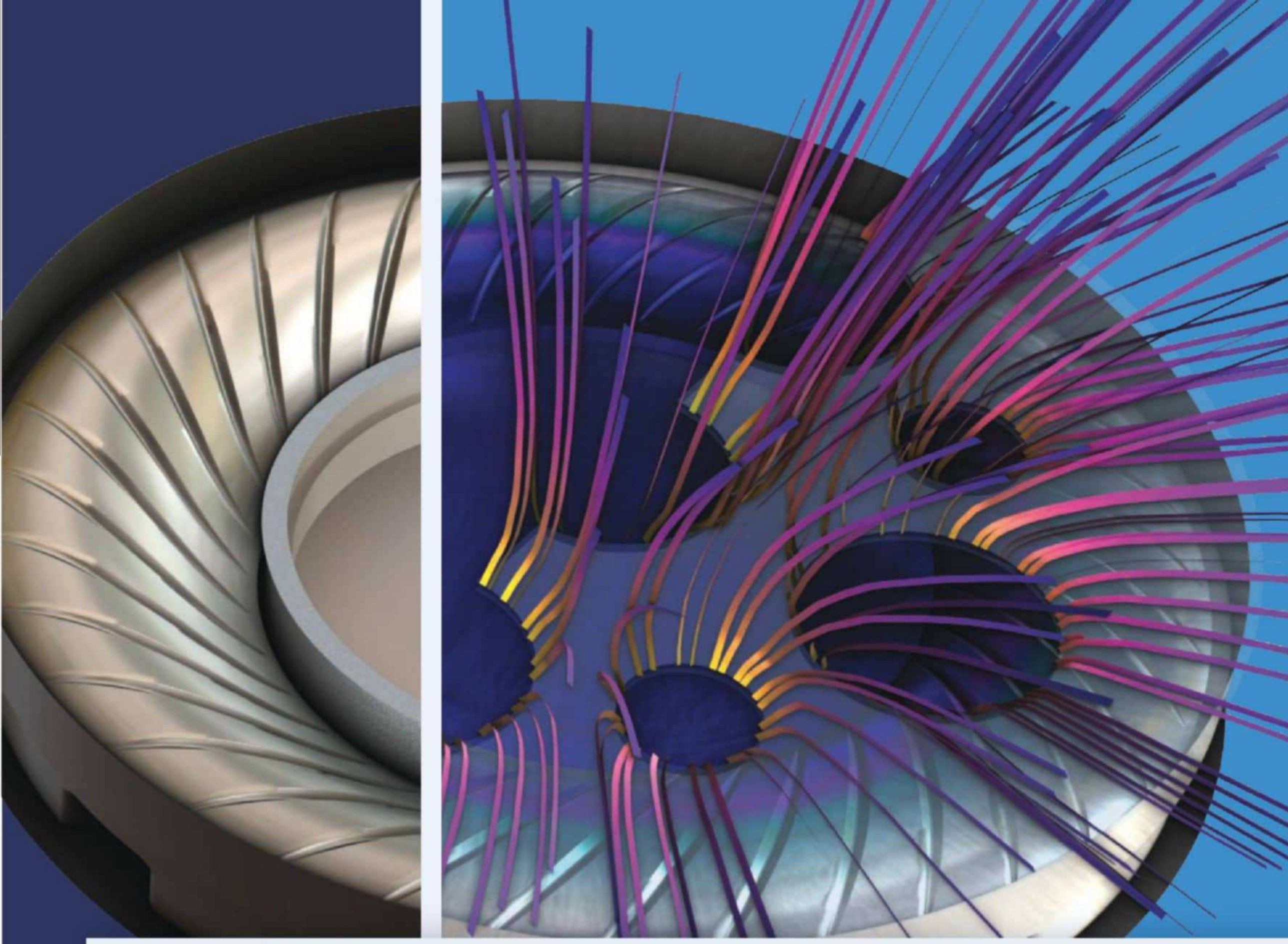
Yes, it's no big surprise that the world of speakers is dominated by the result of more than 100 years of Scandinavian expertise, with Denmark at its epicenter. With a notable heritage that generated some of the world's leading speaker brands, today it's mostly through the R&D and engineering efforts that Danish companies stand out even if the largest brands have all joined forces with Asian manufacturing powerhouses, from Eastech to Tymphany and Goertek to SB Acoustics.

But one needs to closely follow Danish companies to see the tip of the iceberg when it comes to the cutting edge of high-end audio and consumer audio applications. A remarkable example is that of Scan-Speak and its expanding Ellipticor range, now with six base models, including two tweeters, one midrange (the remarkable D8404/552000 3" Dome Midrange), one woofer, and two midwoofers. These advanced speaker drivers, all with elliptical voice coils and AirCirc optimized magnet system, have recently been complemented with the 18ME/9642T00 midrange, which is the first OEM-only Ellipticor driver to be offered by Scan-Speak, not available to the retail market.

As usual, the new driver has been fully characterized by Vance Dickason in a Test Bench article in *Voice Coil*, so all the details



The Scan-Speak Ellipticor range now covers all frequency ranges with six models, plus its first OEM-only midwoofer. Manufacturing might come from Asia, but innovation is still very much based in Denmark.



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The new R3004/602200 Illuminator compact tweeter improves on all fronts the existing models of the series, and offers even better cosmetics, with a black anodized aluminum faceplate, improved gold-plated terminals, and a high degree of design flexibility.

are available there. What's important to highlight is that this new OEM-only driver is a clear sign that the benefits of the oval (elliptical) voice coil shape in defeating cone and dome standing wave modes are being well-recognized by manufacturers. Very similar to the existing standard 18WE, the new 18ME features an uncoated paper cone and oval dust cap, with a big difference being the coated pleated M-type cloth surround, similar to those used in professional audio, instead of an NBR surround used in other midwoofers in the series.

Another example of Scan-Speak's ability to surprise an industry where excitement doesn't come often, the new R3004/602200 ring radiator tweeter, a 1" replacement for older Illuminator models, is a compact option for speaker designers, providing a high degree of flexibility, fully comparable to the renowned larger Illuminator and Revelator tweeters. Following the launch of the updated D3004/602200 Illuminator dome tweeter, Scan-Speak optimized the original designs based on the company's patented phase plug design and symmetrical drive (SD-2) motor.

The new Scan-Speak R3004/602200 Ring Radiator Tweeter will display lower distortion and improved frequency response extending to beyond 40kHz due to optimized internal construction, and increased sensitivity (+2dB) due to higher grade neodymium magnet. Improved Q is also a result of the optimized internal volume in a new high precision reenforced polycarbonate rear chamber.

The Purifi Example

Currently, when we speak of Denmark, the name of Danish audio powerhouse Purifi Audio immediately jumps to mind, even though it is one of the younger companies in the sector. But in a short period of time, the Purifi all-star engineering team has managed to make its mark, and push innovation further in both audio amplification and speaker transducers.

Before launching what will be the company's first tweeter—a driver that was demonstrated at the High End 2023 show in Munich, both in private listening sessions and in the company's booth—Purifi has focused on completing its range of midrange, woofers, and extended woofers. Those are available in both retail versions, recognized by the Purifi-red magnet top plate and sold in nice-looking one-unit boxes, and OEM models, with all-black magnet system and more variants, sold only in volume quantities.

Adding to the original formula—featuring proprietary fiber paper cones—used in the successful PTT4.0X and PTT6.5X midbass drivers, Purifi also announced aluminum-cone versions of the PTT6.5X, delivering extended frequency range with ultra-low distortion for superior midrange performance.

And the 6.5" models were followed by an 8" "Ultra Low Distortion Extended Woofer," the PTT8.0X04-NAB-01/PTT8.0X08-NAB-01 and corresponding passive radiator matching driver. The new 8" driver expands Purifi's extended-stroke woofer family and incorporates the same recognizable NeutralSurround suspension and ultra-linear PureDrive motor system technologies to enable record-breaking linearity specifications over an extended bandwidth.

This driver was followed in 2023 with the latest 10" extended woofer, the largest from the brand so far. The Purifi PTT10 is a member of its ultra-low distortion extended-stroke woofer family, incorporating the same NeutralSurround suspension and ultra-linear PureDrive motor system. Like the 8" model before, the 10" features a vibroacoustic optimized cone and dust cap in black anodized aluminum, all optimized according to the principles determined by the company's research.

As Purifi details, research has identified the parameters that have so far prevented extended stroke drivers from breaking through high performance audio parameters.

"The distortion of the acoustic output of a driver is a combination of several separate distortion mechanisms in the motor, cone, and suspension. When testing a complete driver with sine waves, a situation often arises where a small tweak seemingly improves harmonic distortion (HD) by letting two mechanisms counteract each other. This always leads to a clear worsening of intermodulation distortion (IMD) that becomes obvious when testing the unit with a more complex signal. This is why traditional HD tests fail to predict subjective sound quality."

Purifi takes care to optimize the various distortion mechanisms separately, thus guaranteeing that any measured improvement truly reflects a real audible improvement no matter what the signal is. Accurate mathematical models are developed for the motor, suspension, and the vibroacoustics of the dome and surround. These models clearly explain several distortion mechanisms in each domain, which provides fundamental insights into better ways of constructing the motor, cone, and suspension. Finally, the



The Purifi Audio booth at High End Munich 2023 (included in the IPS B2B section of exhibits) allowed some visitors to see and hear—for the first time—the new Purifi tweeter.



The new Purifi PTT10 10" woofer follows the same design principles as the company's smaller woofers, featuring long-stroke performance with very low distortion over its full excursion.

same models are used to numerically fine-tune the geometry. This way of working reliably reduces multiple distortions mechanisms, which secures low IMD for complex signals.

The Purifi engineering team targets low force factor modulation to prevent voice coil current from modulating force factor (Bl). The Purifi motor completely avoids this classical Achilles' heel of long-stroke drivers. This translates into low IMD for clean, undistorted midrange even in the presence of a heavy bass, and equates to low impedance modulation, meaning that drive current is not distorted by cone motion.

Another area of attention is to achieve very constant force factor over excursion, to prevent voice coil position from modulating the force factor. This is the classic cause of "burbling" (i.e., amplitude modulation) of the midrange by large low-frequency cone excursions. And because the surround also contributes to sound output and produce distortion as they deform, Purifi's Neutral Surround geometry reduces harmonic distortion as well as intermodulation distortion, without constraining motion.

As detailed in the company's research, another major goal for Purifi is to achieve low magnetic hysteresis distortion. "Hysteresis means that magnetic domains in iron retain traces of previous magnetization/demagnetization cycles, which cause distortion when magnetized or demagnetized again. This distortion masquerades as benign harmonic distortion when tested with a sine wave but takes on a crackling or noisy character with more complex signals. Removing hysteresis distortion translates into a very "fluid" presentation with excellent front/back separation in the stereo image and a perfectly black background between instruments."

From Norway to Taiwan

Also from the Scandinavian region, but further north, SEAS Fabrikker of Norway continues to introduce cutting-edge drivers, not only for high-end audio applications—the company's traditional focus—but also expanding toward larger volume, consumer audio designs. An example of such efforts was the Orbit series, benefiting from the fact that SEAS is now part of a much larger organization within Sonavox and Merry Electronics.

SEAS Fabrikker has long been associated with the high-end audio market and very high-quality speaker drivers for some of the most demanding applications, but since its early days manufacturing speakers for radios, the Norwegian manufacturer always served a vast variety of applications for its OEM clients, including higher-volume consumer products. Today, SEAS is an integral part of the Merry Transducer Business Unit (BU) with R&D teams both in the Norwegian town of Moss, and Taichung in Taiwan. The company provides high-volume manufacturing in addition to its production facility in Moss, which is mainly focused on products intended for the high-end market.

To quickly recap, in 2014 SEAS Fabrikker was acquired by Sonavox Canada, a product design and development company, specializing in home and professional audio speakers and amplifiers. In 2017, Merry Electronics acquired 100% control of the Sonavox group, which is one of the world's leading manufacturers of audio products and systems for the automotive industry. The new SEAS Orbit speaker series has been developed on that strong foundation of more than half a century of engineering and manufacturing expertise, combining the core knowledge and competencies of both SEAS and Merry Electronics.



Part of the Purifi team at the Munich presentation of the latest drivers from the company, with Carsten Tinggaard, responsible for Transducer Technology at Purifi, Claus Neesgaard, Bruno Putzeys, and Lars Risbo, all of them co-founders.



The new SEAS Orbit Series drivers were developed in cooperation with Taiwan-based Merry Electronics, aiming to address the requirements of OEMs looking for high-quality speakers for smart speakers, soundbars, and videoconference systems.

Market Update — Drivers and Transducers



SEAS introduced the SM011-08 L22ROY2 8" for use in applications where space is limited, but the need for extreme, detailed, and precise bass is high.

The SEAS Orbit series was launched with three small-size drivers (1.5" and 2"), all full-range models with a recommended frequency range from 200Hz, offering paper and aluminum cone options. These are drivers that highly benefit from the extensive know-how of this integration, offering both advanced designs and the consistency required by higher volume manufacturing. Designed and developed especially for the most demanding videoconferencing and advanced smart-speaker products, the SEAS Orbit speakers are available for ordering in smaller sample quantities and are available and specified for large-scale mass production. Each driver is designed to offer the ultimate combination of quality, performance, and value.

Moving to the company's traditional high-end catalog, we have the new SM011-08 L22ROY2 Extreme 8" driver, which SEAS says is the result of years of development based on the same design philosophy with extreme excursion and low noise as the very successful bigger model, the L26ROY. The L22ROY2 compact woofer was optimized with an advanced magnet system featuring an additional aluminum ring above the pole piece to reduce distortion even further with a symmetric and linear inductance for large excursions.

SEAS says that among the engineering team that created this driver, a favorite application is to use two parallel connected drivers in a closed box, mounted on opposite sides to reduce cabinet resonances. The low end should then be tuned to fit the room and personal preferences

With similar goals in mind, SEAS also recently expanded its Excel Nextel family with the W22NY003, an 8" woofer featuring a FEA-optimized magnet system, titanium voice coil, and copper shorting rings strategically integrated so that the inductance is linearized, ensuring low distortion. The W22NY003 is designed to shine in a variety of applications, with both closed-box systems or for ported designs.

The SEAS Excel Nextel series features a "sandwich" cone using paper coated with Nextel paint on the front surface and damping compound on the rear. The resulting cone is both lightweight and well damped, offering an extremely linear frequency response.

The SEAS Excel woofer series has also been expanded to four proprietary cone materials, available in drivers sized from 5" to 10", including its unique magnesium cones renowned for the high definition that results from extremely high stiffness and good internal damping. And the latest initiatives from the Norwegian company was to expand the series with new Graphene treated magnesium cones, and now being the first company to introduce the latest Metamodal TPCD cones.

First TPCD Metamodal Drivers

Unveiled at AXPONA 2023, the two first models from its new Excel line of TPCD Textreme cone and dome transducers are the Excel T29X tweeter and the Excel W19NX. In this issue, *audioXpress* readers will find the second article about Composite Sound's engineering effort with thin-ply carbon diaphragm (TPCD) Metamodal and the substantial benefits that this fabrication method awards to the already-proven TPCD advantages.

This second generation of the TPCD technology is engineered to have low weight, high stiffness, and controlled damping that SEAS matched with an optimized suspension to deliver a smooth, extended frequency response.

The new SEAS T29X uses a 29mm (1.1") diameter TPCD dome driven by an Excel Hexadym motor structure that features six radially charged neodymium block magnets, effectively eliminating the pole piece directly behind the diaphragm, and consequently a lot of the reflective clutter that degrades high-frequency detail. The resulting T29X driver delivers a short-term power handling of 200W (90W long term) with a sensitivity of 92dB at 2.83V/1m, 510Hz resonance, 4Ω nominal impedance (3.6Ω DCR), and a frequency range of 1.5kHz to 32kHz.

The 5.25" Excel W16NX005 midwoofer has a TPCD cone and dust cap, a 39mm (1.5") diameter titanium non-conductive voice coil former wound with round copper-clad aluminum wire (CCAW), a FEA-optimized ferrite motor fitted with copper shorting rings, a minimally reflective injection-molded aluminum frame (basket), with 300W short-term power handling (150W long term), 87.9dB and 2.83V/1m sensitivity, plus gold-plated terminals.

The TPCD Metamodal's state-of-the-art technology is combined in these drivers with the experienced craftsmanship from SEAS. Olav Mellum Arntzen, SEAS' CEO explained why his company was the first to embrace Metamodal TPCD. "In the loudspeaker industry, manufacturers like ourselves, are always looking for new, exciting materials. When the TPCD cones came to market, they immediately caught the interest of many. We were no exception. From early on, we embarked on a long mission together with Composite Sound, with the end goal to make the ultimate cones. There have been countless simulations, testing, and sample assemblies. And here we are—some few years later, with the result.

"We are more than proud to finally present and launch the second generation TPCD—the Metamodal TPCD. In itself the TPCD material exhibits exceptional stiffness-to-weight ratio. The groundbreaking part is how the new Metamodal construction controls the inevitable cone break-up to push the boundaries of

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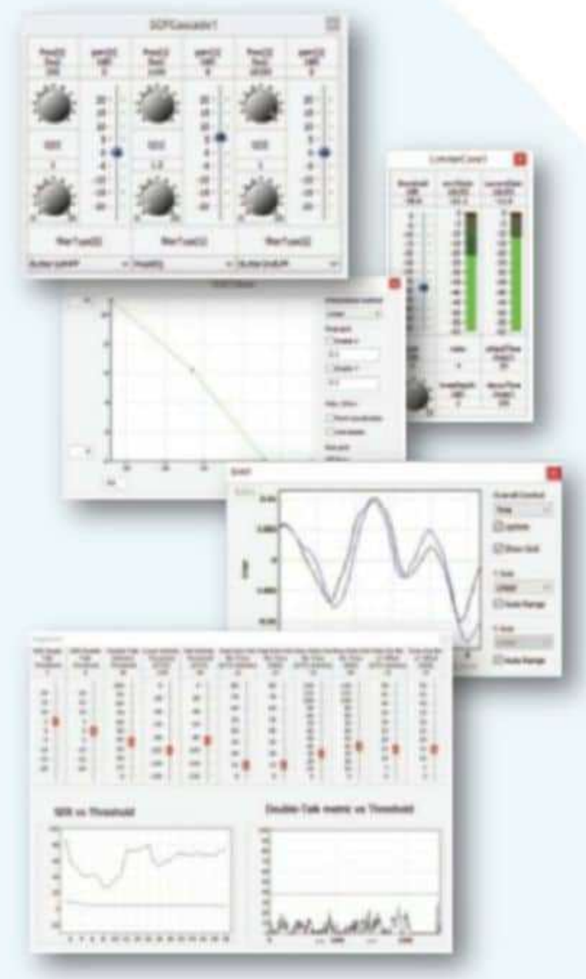
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Market Update — Drivers and Transducers



The new SEAS Excel W16NX005 6" woofer is the first product to truly demonstrate the full benefits of a Metamodal TPCD cone, the second generation of TPCD.

sound reproduction to an unprecedented level. This is achieved by its unique composition, combining thin spread carbon tapes in several layers where the thickness and stiffness, and the direction of the stiffness, is varied throughout the cone. Engineering the cone like this to optimize the modal break-up of the cone make it possible to have it all—high stiffness, low weight, and good damping.

"The technological marvel of the cone enables the other parts of the driver to really shine as well. The W16NX005 loudspeaker driver incorporates a meticulously optimized linear and symmetric magnetic field with precisely fitted copper parts to also linearize the inductance. Thus, eliminating unwanted distortions and allowing the driver to effortlessly handle all the dynamics in the music. Furthermore, the suspension system of the driver has been fine-tuned to provide unparalleled linearity and control.



The new SEAS T29X001 ultra-high-performance tweeter also features a Composite Sound Metamodal diaphragm (second-generation TPCD) combined with the patented HEXADYM magnet system.

The carefully selected materials and optimized geometry work in perfect harmony, enabling the driver to respond swiftly and accurately to the input signal. Whether it's a gentle whisper or a thunderous roar, the W16NX005 maintains its composure, ensuring a truly immersive and engaging listening experience."

"In my time at SEAS we have tested all available materials for loudspeaker parts. Always. It is easy for our team to glue in a new part, measure, and listen. And I mean all kinds of material. Once we even cut out an eggshell and glued it into a tweeter. Of course, too heavy, but just so you get the point.

"Metamodal TPCD is a technology where you can build up a cone with the building bricks: thickness, damping, and stiffness. You can tailor the frequency response, add damping, reduce resonances, and increase stiffness, exactly where you need it to be. Instead of doing it with DSP or a ton of passive components before the driver, you can manage it inside the cone," adds Olav Arntzen.

"We understand that true audio enthusiasts demand nothing but the best, and the W16NX005 loudspeaker driver is designed to exceed those expectations. It represents the pinnacle of performance and precision, where every note is rendered with astonishing clarity, depth, and realism."

From Denmark to Indonesia

Because we mentioned TPCD, it's only fair that we mention SB Acoustics, one of the very first speaker driver manufacturers to announce the adoption of TeXtreme TPCD materials for cones and diaphragms. Although it wasn't first to market with TPCD material diaphragms, SB was probably the brand that most contributed to its market recognition and acceptance.

Gradually, SB Acoustics expanded its offering to all categories, from extreme high-end tweeters all the way to woofers, now with four available sizes—including a 9.5" woofer with a 2" voice coil. With an increased rated power, the TPCD Satori range now offers extensive options for speaker designers. The simple fact that SB Acoustics launched TeXtreme-based drivers in its Satori high-end audio range, designed by Danesian Audio, was a clear sign of the performance that Frank Nielson and Ulrik Schmidt managed to achieve.

Originally developed for aerospace and Formula 1 applications, Swedish company Oxeon perfected the TeXtreme technology based on thin-ply and spread tow principles. In 2016, Oxeon successfully applied its TeXtreme thin-ply carbon reinforcements to create a new type of speaker diaphragm. The significant technical and commercial success led Oxeon to spin off its audio-related business to a dedicated, highly focused company—Composite Sound, which has now expanded TPCD to a whole new level thanks to its new Metamodal generation of cones and diaphragms.

Like other manufacturers, SB Acoustics first experimented with the full potential of TPCD in the high-frequency range, with a version of its successful Satori TW29B beryllium tweeter and launched the Satori TW29TXN-B Advanced TeXtreme dome tweeter. The advantages and similarities with the Beryllium version was evidenced by Vance Dickason's *Voice Coil* test bench. As he stated, "I have no doubt that this tweeter performs admirably. As far as the TeXtreme diaphragm goes, so far it seems to be a less

costly substitute for beryllium. I have done informal subjective comparisons of the TW29TXN-B-4 to a very good sounding aluminum diaphragm, and the TeXtreme diaphragm sounded very detailed with more high-frequency content. And comparing the TW29TXN-B-4 to a high-end beryllium dome tweeter, I thought that in terms of detail and definition, TeXtreme is very close, with the beryllium sounding a bit "warmer" and the TeXtreme a bit more forward."

But quickly the market also realized the advantages of TPCD for the lower range and particularly the critical midbass frequencies. The MW16TX-8 6.5" woofer was measured by Vance Dickason in the June issue of *Voice Coil*, and he recognized the unique properties and the timbral desirability of the TPCD diaphragm material as well the overall design and build quality of this Satori woofers.

The fundamental design, with its proprietary six-spoke cast-aluminum frame, and a completely open below the spider (damper) mounting shelf for cooling has been consistent through the four models launched so far. Additional cooling for these drivers is provided by large vents in the voice coil former, as well as pole vent. The TPCD cone assembly consists of a curvilinear inverted single piece "bowl" shaped cone, with compliance provided by a generous NBR surround that has a mostly shallow transition to the cone attachment, with the remaining compliance coming from a flat BIMAX spider.

The 5" Satori MW13TX-4 midrange was one of the most well-accepted models in the range and we can find it in numerous hi-fi speakers, from renown brands and DIY speaker builders alike. This was followed by the larger 7.5" Satori MW19TX-8 midbass woofer.

As the previous models have shown, the TPCD cone with the SB Acoustics Satori motor system makes these woofers a good candidate for high-end, two-channel home theater or studio monitor system designs. The new larger SB Acoustics 9.5" Satori WO24TX woofer now offers improved piston operation for its diameter ratio, a large, optimized motor system (with a 1.36kg magnet) with under-cut pole piece and dual shorting rings, and a linear symmetric suspension design for improved dynamic performance.

With the Satori line being SB Acoustics' flagship products, it's also significant to see the first Satori coaxial driver to appear in the market. Coaxial transducers are closer to a true point source compared to separate woofer/tweeter transducer combinations. This brings some advantages with regard to physical time alignment so that the sound source stays time coherent at any listening angle. The new Coaxial Satori MT19CP-8 is the company's take on a high-end coaxial transducer that is equally suited to midrange/tweeter reproduction in a larger multi-way speaker system or as a midwoofer/tweeter for a smaller monitor-style speaker.

The 7.5" Satori MT19CP-8 coaxial driver features an Egyptian papyrus fiber cone developed and perfected in its own cone/spider facility in Indonesia, matched with a dedicated 26mm high performance soft dome tweeter. A well-engineered driver for refined home audio designs that leverages on the extensive R&D that already characterizes the Satori speaker driver range.

The larger woofer cone surface almost masks the presence



The SB Acoustics Satori TeXtreme range using thin-ply carbon diaphragms (TPCD) gradually expanded to four sizes. The biggest commercially available TPCD driver using the increasingly popular technology, the SB Acoustics Satori WO24TX 9.5" woofer, is available in 4Ω and 8Ω variants.

of the relatively small dome tweeter at the center, making the proprietary papyrus infused black paper cone, suspended in a soft rubber surround, its most visible frontal feature. The rubber surround is a purposefully developed low-profile triple-roll design for minimum high-frequency diffraction.

And from Indonesia we go to another renowned Asian speaker manufacturer, Tang Band Speaker, also known as TB Speaker, established in Taipei, Taiwan in 1996. In 1999, Tang Band moved its factory to Ningbo, Zhejiang province in China, leaving only its sales and part of its R&D activities in Taiwan. The company, known for its unique and creative transducer designs, was strongly affected by the COVID global pandemic in terms of its manufacturing output, but not so much in its already intense R&D activities. The immediate result was visible in 2022 with the introduction of the latest TB Speaker 75mm (3") soft dome midrange driver, the 75-1558SH.

This driver, already characterized in *Voice Coil*'s Test Bench, stands out for its coated cloth diaphragm and surround, voice coil wound with round copper wire on a black anodized vented aluminum former with magnetic fluid in the gap area.

Tang Band has added a double rear back chamber design, allowing the lower frequency to go down to 260Hz, with a recommended crossover point around 520Hz. The steel cup has



The MT19CP-8 is the newest addition to the Satori family and is the first coaxial driver to appear in the Satori line, which currently consists of a wide range of midwoofer, midrange, and tweeter transducers.



The Tang Band 75-1558SH 75mm soft dome midrange features a large neodymium motor system with a low carbon content steel return cup with a black emissive coating that also incorporates a distortion reducing shorting ring.

a foam-damped vent that loads into a shaped and 100% polyester-filled injection-molded back enclosure. The unique soft dome design also uses a different glue method to bond the soft cloth, making the overall sound much softer and more transparent with an improved midrange. Other features include 90dB 1W/1m sensitivity, an injection-molded faceplate, gold-plated terminals, and 25W rated power handing (200W on program material).

Tang Band has greatly expanded its product portfolio over the years, and recently, also developed a series of small speaker drivers and innovative Sound Modules, significantly expanding its OEM/ODM activities. No doubt, its standard portfolio of products available for retail benefits greatly from that experience, and is reflected by very unique products such as the latest W4-2356 driver.

This full-range driver is built on a proprietary and patented injection-molded frame that allows the voice coil not to be centered in the 4" polypropylene cone but offset by about a half-inch. The aim for this original off-axis design is to defeat concentric cone modes, a well-known source of coloration in round voice coil transducers. In its literature, Tang Band mentions "diffraction," which is a term associated with bending of waves around the corners of a speaker cabinet, eventually turning it into a secondary source—and the company effectively measures the benefits of the W4-2356 driver against a similar-sized concentric driver assembled in the same speaker cabinet.

The W4-2356 driver leverages the injection-molded cone suspended by a Santoprene surround and a cloth spider, which combines with a three-sided aluminum phase plug, 1" voice coil, and an underhung neodymium ring magnet motor structure. The whole design sits on a patented two-piece frame that effectively helps to make the driver effective when placed on a cabinet box.



Tang Band Speaker's W4-2356 features a unique asymmetric design, where the voice coil is not centered in the 4" injection-molded polypropylene cone, but offset by about 0.5".

Glass Acoustic Diaphragm

Proving that innovation in drivers and transducers is now also coming from Asia, we need to include Glass Acoustic Innovations Technology (GAIT), the manufacturer of glass diaphragms from Taiwan with manufacturing also in China. Expanding from the early efforts of Beijing Design and Creation Electronics Co. (BDNC), founded in 2007 as the R&D arm of a Taiwan semiconductor company, the original damped glass diaphragm speakers is now continued through GAIT.

The company is now able to offer an extensive range of glass acoustic diaphragms for all types of driver form-factors and sizes, covering multiple applications, from headphones to soundbars, smart speakers, and larger-size hi-fi speakers. GAIT's comprehensive product line of glass diaphragms was on display at the 2023 High End show in Munich, where the company exhibited different models that are fully supported by established China ODMs, which should inspire confidence from any manufacturer that wants to design new projects with this proven technology.

Glass material research has already been tried and tested in speaker applications, and the results have proven effective for very thin diaphragms that can be shaped as domes or flat parts and can even be enhanced by applying color or simply being illuminated with LED lights underneath the speaker—very popular in portable lamp ambience speakers.

GAIT is also able to transform glass displays into diaphragms and implement chip-on-glass diaphragms for unique applications, and the material offers other distinctive benefits, such as being obviously waterproof, UV resistant, corrosion resistant, and easy to clean.

But apart from the obvious cosmetic considerations, GAIT has shown in Munich that using glass diaphragms is truly a good idea



A range of glass diaphragms for speakers from GAIT. For designers, a glass diaphragm is a canvas waiting to be painted. From incorporating various colors, to integrated LED light designs, to the ability to be molded into any shape or size, glass diaphragms are elegantly morphed into any application necessary, waterproof by nature, environmentally friendly, and durable for any occasion.

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The physical properties of glass, such as its high mechanical strength and fast sound propagation speed, make it an ideal material for acoustic diaphragms. GAIT glass speakers can respond to a full range of frequencies, with optimized damping characteristics. They are extremely lightweight and high-strength.

for high-quality results. The prototype soundbar was particularly impressive in Dolby Atmos, but so were the tower speakers that played throughout most of the demonstration in Munich—where the ability to impress visitors is always a good sign. The reproduction from the glass diaphragm speakers was not only clear and detailed, but bass was also fast, impressive, and distortion-free in the complete range.

As GAIT demonstrated, glass diaphragms offer a frequency band upgrade and optimized damping that is the result of a refined technology. GAIT selects glass material from the world's leading manufacturers and processes it using its advanced ultra-thin glass processing technology, which allows diaphragms thinner than 0.4mm in many different shapes and with different levels of rigidity. Its material properties are highly beneficial in transducer applications, with a very pure, clear sound signature. And yes, although "glass" is the core description the technology allows the creation of flexible diaphragms and extreme durability characteristics. As GAIT also highlights, glass is made from natural mineral sand, which is completely recyclable and environmentally friendly.

Currently, GAIT offers glass speakers in rectangle shapes, ideal for woofers in soundbars, or a range of full range and tweeters in all sizes. GAIT uses glass from Corning, NEG, and Schott to ensure the highest level of quality. The technology has been applied to microspeakers in laptops, and even headphone drivers from 10mm to 50mm and renowned ODMs such as Eastech or Meiloon have experience with the GAIT technology in drivers from 1" to 3", or a 1.5" super tweeter with a frequency response up to 55kHz.

Unique Speakers Made in the US

Even though manufacturing moved in large part to Asia, there are still speaker companies in the US. The Minneapolis Speaker Company, today MISCO Speakers, has expanded significantly over the past two decades to become a US-based audio solutions manufacturer with a large portfolio of OEM/ODM speaker solutions. In business since 1949, the St. Paul, MN-based company is a respected loudspeaker manufacturer with a range of specialized brands that includes the expanding Oaktron by MISCO, following



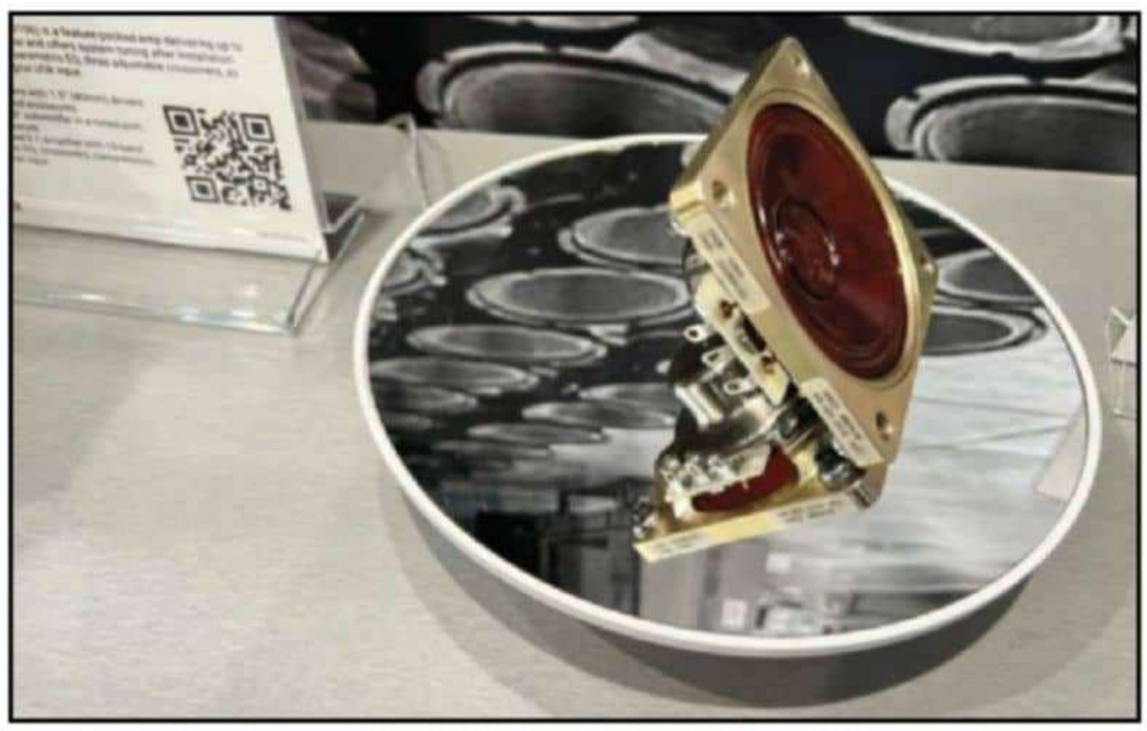
Dan Digre, president of MISCO, holding the MISCO speaker prototype developed for NASA's Orion spacecraft.

the acquisition of the brand and its IP in 2016. MISCO also created the Bold North Audio brand of accurate and thoroughly engineered audio transducers for the most demanding applications.

MISCO is also an experienced and successful source of turnkey audio solutions for a wide variety of markets, being able to completely design and manufacture custom audio products for aerospace, home audio, military, commercial, professional audio, and many other industries.

We recently met the company at the InfoComm 2023 show in Orlando, FL, where MISCO displayed a wide array of products from its different brands, while also showcasing for the first time in public a singular product that demonstrates well its unique capabilities: a speaker made for NASA!

Those in the market for high-reliability speakers could do well to compare with some of NASA's requirements for speakers going into space. Since 2017, MISCO has been working on the speakers



The Orion speaker has two identical transducers in it for redundancy purposes because no one wants to take any chances in space. Each transducer uses a powerful neodymium magnet to optimize speaker output and help reduce size and weight.

for the Orion spacecraft and its capsule that will carry humans further into space than they have ever traveled before, bringing people back to the Moon, and, possibly one day, to Mars.

Seeing the unique phenolic-resin driver on display, with its shiny gold and chrome frame, should be a clear indication of MISCO's expertise and customized capabilities.

As Rich Field, Chief Engineer at MISCO, who led the Orion Speaker Prototype project details, this speaker needs to withstand 4,000Gs (!) and it must withstand that force along multiple axes and still be able to perform during and after that impact. "It's an example of the 'margin on margin' thinking that is critical to assure success and safety of each NASA mission."

MISCO prides itself on designing robust speaker drivers and offering extensive testing capabilities, but to design and test something able to support the extremes of a rocket launch is at a whole other level. The resulting speaker was able to withstand these NASA tests and continue to work as specified—but that is just the start.

The MISCO speaker needed to solve the unique acoustical challenges of working in space, where variations in air pressure that generate the sound waves are completely different. This NASA speaker needs to be able to produce sufficient decibel sound pressure level in an atmosphere of 9 PSI (minimum specified) just as well as it does in 14.7 PSI (the air pressure at sea level on Earth). Extensive modeling and calculations were done at MISCO to show the chosen speaker design would in fact work and then was validated with measurements in pressure chambers.

The choice of materials was another challenge for the project, requiring MISCO to create a solution for all shock and temperature requirements, plus the general requirement that the speaker operate "in all conditions and all environments." MISCO was selected for the Orion project because of its background working on speakers for military applications, and because its Oaktron division had previously developed the speakers for the International Space Station (ISS).

The Orion will be lifted into space for the first time in 2025, atop the Artemis I rocket, the most powerful NASA has ever used.

Resources

Composite Sound | www.composite-sound.com

Glass Acoustic Innovations Technology Co., Ltd. | www.gai-tech.com

Minneapolis Speaker Co., Inc. (MISCO) | www.miscospeakers.com

Purifi Audio | www.purifi-audio.com

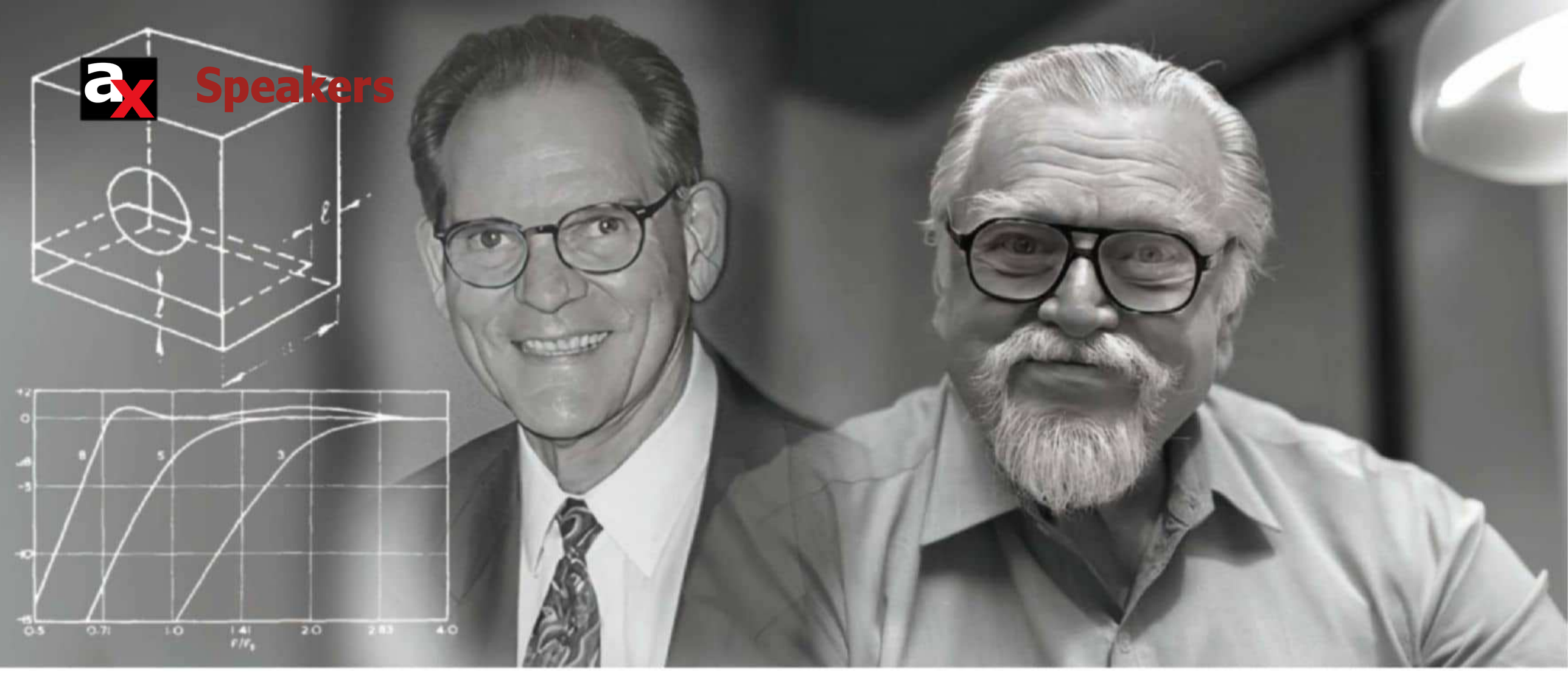
SB Acoustics | www.sbacoustics.com

Scan-Speak | www.scan-speak.dk

SEAS Fabrikker AS | https://seas.no

Tang Band Speaker | www.tb-speaker.com

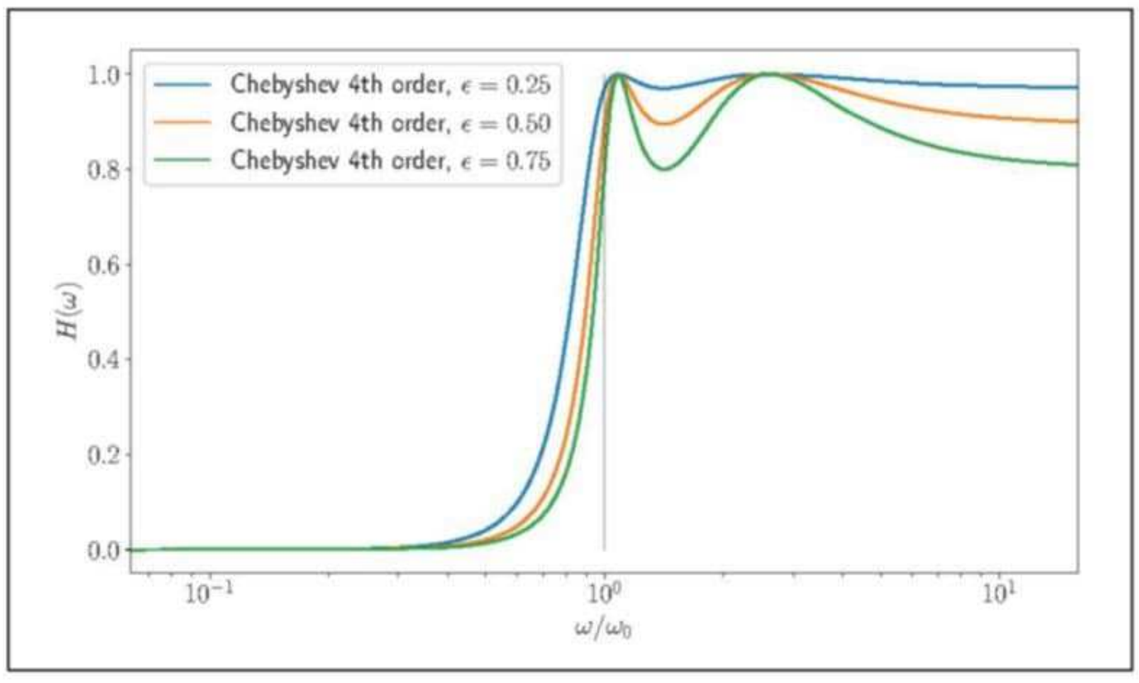




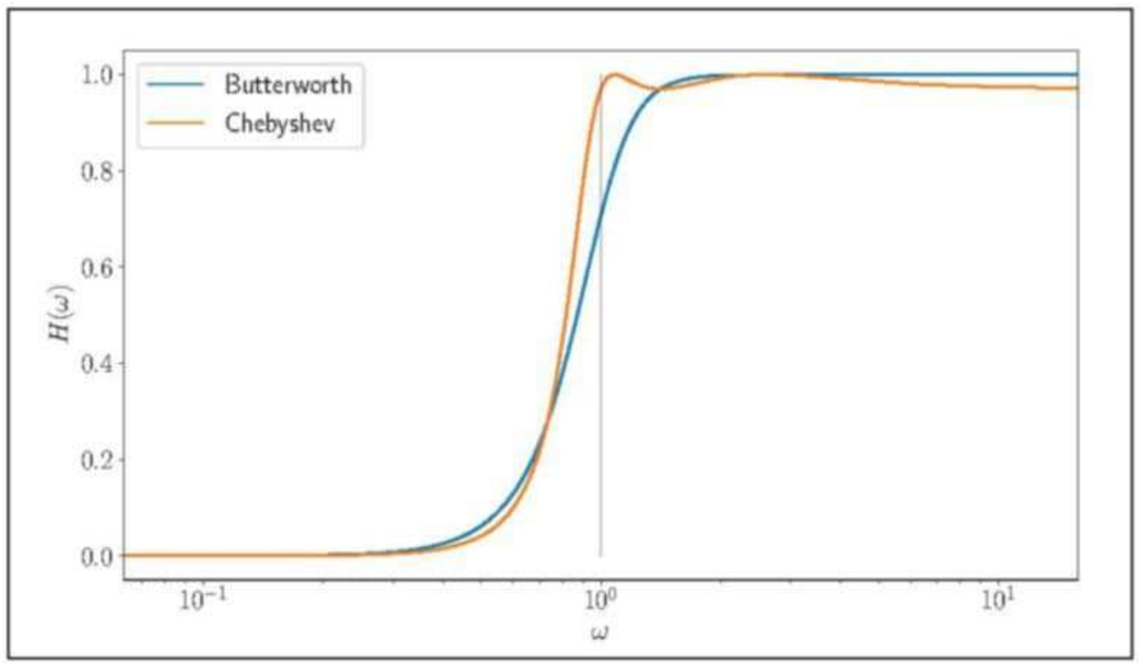
Loudspeaker Bass Reflex Alignments

By
Claus Futtrup

The use of standard filter alignments represents a foundational concept in the loudspeaker industry, with an impact nearly as significant as the invention of the electro-dynamic loudspeaker principle itself. This article describes the evolution of enclosure designs based on this electrical filter theory, including the concept of an alignment chart.



Normalized fourth-order Chebyshev high-pass filter frequency response with different ripple settings



The Butterworth and Chebyshev filter functions

Bass reflex alignments are a vast topic with a history that fundamentally starts in 1961 when Albert Neville Thiele published his seminal articles “Loudspeakers in Vented Boxes” Part I and II. These were presented at the 1961 Institute of Radio Engineers (IRE) convention in Sydney, Australia, and later published in the proceedings.

To the best of my knowledge Thiele invented the concept of alignments in relation to loudspeakers. So, what is an “alignment?” Inspired by RF terminology, Thiele’s idea was to align a loudspeaker bass reflex vented box high-pass response with an equivalent high-pass filter function. So, the word “alignment” comes from this idea that we align the response with a filter function—the advantage being that the behavior of a filter function was well known at the time. Prior to Thiele’s idea of aligning the response with a filter function, bass reflex was designed based on good practice rule of thumb, and/or possibly an iterative process. Generally speaking, we never really knew what we were going to get.

Thiele’s idea changed all that. The way I see it, it was a stroke of genius on his part and indeed he was a genius, as his later work demonstrates. So, the purpose of this design method where we align the bass reflex response with a filter function, is to design a loudspeaker predicting what we will get in regard to frequency response, time response, and more, even before building the box.

At this stage we could ask, “OK, but what filter function should we use?”

Chebyshev Filter Function

Let’s look at the known filter functions at the time, in 1961. But first we need to mention the Chebyshev filter function, so named after Pafnuty Chebyshev. He was a Russian born in 1821 and he

died in 1894, long before we were doing loudspeaker designs. But he did some of the underlying math which then came into use here.

What characterizes a Chebyshev filter function or bass reflex response is that we allow the response to have some ripple. We have a response that goes down in frequency, and we allow some ripple before roll-off. And we extend that roll-off as far down in frequency as we can, while we make the ripples equal in height so overall they are minimized. And we can regulate the ripple to our desire, within limits. We can choose, for example, half a decibel ripple, which some people would say is almost unnoticeable, or one decibe. It's up to us. Obviously, there are an infinite number of Chebyshev filters from which to choose, and in relation to loudspeakers this was first described by F. J. van Leeuwen in 1956.

Van Leeuwen published an article that was 40 pages long. It was published in Dutch and in a scientific magazine named the *Netherlands Radiogenootschap*. Since I don't read Dutch, it was very difficult for me to muddle my way through it, but he writes a lot of nice math, including how to measure the driver parameters where he uses the Philips-type 9710/M01 driver as an example.

With the high Qt of that Philips driver, he describes two types of equal ripple responses that he can obtain with this driver. He never mentions Chebyshev explicitly and he isn't clear about aligning his bass reflex response with a filter function.

If we take this Chebyshev alignment and instead we set the ripple to zero, it becomes a maximally flat magnitude filter, which is also named the Butterworth response.

Butterworth and Bessel Filters

And that brings us to the next filter function: This one is named after a mathematician named Stephen Butterworth and he was born in 1885 and he died in 1958. He described his Butterworth response around 1930.

This filter offers the sharpest roll-off; the most extended bandwidth, and therefore gives us the sharpest knee point without any ripple in the passband. In relation to bass reflex loudspeakers, it's a discrete alignment, which means we need to select a specific driver with a specific Qt value, and tune that with a specific box and port, and then we can obtain this discrete alignment.

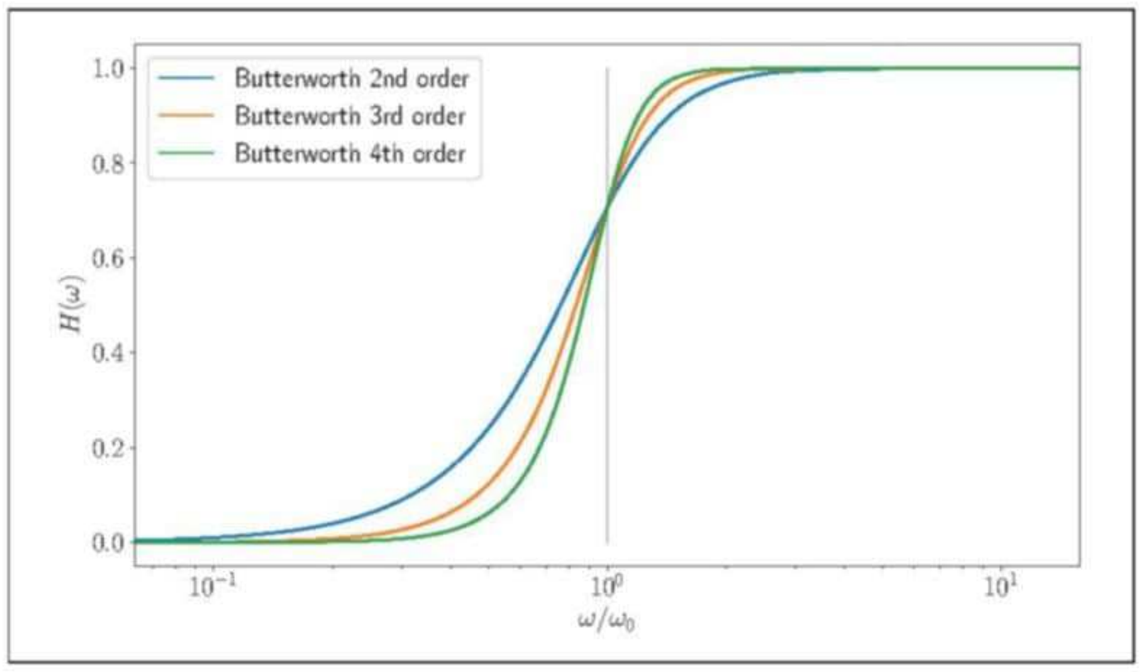
We go from bass reflex and fourth-order responses; instead we look at a second-order filter function. The knee point Q—the roll-off—is a Q of 0.707 or $1/\sqrt{2}$. But for a fourth-order response it is not defined as a single Q value. Although on the Internet we sometimes see the Q equal 0.707 mentioned for a fourth-order system, it's not really like that—don't trust everything you read on the Internet.

If these filter functions were described before loudspeaker alignments were invented, who did the work related to loudspeakers? I believe that Thiele's article series is the first time where we see this correlation.

Another one is called the Bessel filter function. The Bessel filter is named after Friedrich Wilhelm Bessel, a German mathematician and physicist born in 1784, who died in 1846. Although it was already known at the time, it wasn't mentioned by Thiele, but Bessel laid out the math that described the filters with the fastest settling time. I'm sure he didn't have any clue about the particular application to loudspeakers, but he did a lot of nice math back in his day.

What is special about the Bessel filter is that, in the time domain, when we do a step response on a filter like that, it will overshoot a little bit to settle faster. So, we get the fastest settling time.

The practical application was worked out by W. E. Thomson in 1949, in a scientific article titled "Delay Networks Having



Normalized Butterworth maximally flat high-pass filter frequency responses of second order (closed box), third order, and fourth order (bass reflex box)



This is an example of a Philips 9710 M1 full-range driver.



Stephen Butterworth worked for several years at the National Physical Laboratory (UK), where he conducted theoretical and experimental work for the determination of standards of electrical inductance and analyzed the electromagnetic field around submarine cables.



Friedrich Wilhelm Bessel (1784 -1846) was a German astronomer, mathematician, physicist, and geodesist.

Maximally Flat Frequency Characteristics," where he described this filter function applied to delay lines.

Thomson worked at the post office research station, and he was active as late as 1977. We can imagine that when we develop a delay line, we would like it to have a flat delay, and a flat group delay—not a minimum group delay—and that's why he used these Bessel filters, which sometimes are also called Bessel Thomson filters.

Again, second-order filters are described as having a Q value of 0.577—so a lower Q value than a Butterworth—or we could also say $1/\sqrt{3}$, but again for fourth-order response function the Q is not uniquely defined.

Critically Damped Filter Function

The last one I would like to mention is the Critically Damped Filter Function. The second order has a Q of 0.5, which means no overshoot in the time domain, but it suffers from very early roll-off when looking at the frequency domain. A Gaussian filter is designed to give no overshoot to a step input function, while minimizing the rise and fall time. In my opinion there is no point in going lower than a Q of 0.5, which is also $1/\sqrt{4}$.

This is just to make it clear that there is a relationship with the square roots. Obviously 0.5 is an exact number, so we don't need to calculate square roots to get all the decimals.

It is only recently that a work colleague, Jacob Eriksen, reminded me that fourth-order responses do not have a Q value. The concept of Q values belongs to second-order filters.

If we want to we can describe a fourth-order filter with its polynomial—we can describe it as two second-order polynomials that then can be multiplied or expanded into the fourth-order polynomial with which we're working. Or vice versa, when we have the fourth order, we can find two second orders that describe it.

Each of these second-order filters that makes up our fourth-order response, have a Q value that we can find in the polynomial. Therefore, when we see anyone mentioning the Q value of a fourth-order response, it must be some sort of composite of those two Q values, but that's not uniquely defined. Meaning we can't just add them up or multiply them or anything like that... So maybe it's better not to talk about Q values when we talk about a fourth-order response.



From left to right are Paul W. Klipsch, J. Robert Ashley, A.N. Thiele, J.E. Benson, and Richard Small.

2nd-order filter has a Q value:

$$H_2(\omega) = \frac{s^2}{s^2 + s/Q + 1}$$

4th-order filter is **not** characterized by a single Q value

$$H_4(\omega) = \frac{s^4}{(s^2 + s/Q_1 + 1)(s^2 + s/Q_2 + 1)}$$

A second-order filter has a Q value, but a fourth-order filter is not characterized by a single Q value.



The author Claus Futtrup (left) and Jann U. Evers (right) are pictured with Neville Thiele during his visit to Scan-Speak in 2009.

Let's Talk About Bass Reflex

Albert Neville Thiele was born in 1920 and he died in 2012. He came up with this idea to align bass reflex loudspeaker response functions with equivalent filter functions. Computers were not readily available yet, so the concept of alignment was very helpful in determining our end goal. Thiele described fourth-, fifth-, and sixth-order Butterworth bass reflex responses (the higher orders required auxiliary electronic filtering), as well as Chebyshev responses with different amounts of ripple. Plus, he coined the term sub-Chebyshev alignment for lower Qt drivers, and the quasi-Butterworth's response, or quasi-third-order Butterworth's response, which as far as I know was a new concept at the time.

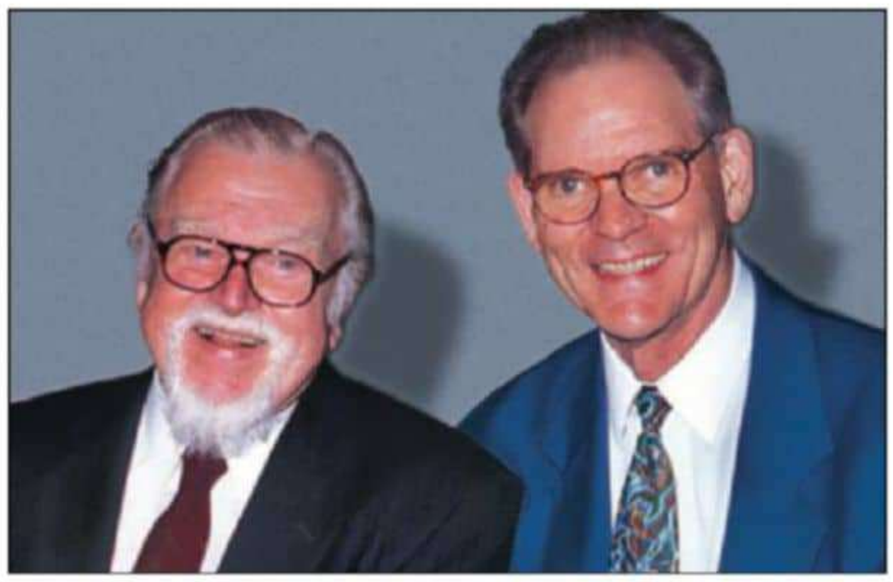
This response, named QB3, allows to tweak a box and a driver with a given port tuning to have a soft roll-off, eventually becoming a fourth order, of course, because that's how a bass reflex works... It's always a fourth order in the end. But it's a soft roll-off and it has this tendency to be more like a third order.

With this concept we could choose a discrete, specific driver, box, and port tuning, or we could select from a range of drivers with some Q values, and then of course we would have to pick the right box size and port tuning. But suddenly we have the flexibility to choose from a range of drivers, or drivers with a range of Qt values, and that's quite nice when we have a degree of flexibility with regard to the choice of the driver we want to use.

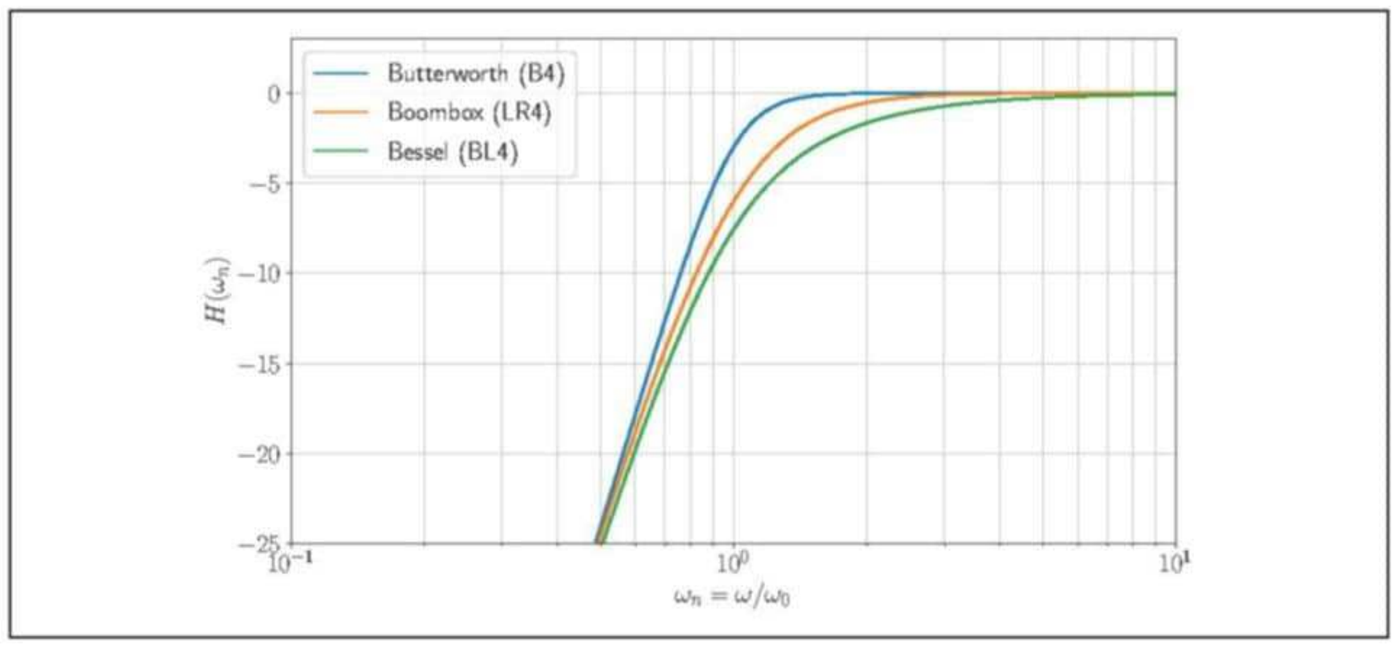
Thiele's two-part article is so important that it was later republished in the *Journal of the Audio Engineering Society* in 1971, 10 years after being published for the first time.

Whereas the Chebyshev alignments are an entire family of alignments, and we have the quasi-Butterworth being suitable for a range of drivers, the fourth-order Butterworth, the B4 alignment, exists only for a specific driver Qt, which is around 0.4 or if we operate in a lossless design 0.38, and it's therefore a discrete alignment.

The Bessel or Bessel Thomson alignment is also a discrete alignment, which only exists for a specific driver to a Qt around 0.3 or—this is more or less lossless but if we add a little bit of a loss—we can say 0.32 or whereabouts (in practice there's always some losses).



Neville Thiele and Richard Small



Butterworth fourth order, (Sub-) Boombox in the Linkwitz-Riley case fourth order, and Bessel fourth order normalized frequency responses (decibel on the y-axis)



The QA403 Audio Analyzer...$599

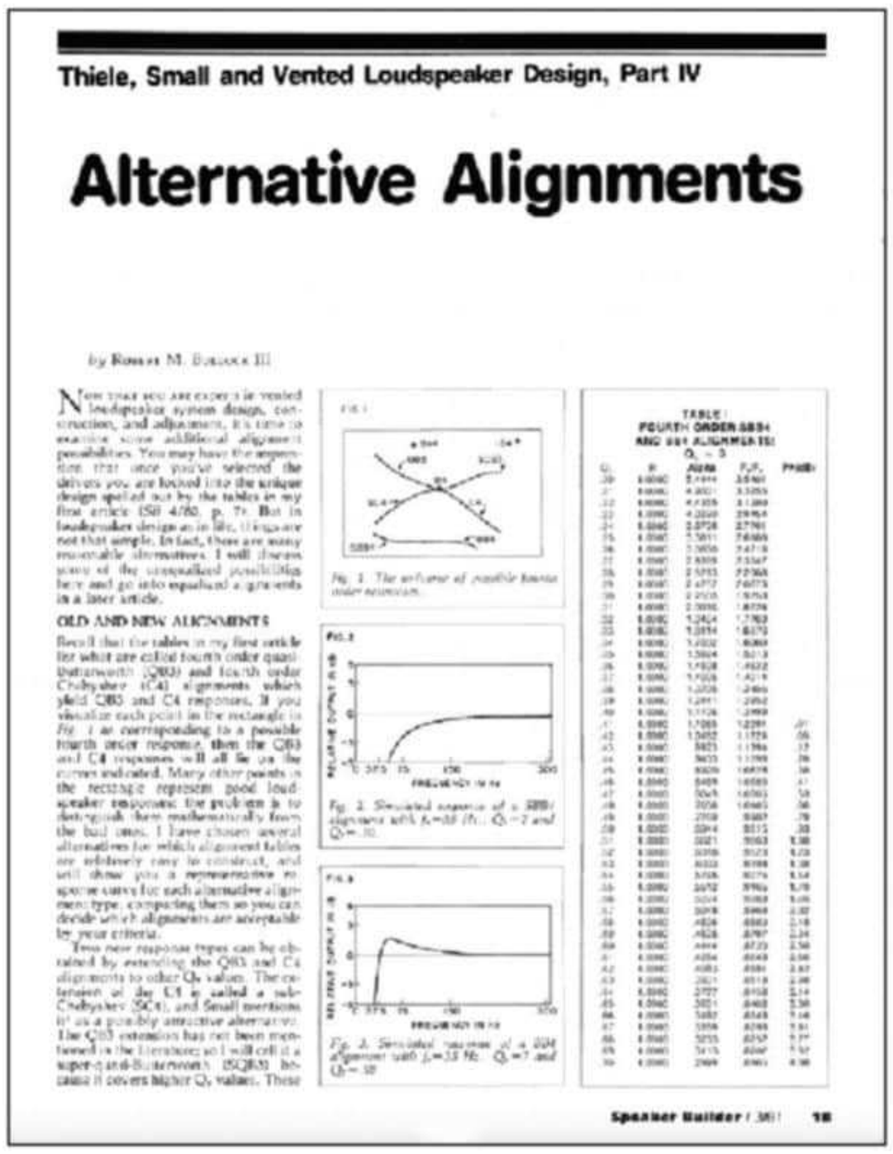
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The first page of Robert Bullock's article as shown in *Speaker Builder* magazine in 1981.



Thiele, Small and Vented Loudspeaker Design, Part IV

Alternative Alignments

by Robert M. Bullock III

Now that you are experts in vented loudspeaker system design, construction, and adjustment, it's time to examine some additional alignment possibilities. You may have the impression that once you've selected the drivers you are locked into the unique design spelled out by the tables in my first article (SB 4/80, p. 7). But in loudspeaker design as in life, things are not that simple. In fact, there are many reasonable alternatives. I will discuss some of the unequalled possibilities here and go into equalized alignments in a later article.

OLD AND NEW ALIGNMENTS

Recall that the tables in my first article list what are called fourth order quasi-Butterworth (QB3) and fourth order Chebyshev (C4) alignments which yield QB3 and C4 responses. If you visualize each point in the rectangle in Fig. 1 as corresponding to a possible fourth order response, then the QB3 and C4 responses will all lie on the curves indicated. Many other points in the rectangle represent good loudspeaker responses; the problem is to distinguish them mathematically from the bad ones. I have chosen several alternatives for which alignment tables are relatively easy to construct, and will show you a representative response curve for each alternative alignment type, comparing them so you can decide which alignments are acceptable by your criteria.

Two new response types can be obtained by extending the QB3 and C4 alignments to other Q_t values. The extension of the C4 is called a sub-Chebyshev (SC4), and Small mentions it as a possibly attractive alternative. The QB3 extension has not been mentioned in the literature, so I will call it a super-quasi-Butterworth (SQB3) because it covers higher Q_t values. These

Whenever a driver Qt deviates a bit from these numbers, we talk about an approximation to that alignment, which is perfectly fine if we are within 5% or so.

Richard H. Small

Richard H. Small is from San Diego, CA, in the US, but he went to Australia, where he got a part-time job at The University of Sydney, as a teaching assistant in the electronics lab. He wrote his PhD thesis on loudspeakers, which came out in 1972, when he stumbled on the work by Thiele [2].

Richard Small wrote articles for the *Journal of the Audio Engineering Society* starting in 1969—as far as I remember and continuing with a four-part series on vented box loudspeakers in 1973. These articles created a lot of interest across the world and essentially, within a few years, loudspeaker manufacturers were documenting their drivers with Thiele-Small parameters. This allowed engineers to sketch their designs prior to buying the driver and identify which driver was most suitable for their desired target response.

Thiele did the groundwork with the idea of alignments and Small's work put it into practical use worldwide. Thiele once visited the Scan-Speak facilities in 2009, and he told me that he had decided to publish his two seminal articles with an Australian publisher to give Australians a leading edge.

Since Thiele's papers were largely ignored until the work by Small this didn't work out. I also have a suspicion that the reason could be a different one. Maybe it wasn't so clear-cut who to use to publish papers, and there were more options back in the day.

My colleague at Scan-Speak in Denmark, Knud Thorborg, confirmed that he read the articles by Thiele shortly after they were published, in 1962. Thorborg [3] once told me that they received Thiele's original IRE articles at the Peerless Factory on Amager in Copenhagen. The engineers at Peerless were quite interested, and they tested the theory but for some reason it didn't work out.

Thorborg speculates that the reason it didn't work for Peerless drivers around 1962 was because they were using open weave dust caps. Meaning that air could penetrate through the dust cap, creating a lossy design. And indeed, at the time, before hi-fi, the dust caps were there just to prevent dust from entering the magnet system! Later, dust caps were sealed, and air pressure had to be relieved from between the dust cap and the pole piece underneath, which at Peerless they do by putting holes in the diaphragm. Other designers vent through the magnet system.

References

[1] Speakerbench is a web-based application for loudspeaker design with an emphasis on an advanced transducer model. It has been available since March 2020 at speakerbench.com and requires only a web browser to use. Speakerbench simulations are more accurate than what can be accomplished with the classic Thiele-Small approach. An online manual provides complete references to the research.

[2] *Voice Coil* Interviews Dr. Richard H. Small, *Voice Coil*, 2006. https://audioxpress.com/article/voice-coil-interviews-dr-richard-h-small

[3] K. Thorborg, *Traditional and Advanced Models for the Dynamic Loudspeaker,* Scan-Speak, Videbæk, Denmark, kt@scan-speak.dk https://cfuttrup.com/Thorborg_31.pdf

[4] J. W. Hoge, "A New Set of Vented Loudspeaker Alignments," *Journal of the Audio Engineering Society* (JAES) Volume 25, Issue 6 pp. 391-393; June 1977, www.aes.org/e-lib/browse.cfm?elib=3364

[5] R. M. Bullock III, "Thiele, Small and Vented Loudspeaker Design, Part IV," *Speaker Builder,* 3/1981

[6] WinISD is a freeware speaker design software for Windows, http://www.linearteam.org

[7] Speakerbench, https://speakerbench.com

[8] J. Candy and C. Futtrup, "An Added-Mass Measurement Technique for Transducer Parameter Estimation," *Journal of the Audio Engineering Society* (JAES), Volume 65, Issue 12, pp. 1005-1016; December 2017, www.aes.org/e-lib/browse.cfm?elib=19366

[9] C. Futtrup and J. Candy, "Speakerbench," *Loudspeaker Industry Sourcebook* (LIS) 2020, page 124-131, www.loudspeakerindustrysourcebook.com/articles/speakerbench

[10] C. Futtrup and J. Candy, "Parameter Estimation and Box Simulation with Speakerbench (Part 1 and 2)", *audioXpress*, September/October 2021. https://audioxpress.com/article/project-articles-parameter-estimation-and-box-simulation-with-speakerbench-part-1-model

There could be other reasons why there were discrepancies from the theory by Thiele, but I think this is the real reason why Thiele's work didn't pick up from the beginning. To put it simply, the world wasn't ready for it yet.

Fast forward to the 1970s and a third name sometimes shows up, which I would like to mention and that's John Ernest Benson (1911-1989). He was also associated with Sydney University, and he wrote a three-part series of articles printed in 1968, 1971 and 1972, which became an entire book called *Theory and Design of Loudspeaker Enclosures*.

Benson was the examiner of Richard Small's PhD thesis and Benson's work was quite comprehensive but maybe a bit pedantic in his approach. Personally, I think it was Small's work that presented a design method that was accepted worldwide.

Linkwitz-Riley Filter Function

In the mid-1970s Siegfrid H. Linkwitz described an electrical filter named the Linkwitz-Riley filter, named after himself and a colleague, Russ Riley. This filter function applies two second-order Butterworth responses in series, so-called cascaded filters—as described previously, taking two polynomials, each of those having a Q of 0.707, so they are Butterworth responses, and they are identical.

This article was published in February 1976, but the point to retain is that there is no bass reflex alignment named "Linkwitz-Riley." Also, when it comes to bass reflex alignment, I would like to mention William Hoge because he described what he called the Boom Box alignment or the BB4 alignment. He also cascaded two identical second-order filters and Hoge's description of the Boom Box alignment is from October 1976—eight months later [4].

Hoge doesn't work with low Qt drivers... In fact, he proposed the Boom Box for high Qt instrument speakers, and therefore this response will always have a peak before it rolls off.

The idea to extend the Boom Box alignment for lower Qt drivers is the work of Robert M. Bullock, III and his extension is called the Sub-Boom Box or SBB4 alignment. He described it in an article published in *Speaker Builder* magazine [5], in 1981. In principle his reference was just to explain that good bass reflex configurations exist that are not described by classical alignments.

If we choose our parameters correctly, the Boom Box/Sub-Boom Box alignment will be the bass reflex equivalent of a Linkwitz-Riley filter. We get in that ballpark if we choose the driver Qt around 0.36, or 0.37, and it's in a design including losses.

In similar ways, Bullock also shows the SC4—the so-called sub-Chebyshev fourth-order, which was previously described by Thiele, as well as the super-QB3—where "super" is not "super good" but it means that the Qt of the driver is higher than what we would choose for a B4 alignment. So, we're above a Qt of 0.4. He extends this QB3 to drivers of a higher Q value, and it becomes this SQB3 or super-QB3. And because the Q value is like it is, it's

Speakerbench YouTube Channel

The Speakerbench YouTube channel at youtube.com/@speakerbench contains videos created by engineers Claus Futtrup and Jeff Candy related to Speakerbench, a web-based application for designing and engineering loudspeakers. In addition to instructional videos on driver measurement and use of the Speakerbench web app, their videos cover educational and historical perspectives on loudspeaker design, which created the foundation for this article.

Two released videos describe the principle of loudspeaker alignments as pioneered by Neville Thiele. These were widely used in the industry from the 1970s until modern computer software took over. The videos, and now this article, may be considered a tribute to Thiele for his invaluable contribution to how we design loudspeakers.

Direct links to the two videos:

https://youtu.be/xCVSRBdTBWM

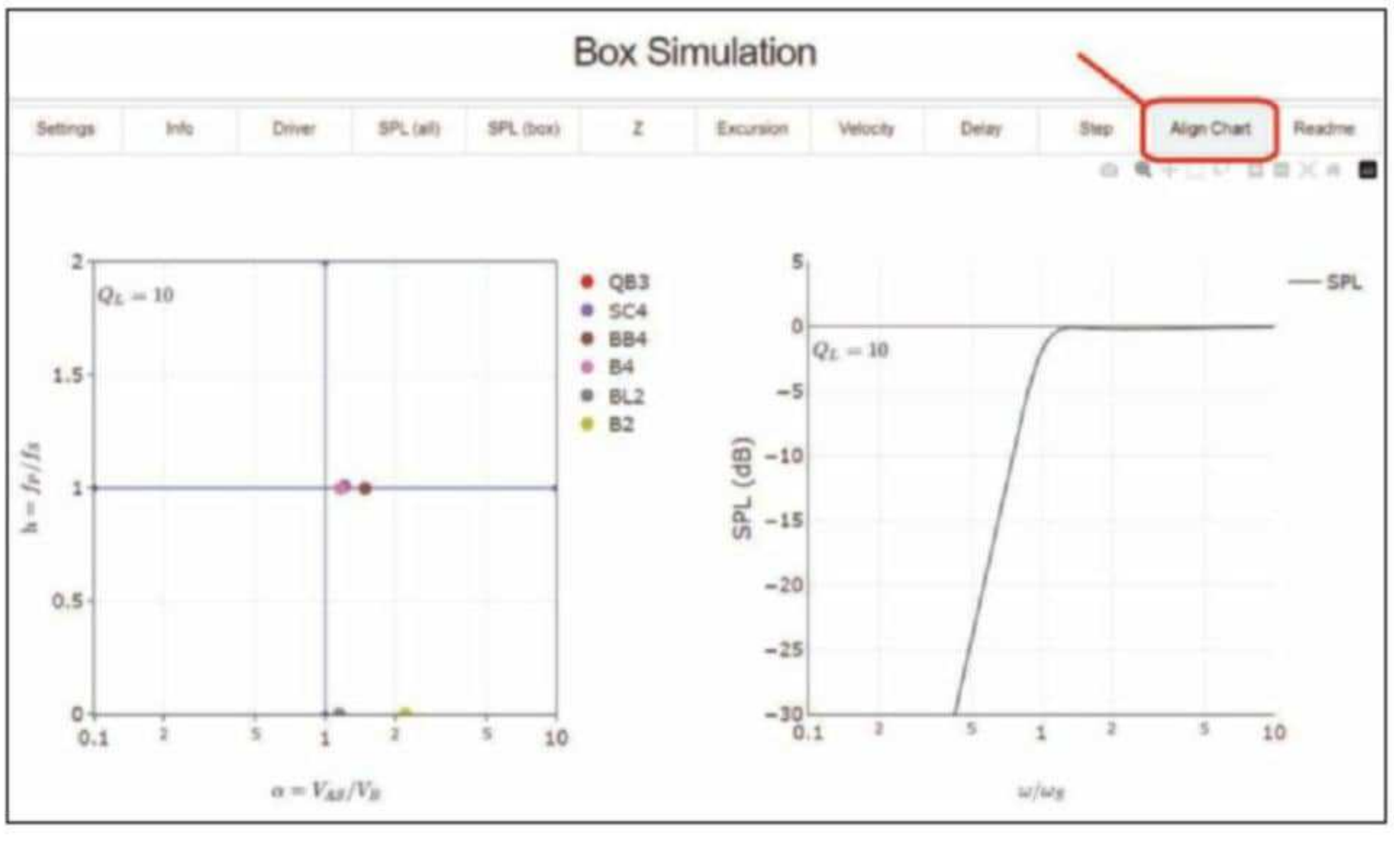
https://youtu.be/J0wvMkreUhY



not linear in frequency response but it has a peak in the response.

Regarding these alignments, I once attended a practical demonstration by Ulrik Schmidt, who worked at Scan-Speak at the time (now with Danesian Audio/SB Acoustics), and while I was working at Dynaudio, around 2001. It was a presentation for the Danish Electroacoustic Fellowship (DEF) and there were many people in the room. Ulrik had designed three boxes, each made with a similar driver but tweaked for different Qt values, so that he could align them exactly with a C4 alignment, a Chebyshev alignment; the B4, which is this discrete alignment where we need a very specific Qt; and the Bessel alignment.

With this demonstration, we could clearly hear the difference in sound quality. It became immediately apparent that the C4 alignment sounds “dark” and “dull” and lacks detail, whereas the Bessel alignment has the best bass performance of the three. It doesn’t go the deepest, but it sounds the best, it’s very impulsive and musical sounding. So, thank you Ulrik for that one - it was a great demonstration!



The Alignments window in the Speakerbench loudspeaker design web-based app

Software and Alignments

When software became available in the 1990s and into the early 2000s, loudspeaker designer software presented us with a drop-down box where we could select available alignments for whatever driver we had in the software. Or we could choose a user-defined alternative on our own.

In my opinion, this user defined box wasn’t useful because I found it difficult to understand where I was with regard to alignments and to

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Polypropylene Film & Foil – 0.01uf to 22uf, 100Vdc to 1200Vdc
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my goals. So, when Juha Hartikainen and Janne Ahonen [6] were programming WinISD—this was around 2001 maybe 2002—I proposed they should present the available bass reflex alignment in a two-dimensional chart, instead of this drop-down box. Because in a two-dimensional chart we would then have alpha on one axis and "h" on the other—alpha is a ratio of the Vas of the driver and the box volume, and the h-value is a ratio of the port tuning frequency versus our driver's resonance frequency in free air. I believed this would make it clear to see where we were. But they decided not to do it.

Therefore, I think Speakerbench [7,8,9,10] is the first software that gives us this 2D chart of alignments, which for me gives us an intuitive feel for where we are heading. Particularly when we do something non-conventional in the box tuning, something that doesn't fit an exact alignment with a filter function, and we can see approximately where we are between alignments. Readers can check out the alignment chart available in the Speakerbench box simulator.

Conclusion

The concept of alignment was a major drive for speaker builders some decades ago. But with help from our personal computers, we can now calculate the results in the form of frequency response, impedance, time response, group delay, port output, air velocity in the port, and so on...

Maybe the concept of "alignment" with the filter function is no longer needed in principle. We can just look at all our graphs and see exactly how a particular box is behaving.

But, in my opinion, the concept of alignment remains useful as an anchor for what we do with the box tuning, even if the Thiele-Small as an approach is not the most accurate way to design loudspeakers, considering that we now have better loudspeaker models available.

Yet, I still think it's a good way to have a good sense of where we are in our design.

About the Author



Claus Futtrup received his MSc in ME from Aalborg University in 1997 and has worked in the loudspeaker industry since. From 1997 to 2006, he worked at Dynaudio A/S first as an R&D Engineer, designing loudspeaker boxes and later as a System Engineer for automotive. From 2006 through 2007, he was employed as a Transducer Design Engineer at Tymphany Denmark and, in 2008-2013 he was R&D Manager at Scan-Speak. From 2013 to 2015 he worked as Technical Sales Manager for SEAS, Norway, and in 2015 he was promoted to Chief Technical Officer. From August 2021, Claus moved back to Denmark and started working at DALI as manager of the acoustics team.

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By
Timothy Brault

Total Sonics, formerly dbx-tv, a business unit of THAT Corporation

dbx-tv is Evolving into Total Sonics

dbx-tv audio processing has long enabled fantastic sound quality in televisions. We've recently extended that great performance to speakers and other electronics, but now have a problem – "dbx-tv" implies our processing is only for TVs, and our technology sounds too good to keep it away from other devices!

With an eye toward our bright future, we are changing our name from dbx-tv to Total Sonics, to better reflect the broad value of our audio expertise. To appreciate that future, it might help to understand our evolution, starting from our dbx Inc. origins.



Emmy® award-winning dbx-tv processing delivers great sound in over 1 billion devices worldwide.

dbx-tv's History

In the early 1980s, the Broadcast Television Systems Committee (BTSC) solicited proposals for companding methods to ensure noise-free stereo sound in its new TV audio broadcast system. The legendary dbx Inc. responded with the "dbx-tv" noise reduction system, unanimously selected by the BTSC over competitors from Dolby, Telefunken, and CBS, and the technology was licensed to every manufacturer of TVs, VCRs, and broadcast equipment in the world.

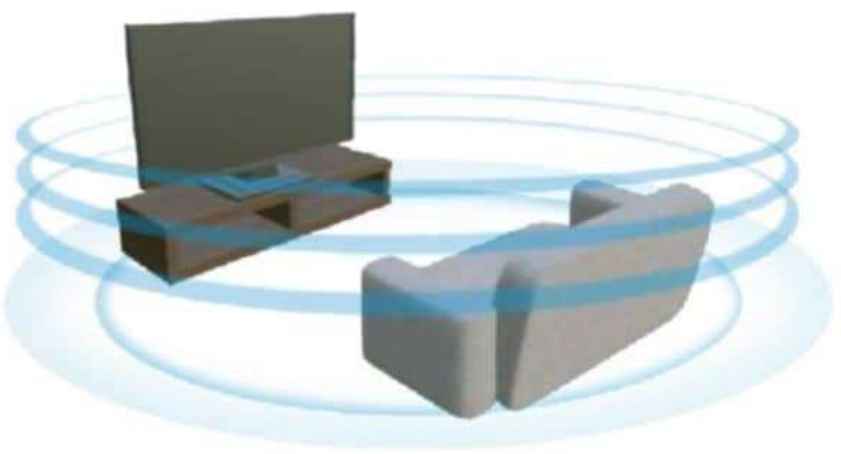
dbx Inc. was later broken up, with different entities acquiring rights to its various business units. In 1989 the unit responsible for dbx's integrated circuits and analog noise reduction boards was spun off as "THAT Corporation." THAT continues to produce best-in-class analog audio ICs and offers outside foundry services to customers looking for high-performance, medium-voltage analog applications.

THAT acquired dbx Inc.'s IP rights in 1994, launching the "dbx-tv" business unit to license technology to the TV industry. dbx-tv then fueled the digital TV revolution with digital implementations of the BTSC encoder and all other analog TV audio decoders worldwide.

Total Sonics, Total Volume®, and Total Surround® audio processing

Following the digital TV transition, we changed our focus to licensing efficient, effective, and easy-to-use audio processing and calibration solutions to improve audio experiences in the home. The first problem to tackle was obnoxiously obvious – universally scorned, too-loud commercials! World-renowned dbx-style compression, now delivered digitally by THAT, provided the perfect solution: Total Volume loudness control.

At the same time, TV set designs were trending toward slimmer cabinets and bezels, leaving no room for full-sized, front-facing speakers with proper separation. Audio quality suffered, yet design and budget constraints left manufacturers without viable hardware solutions. Processing to get the best possible sound from challenging drivers was needed, inspiring the patented Total Sonics audio processing suite.



Total Surround stabilizes then expands the sound field to create a natural, enveloping experience with consistently clear dialog.

Using clever dynamic EQ, Total Sonics processing optimizes the device's electroacoustic response to extend bass performance without creating distortion. Total Bass™ psychoacoustic bass extension, the latest addition to the Total Sonics suite, further extends the perceived frequency response of speakers beyond their physical capabilities by tricking the brain into hearing pitches not actually there. The one-two punch of Total Bass and dynamic EQ ensures low-frequency sounds are enjoyed on products of any size, from portable speakers to mobile devices.

Augmented by novel combinations of our Total Surround and Total Volume algorithms, Total Sonics solves the primary audio quality issues facing consumers.

Total Cal® Pro Automated Speaker Tuning System

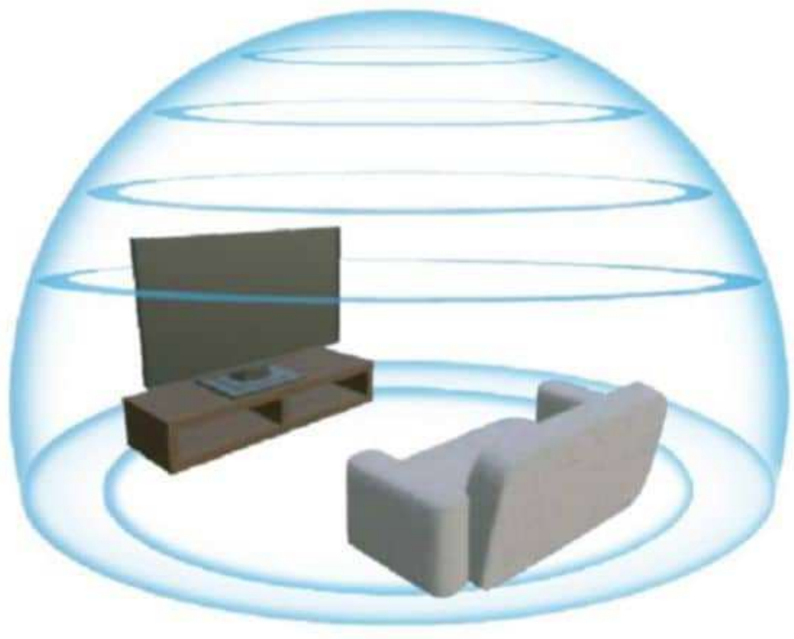
Our next challenge came directly from TV manufacturers — the time needed to tune so many unique TV models in the face of accelerated design cycles and shrinking budgets. We responded with the Total Cal Pro automated measurement and calibration system.

The genius of Total Cal Pro lies largely within its patented genetic EQ algorithm, inspired by THAT co-founder Les Tyler's co-invention of the first automatic EQ, the dbx 20/20. Armed with knowledge of the device's performance, the speedy genetic algorithm adapts Total Sonics' parameters to meet one or more frequency response targets chosen by the designer, delivering a customized tuning for each unique combination of driver, amplifier, and enclosure. This breakthrough allows manufacturers to achieve "their sound" in minutes instead of days or weeks of trial and error, an achievement which earned the 2012 IMS Research TV Innovation Award in the Audio Experience category.

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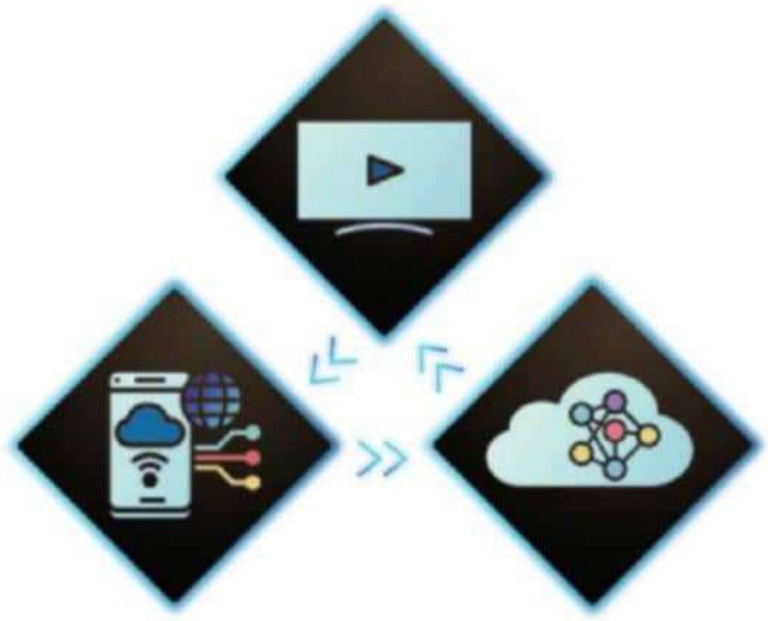
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Relying on the cloud, the mics in the customer's phone, and the efficient Total Sonics processing running in the SoC already factored into the product's BOM, Total Cal requires no additional hardware to implement—a stark departure from competitive solutions. Total Cal's low integration cost and familiar mobile interface are a significant evolution beyond traditional approaches, bringing room correction into more homes by making it accessible on a wider range of audio products.



The Total Cal in-home tuning system brings Total Cal Pro's powerful algorithms to consumers via the cloud to tackle problems created by speaker placement and room acoustics.

Evolving into Our New identity, Total Sonics

Through a combination of customer demand and our natural curiosity we've proven these innovative technologies perform well not just on TVs, but also on pro and consumer speakers, phones and tablets, computers, and other devices. Though we are incredibly proud of dbx-tv's 40-year legacy of television audio excellence and are reluctant to abandon the "dbx" name, the "tv" suffix doesn't apply outside of the TV market. In order to fulfill our vision of enabling fantastic audio experiences on more devices we must expand our reach beyond TVs and evolve into our new identity.

I am proud to re-introduce us as Total Sonics, a reflection of both our mission and our core strength — improving audio experiences in the home by creating efficient, easy-to-use audio processing and calibration software.

If your product has speakers, Total Sonics will help your design sound its best — even after the customer brings it home!

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TPCD Technology in Headphones

Engineering to Control Diaphragm Resonances

By
Martin Turesson
(Composite Sound)

Designers of conventional loudspeakers and headphones try to reduce the negative effects of diaphragm resonances but are limited in their abilities to do so. Thin-ply carbon diaphragm (TPCD) technology, pioneered by Composite Sound, uses a radically different approach to solve the problem. Rather than trying to reduce the negative effects of the resonances, it controls its causes. Composite Sound is now entering the over-ear headphone market with the TPCD diaphragm technology, enabling headphones with better sound quality, improved bass performance, and higher sensitivity.

In 2022 Composite Sound, the manufacturer and inventor of thin-ply carbon diaphragm (TPCD) technology [1], started a research project. We wanted to see what would happen if we combined two aspects of our technology—the resonance control abilities of our bigger cones and the thinness, stiffness, and lightness of our tweeter domes—having over-ear headphone diaphragms in mind.

The first step was to produce non-engineered TPCDs with uniform thickness and stiffness, or in other words how conventional diaphragms are made. These non-engineered TPCD diaphragms showed benefits over the current diaphragms—the resonances (break up) were moved to higher frequencies and their amplitudes were reduced. But the resonances were still there.

In the second step, we produced engineered TPCD diaphragms, for which thickness and stiffness had been optimized in different areas of the diaphragms. The results of the engineered TPCDs were game-changing. The problematic resonances were not just moved; they were eliminated. The distortion peaks found in the drivers with the current diaphragms vanished.

For years we had been envisioning adapting the TPCD technology for over-ear headphone diaphragms. This research project showed us we could do it and that it had the potential to make a real difference. The moral of this story is that although TPCD is a great speaker diaphragm material, it is a game-changing speaker diaphragm technology. It is Composite Sound's diaphragm engineering that makes the difference.

Undesired Resonances—An Enemy of Good Sound Quality

In *Sound Reproduction — The Acoustics and Psychoacoustics of Loudspeakers and Rooms* by Floyd Toole, an objective for loudspeaker performance is described as timbral neutrality. In other words, the loudspeaker or headphone should not add or subtract from the sound it is to reproduce. Furthermore, Toole says the biggest culprits in coloration are resonances, and "by reducing the amplitude, the audibility of the resonance is reduced, as is the ringing."

Resonances show up as peaks and dips in the frequency response curve. A bumpy frequency curve with sharp peaks and dips is therefore a strong indicator of a poorly sounding loudspeaker or headphone. However, a smooth and flat frequency response curve that extends over the entire audible frequency range is an indicator of a good sounding, or at least neutral, loudspeaker or headphone.

Most high-end loudspeaker systems have the luxury of using multiple drivers and diaphragms, each covering a limited frequency range. Headphones typically use one driver and diaphragm that should ideally cover the entire audible frequency range. Having the diaphragm operate without undesired resonances in this full-range bandwidth is more challenging in a headphone. That also means that the ability to control diaphragm resonances in headphones could potentially make an even bigger difference than in a loudspeaker two- or three-way system.

The Conventional Way to Handle Diaphragm Resonances

There are many sources of resonances. In this article, however, only resonances originating from the diaphragm will be discussed.

It is usually said that the ideal diaphragm should be stiff, of low density, and well damped. High stiffness to density will move resonances (break up) higher up in frequency. Damping will reduce the amplitude of the resonances as well as their ringing.

Producing an ideal diaphragm in the conventional way means using a diaphragm material with high stiffness, low density, and high loss factor (well damped). The problem is that such a material doesn't exist—a stiff material is usually poorly damped and with a high density. A well damped material usually has a low stiffness. This forces the designer to make a choice as far as what to sacrifice.

There are conventionally two opposite ways to handle this, each with its own set of sacrifices. The first is the "hard diaphragm way," using as stiff a diaphragm material as possible to move resonances far enough in frequency so they are not in the audible range. The downside is that the stiffer the material, the less damped it tends to be, meaning that break up when it arrives is severe, producing audible artifacts (**Figure 1**).

The other way is to accept that there are resonances in the audible range but using damping in the diaphragm to reduce the amplitude, ringing, and thereby the effects of the resonances. Using a more damped material also usually means using a less stiff material, resulting in resonances appearing at lower frequencies.

Producing a conventional diaphragm, whether it's for a loudspeaker or a headphone, is typically done by forming a sheet of material. If the sheet has a uniform thickness, the diaphragm will have a uniform thickness, which is highly sub-optimized, at least if the target is a stiff, low-mass diaphragm with controlled resonances. But, is there a way to reduce the effects of undesired resonances without the drawbacks associated with the conventional ways described above?

What happens if we take a different approach: instead of merely trying to reduce the negative effects of undesired resonances (break-up), we control what is causing them?

Composite Sound—Born from F1 and Space

Composite Sound is not about standard diaphragms. In fact, Composite Sound doesn't have any off-the-shelf diaphragms. However, if you are looking for a diaphragm solution that is tailored to your specific requirements and visions, and a diaphragm that makes a true difference in your headphones or loudspeakers, then we have common ground.

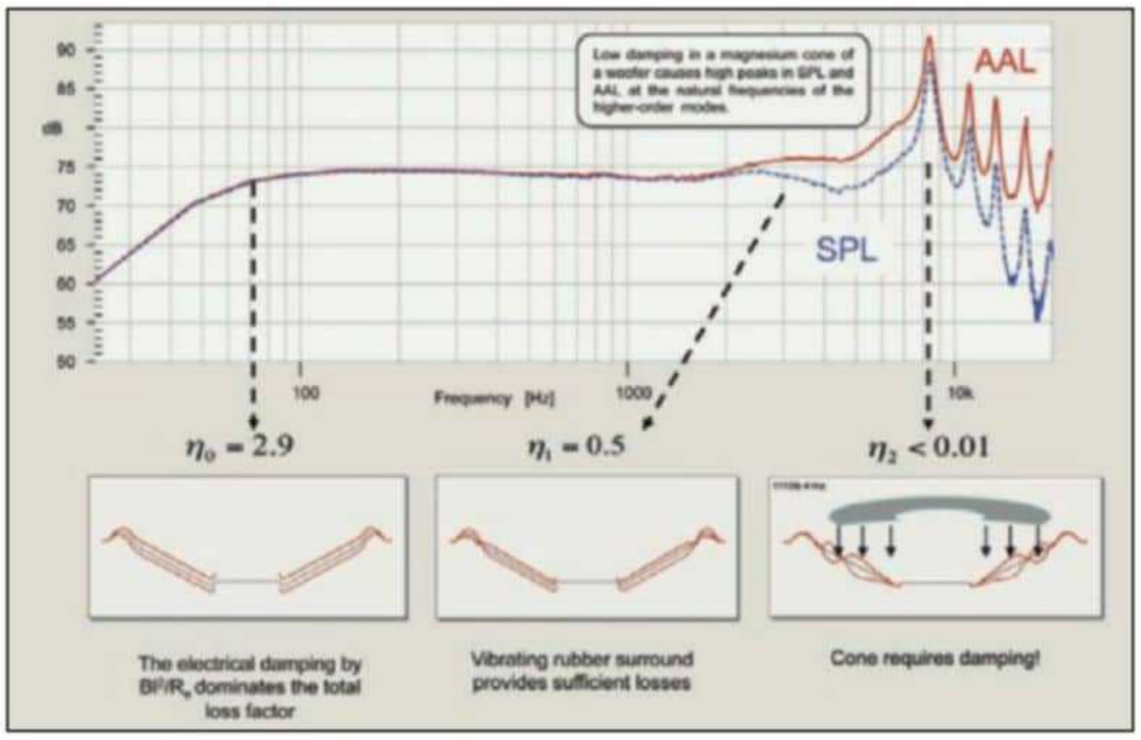


Figure 1: Undesired diaphragm resonances—"break-up" causing massive peaks and dips in a speaker cone. (Image Source: Klippel's Cone vibration poster, www.klippel.de/fileadmin/klippel/Files/Know_How/Literature/Papers/KLIPPEL_Cone_Vibration_Poster.pdf)

To us an ideal diaphragm relieves you from having to make compromises. An ideal diaphragm enables you to make it sound like you want it to sound. An ideal diaphragm enables you to realize your visions. Engineering an ideal TPCD diaphragm is what Composite Sound is all about. We do things differently because we come from a different background.

Based in Borås, Sweden, Composite Sound is a spin-off from Oxeon, the inventors and producers of TeXtreme thin-ply carbon materials. TeXtreme is used in applications with the absolute highest requirements on stiffness, low mass and durability. TeXtreme is used when other materials, including conventional carbon fiber, are not stiff enough, not light enough, or not durable enough. TeXtreme helps Formula One (F1) race cars run faster, and has enabled NASA to explore Mars. NASA's drone Ingenuity is largely built from TeXtreme thin-ply carbon material (**Photo 1**).

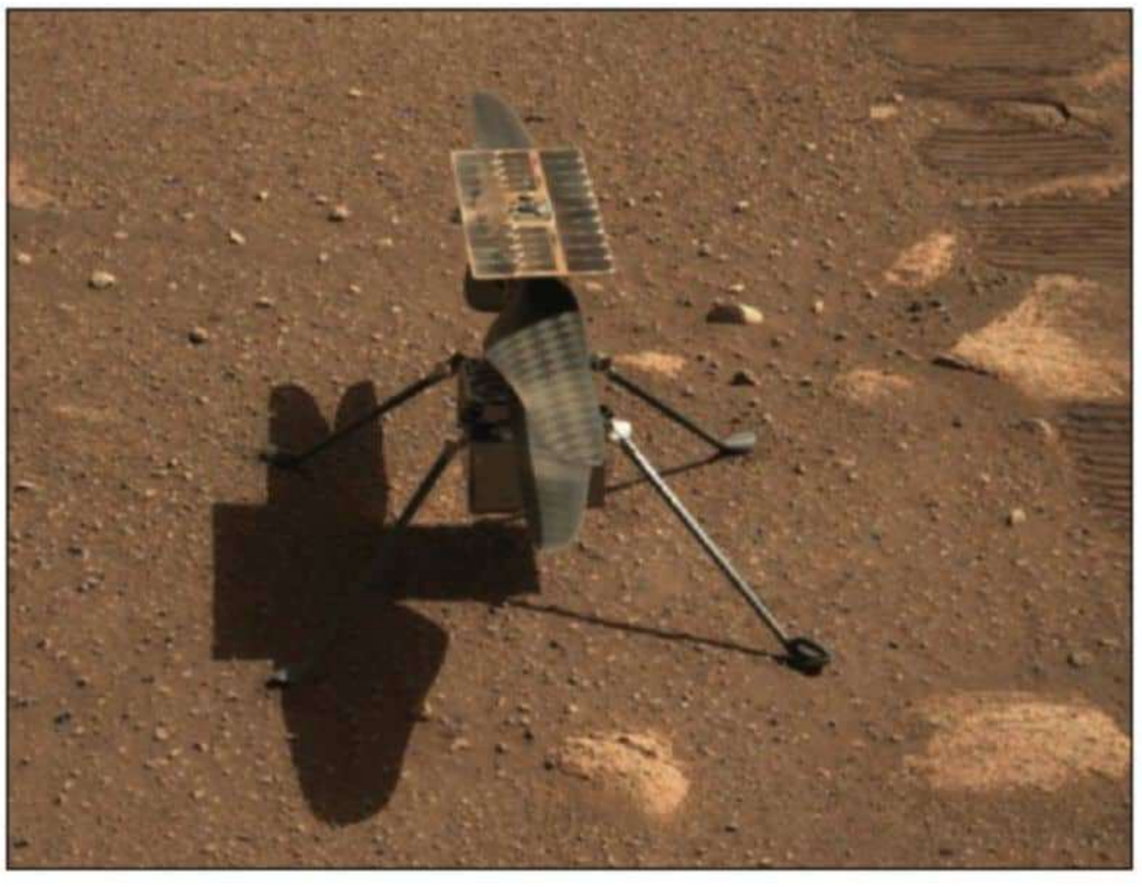


Photo 1: The best material technology on the planet… Mars. NASA's Mars drone Ingenuity is largely built from TeXtreme material. (Image Source: NASA/JPL-Caltech/ASU)

Although high-performance thin-ply carbon composites have great material properties, what makes it truly different from conventional technology is its engineering. Thin-ply carbon composite technology provides freedom of engineering properties in different areas and directions to maximize stiffness, minimize mass, and even enable the composite part to behave in certain desired ways under certain conditions. That is what Composite Sound uses to engineer loudspeaker diaphragms and what we are now introducing to over-ear headphones.



Photo 2: Here are a few examples of Composite Sound's TPCD tweeter domes.

Photo 3: This an example of a Composite Sound's TPCD midrange cone.

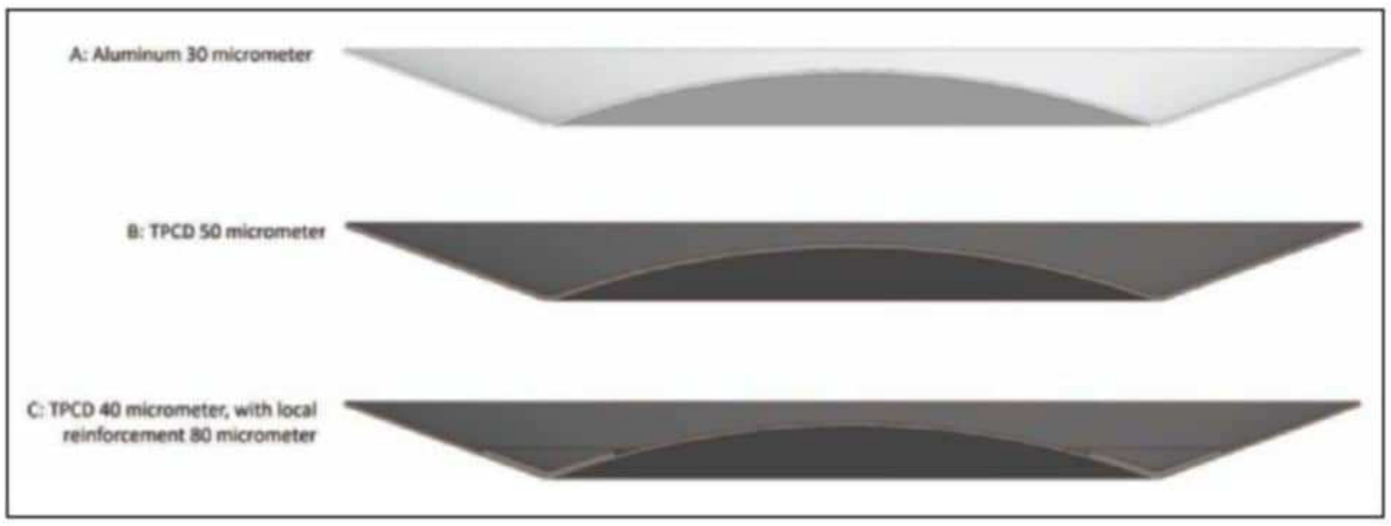


Figure 2: This is a comparison of aluminum and TPCD diaphragms.

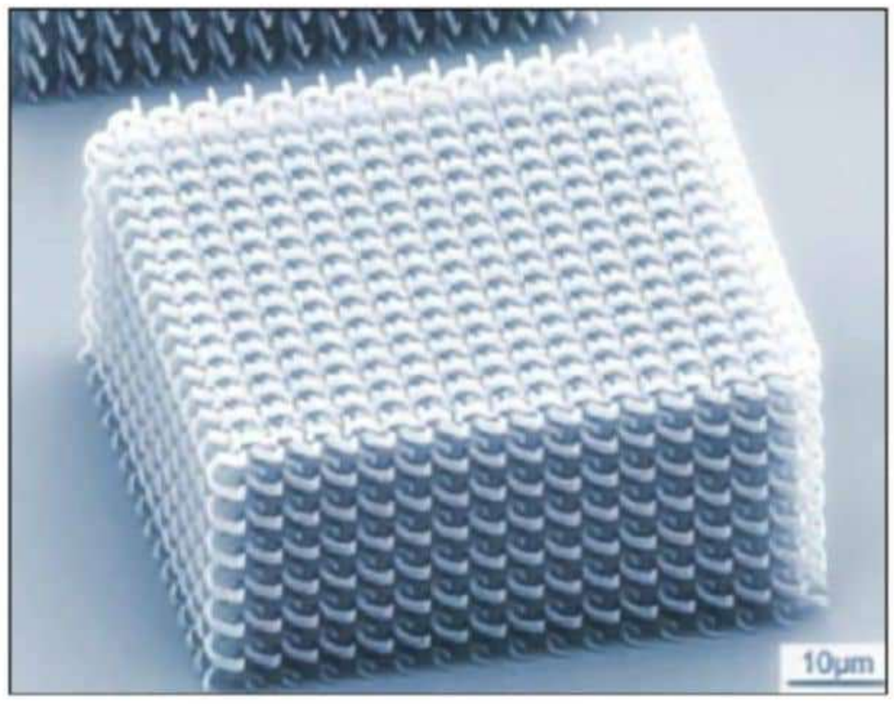


Photo 4: This is metamaterial organized into a structure.

Engineering TPCD for Headphones

Just as we did in in the research project described at the beginning of this article, engineering a TPCD headphone diaphragm is about combining the resonance control abilities of Composite Sound's bigger cones with the thinness, stiffness, and lightness of our tweeter domes (**Photo 2** and **Photo 3**). Composite Sound's production process is scalable, meaning the same principles and processes that enable an optimized 110mm midrange cone with 0.2mm thickness can also be used for a 30mm over-ear headphone diaphragm with 0.05mm thickness.

To Composite Sound, an ideal diaphragm is stiff, of low mass and doesn't suffer from undesired resonances. The most basic way of engineering a diaphragm with these objectives is to optimize the thickness in different areas of the diaphragm. Strangely enough, it is hardly ever used in conventional diaphragms.

Stiffness and Mass

Imagine you want to build a bridge. Your goal is to build a bridge with a large span crossing wide waters. The requirements are tough: the bridge must handle cars and trucks, heavy winds, and even withstand earthquakes and ships crashing into it. To up the challenge, it should use as little material as possible. How would you build it? Would you use a uniform sheet of steel or concrete and build the bridge equally thick in all places?

A headphone diaphragm has more similarities to a bridge than you might first think. Just as the bridge can be made a hundred times longer with an optimized thickness and stiffness profile, combining tension and compression, the TPCD headphone diaphragm can also be tailored to perform at a substantially higher level than current diaphragm technology.

Let's look at an example comparing three 30mm diaphragms for a 40mm over-ear headphone driver with the same geometry and mass, weighing 0.06 grams. Diaphragm A is made from 30µm thick aluminum. Diaphragm B is a non-engineered TPCD with uniform thickness. Due to the 40% lower density, it will have a 40% higher thickness than the aluminum diaphragm, 50µm. Diaphragm C is a TPCD with 40µm thickness but with double

thickness just around the area where the voice coil is attached (**Figure 2**).

Using simulations in ANSYS with simplified boundary conditions of the diaphragm in isolation, the resulting fundamental resonances (break-up) occurs at the following frequencies:

- Diaphragm A, aluminum: 7.5kHz
- Diaphragm B, uniform thickness TPCD: 12.4kHz
- Diaphragm C, varying thickness TPCD: 16.1kHz

The only engineering that has been made to Diaphragm C is doubling the thickness of an important part of the diaphragm. There is much more that can be optimized in this diaphragm.

What possibilities does this provide? To name a few:

- The diaphragm can be made bigger for better bass performance.
- Resonances can be controlled for better sound quality.
- Mass can be reduced for higher sensitivity.
- The level of optimization can be taken much further.



Photo 5: Here is one of Composite Sound's TPCD Metamodal cones.

About the Author



Martin Turesson is Head of Thin-ply Carbon Diaphragms (TPCD) at Composite Sound. Having a master's degree in mechanical engineering from Chalmers University of Technology in 2001, he began working at Oxeon in 2009 helping customers engineer high performance composite products for aerospace, Formula One and industrial applications using Oxeon's TeXtreme carbon materials. In 2016 he started exploring the possibilities of TeXtreme thin-ply carbon materials for speaker diaphragms leading to the spin-off of Composite Sound in 2021. Composite Sound is specialized in engineering and producing TPCD speaker diaphragms. Martin can be reached at martin.turesson@composite-sound.com

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Controlling Resonances

A metamaterial is organized into a structure to give it a behavior or performance that is not naturally occurring in the material itself. In the same way that metamaterials can be artificially structured to control light and metasurfaces to manipulate soundwaves, Composite Sound Metamodal technology can control and manipulate resonances.

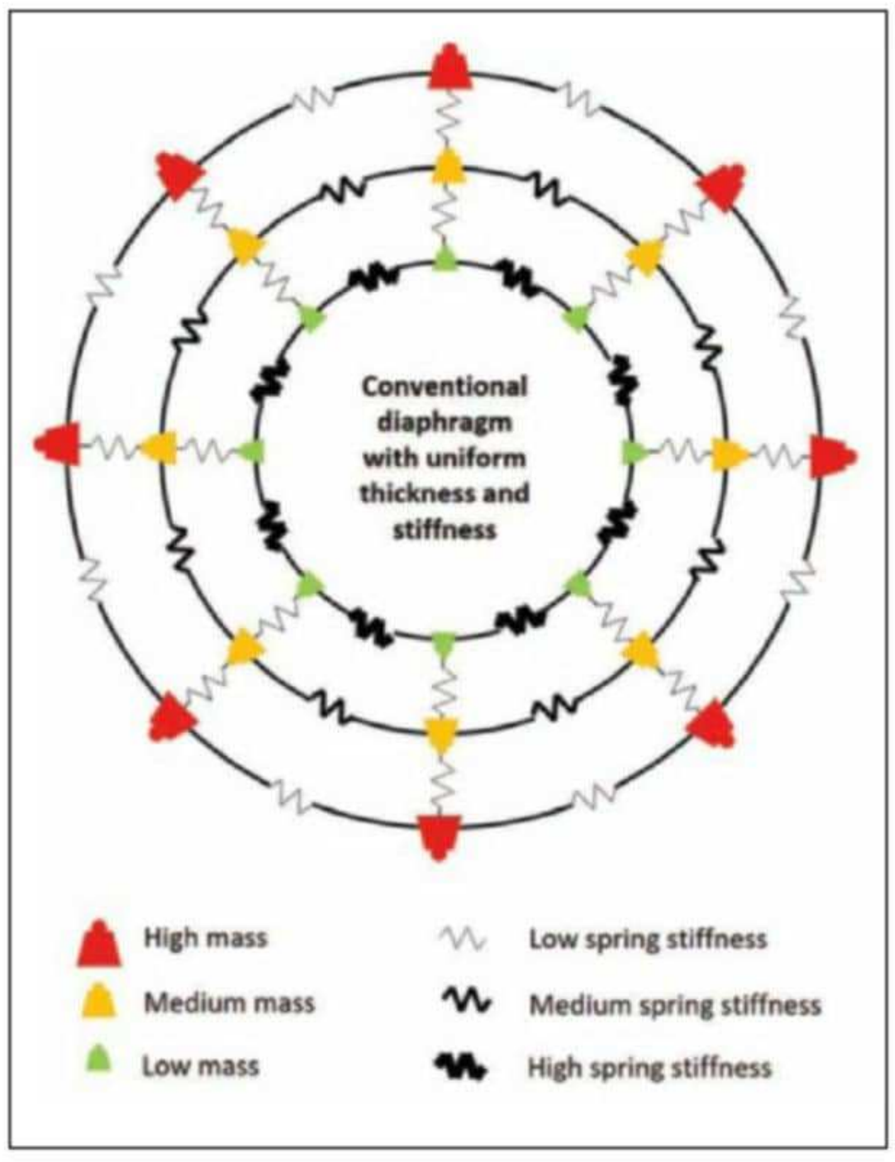


Figure 3: Conventional cone as a mass spring system, as described by F. J. M. Frankort in his 1978 paper, "Vibration Patterns and Radiation Behavior of Loudspeaker Cones."

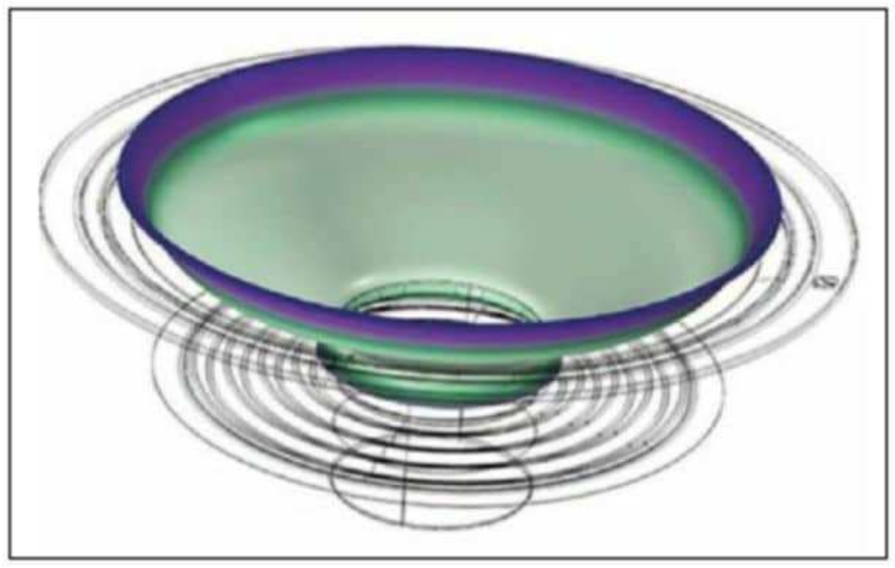


Figure 4: Using a diaphragm as a mass-spring system, the mass will increase the further they are from the center, and spring stiffness in azimuthal direction will decrease causing the mode to start at the outer edge.



Photo 6: This is an AI-generated image of a drop of water in a bowl to demonstrate how the waves will simultaneously arrive at the outer edge around the cone.

An example of metamaterial structure and a Composite Sound TPCD metamodal diaphragm share a common approach, in that its properties can be engineered to fit the application target (**Photo 4** and **Photo 5**).

This may sound like magic, but it is physics. Any time the thickness, stiffness, or density is changed in any location of an object, it will change how resonances occur in that object when excited. Hence, by optimizing these factors, it is possible to control the resonances in that object.

F. J. M. Frankort describes diaphragm resonances in the Audio Engineering Society (AES) paper "Vibration Patterns and Radiation Behavior of Loudspeaker Cones." [2] The diaphragm is defined as a truncated cone with an inner and outer edge, in which the voice coil is attached to the inner edge. When the voice coil moves, it will create a wave that will travel from the inner to the outer edge of the cone. Although some wave energy will be lost in the diaphragm (in the model Frankort disregards the surround), the wave will be reflected on the outer edge and travel back to the inner edge. If the wave is in phase with the voice coil movement, the amplitude of the wave will be increased, and a resonance is created.

Furthermore, Frankort describes the diaphragm (cone) as a system of masses and springs in the longitudinal direction (from the inner to the outer edge) and as springs in the azimuthal direction (going around the cone). As we go further toward the outer edge, the masses will increase, and the stiffness of the azimuthal springs will decrease (**Figure 3**).

Using a diaphragm with uniform thickness and uniform stiffness, as conventional diaphragms are made, means that the diaphragm will not be able to compensate or adjust for this effect with increased mass and reduced spring stiffness going from the inner to outer edge (**Figure 4**).

It will also not be possible to adjust for and tailor how resonances occur. Under symmetric conditions the modes will be symmetric meaning large areas of the diaphragm will move in and out of synch at certain resonance frequencies causing big peaks and dips in the frequency response. The waves will arrive at the outer edge around the cone at the same time, in a similar fashion as the ripples from a drop will reach the edge of a bowl of water (**Photo 6**).

However, if it is possible to change the thickness and stiffness profile of the diaphragm, the mass

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loads and stiffnesses of the springs can be changed, and thereby the modal behavior and how the resonances occur can be changed.

Frankort's model covers axisymmetric cones and conditions. This is fine since most conventional diaphragms are isotropic (same properties in all directions) and can be considered axisymmetric.

With Composite Sound's TPCD Metamodal technology, we can move from 2D axisymmetric conditions to 3D. That means that not only can the thickness and stiffness be tailored going in a longitudinal direction from the inner to the outer edge, but it can also be tailored in different areas and directions as we travel in azimuthal (circumferential direction) around the cone (**Figure 5**).

A TPCD diaphragm can have one thickness and stiffness profile at 2 o'clock, another at 3 o'clock and yet another at 4 o'clock, etc. In other words, in a TPCD Metamodal diaphragm every mass and every spring in radial and azimuthal direction in any position in the diaphragm can be tailored. If this is done in an ideal manner, it theoretically enables complete ability to control how diaphragm resonances occur.

Going back to the analogy with the ripples in the bowl of water, with Composite Sound's TPCD

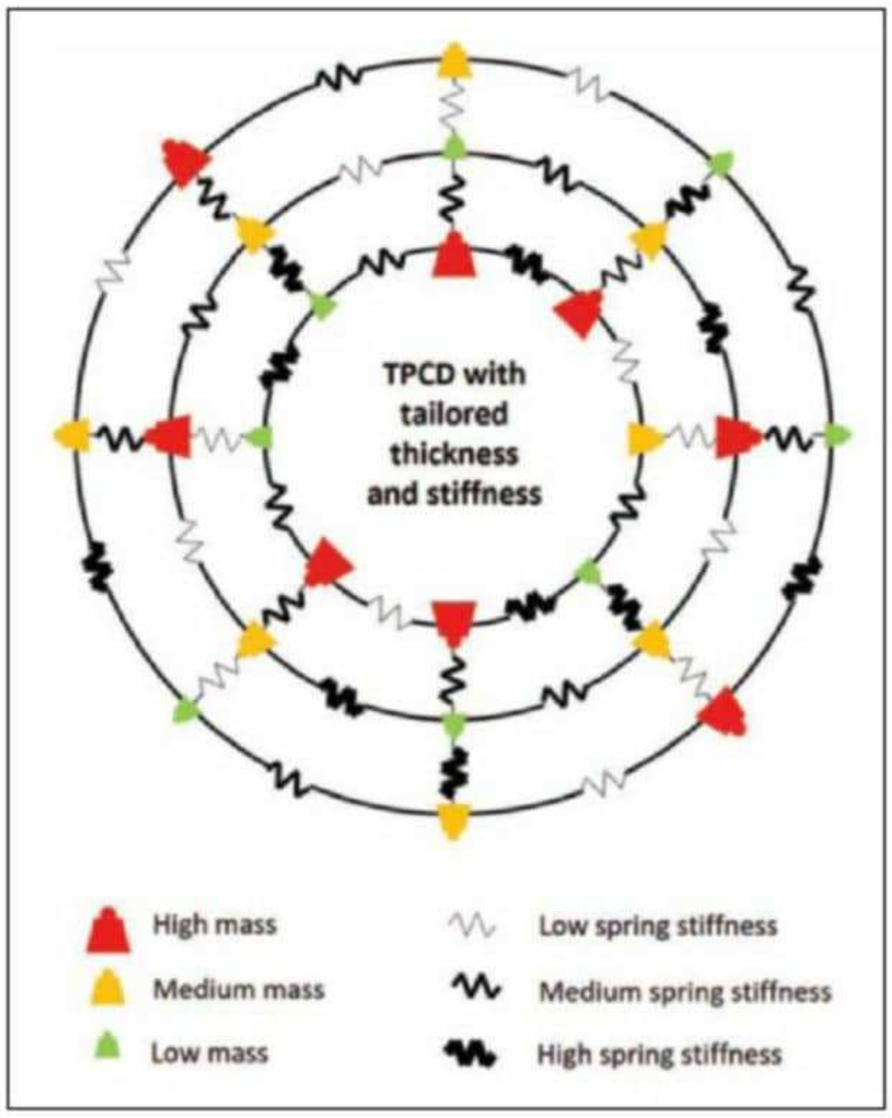
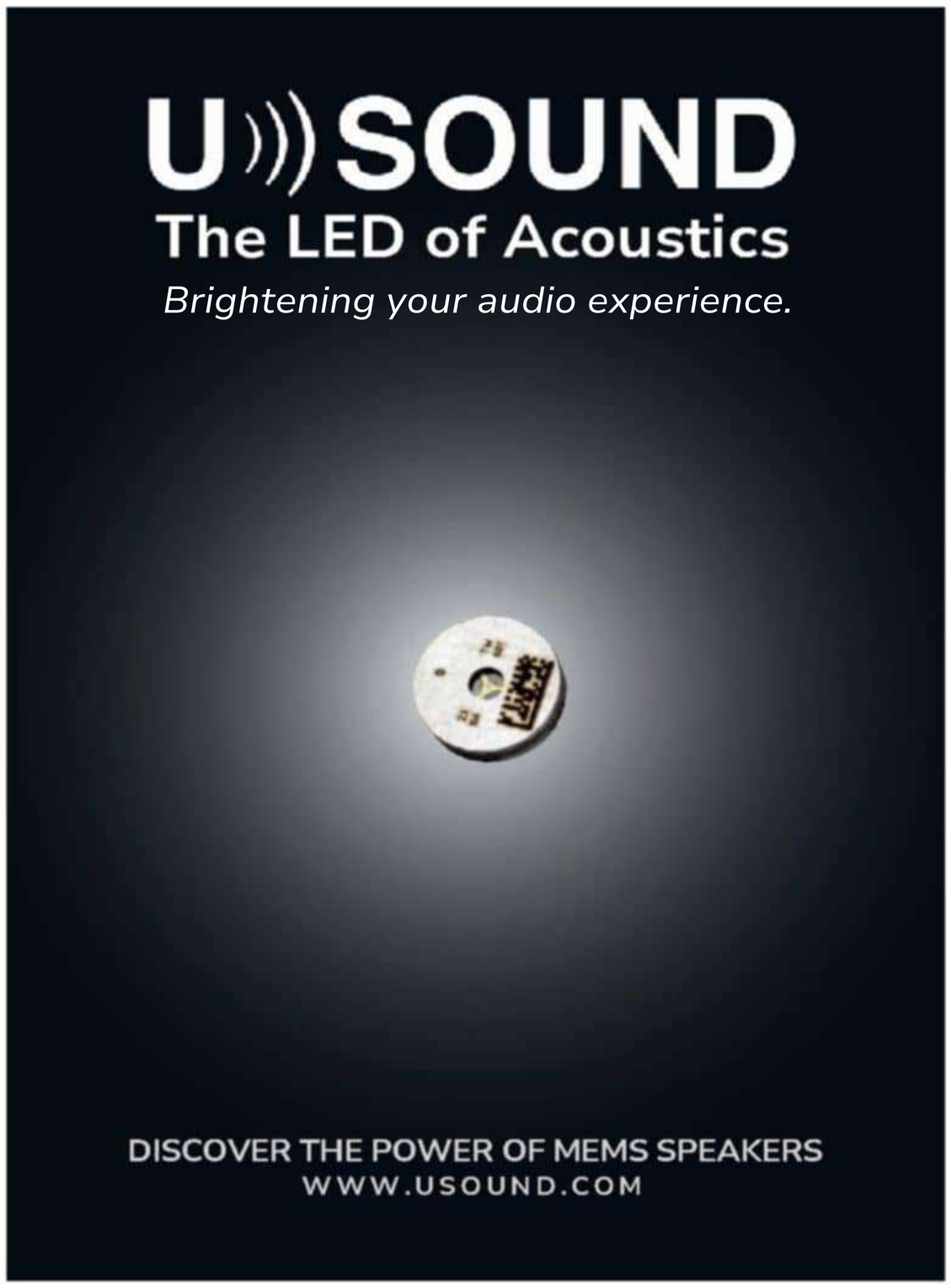


Figure 5: A TPCD diaphragm as a mass-spring system. The mass (thickness) and the spring stiffness can theoretically be tailored in any position and direction in the diaphragm. That means resonance behavior can be tailored.

References

[1] M. Turesson, "Metamodal TPCDs — Next-Generation Speaker Diaphragms from Composite Sound," *Voice Coil*, February 2022, https://audioxpress.com/article/metamodal-tpcds-next-generation-speaker-diaphragms-from-composite-sound

[2] F. J. M. Frankort, "Vibration Patterns and Radiation Behavior of Loudspeaker Cones," *Journal of the Audio Engineering Society* (JAES), Volume 26, Issue 9, pp. 609-622, September 1978.





Metamodal, we can control how the water ripples are formed and travel in the bowl (**Figure 6**).

The toolbox and possibilities are infinite, our main challenge is to effectively simulate the optimum combination of masses and springs. As those familiar with COMSOL simulations can testify, simulating isotropic materials in 2D is challenging enough, simulating an orthotropic diaphragm (different properties in different directions) in 3D is a massive undertaking.

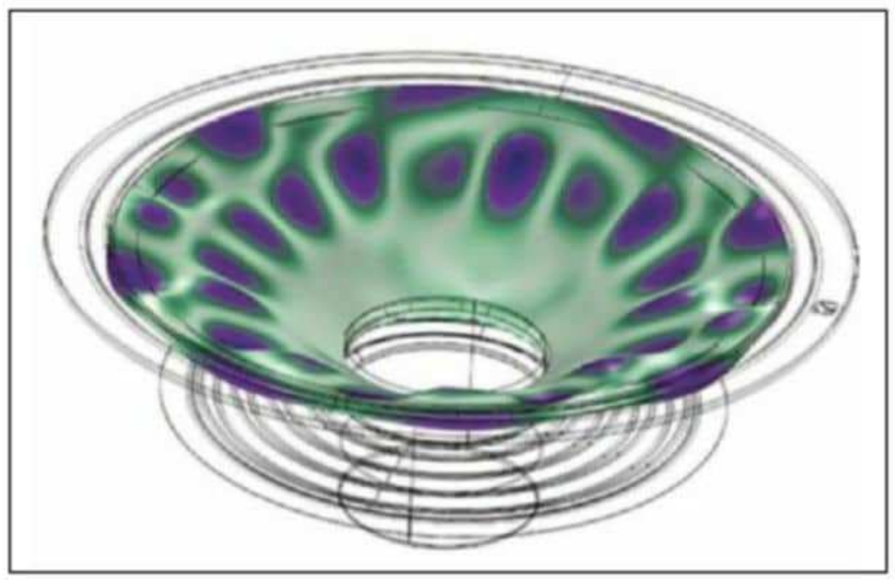


Figure 6: One example of the modal behavior resulting from one TPCD mass-spring setting. With another TPCD mass-spring setting the modal behavior will be different, as will the resulting resonance behavior.

An Example of a Full-Range TPCD Diaphragm

Figure 7 shows frequency curves of the same full-range driver (not a headphone driver), using three different diaphragms of the same geometry and with an outer diameter of 67 mm: Figure 7a shows a conventional glass fiber diaphragm; Figure 7b shows a non-engineered TPCD diaphragm; and Figure 7c is an engineered TPCD Metamodal diaphragm.

As can be seen, the glass fiber diaphragm has a clear fundamental resonance (break-up) just below 8kHz. The non-engineered TPCD still has that fundamental resonance, but it has been moved to around 12kHz. In the engineered TPCD Metamodal diaphragm, the fundamental resonance has been eliminated and it is not possible to identify a break-up.

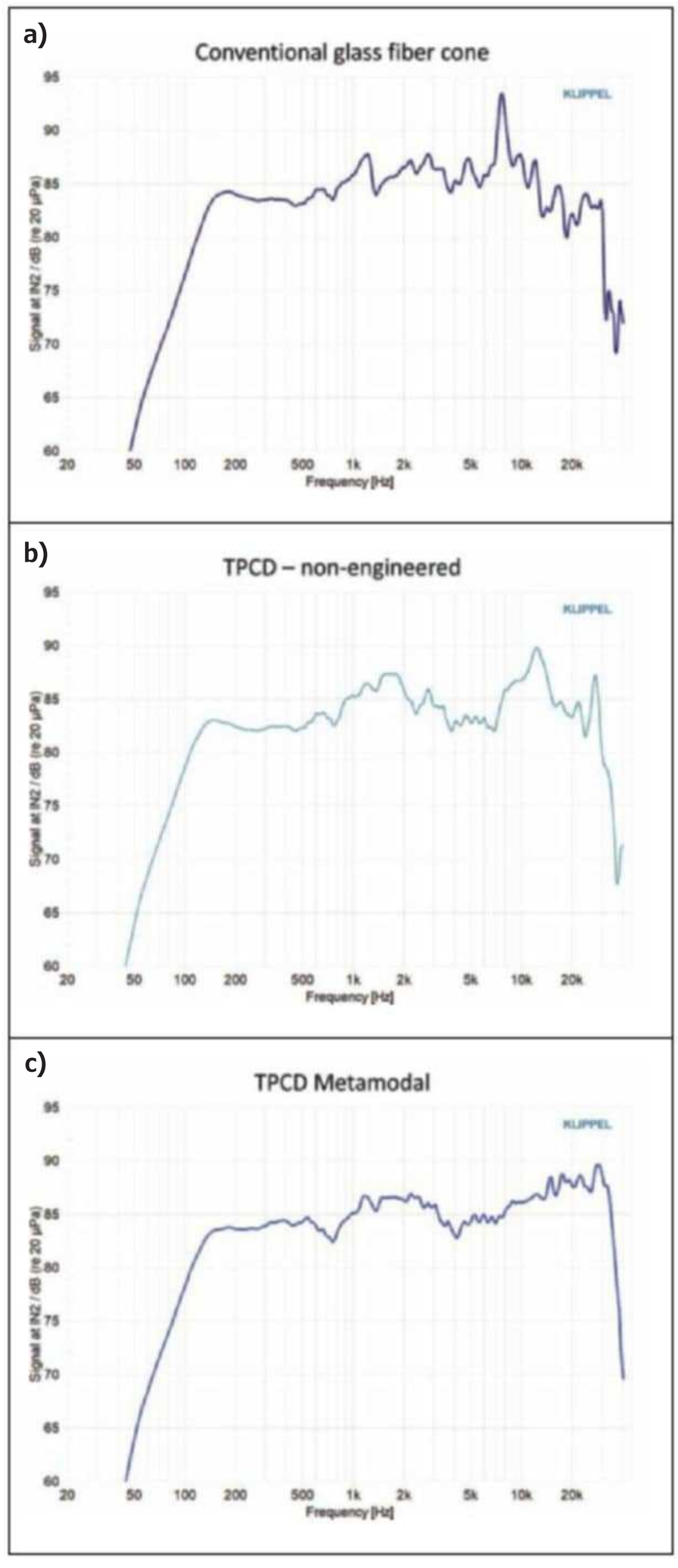


Figure 7: Frequency curves of the same full-range driver (not a headphone driver), using three different diaphragms of the same geometry and with an outer diameter of 67mm are shown with a conventional glass fiber diaphragm (a); a non-engineered TPCD diaphragm (b); and an engineered TPCD Metamodal diaphragm (c).

Explore, Evaluate, and Realize

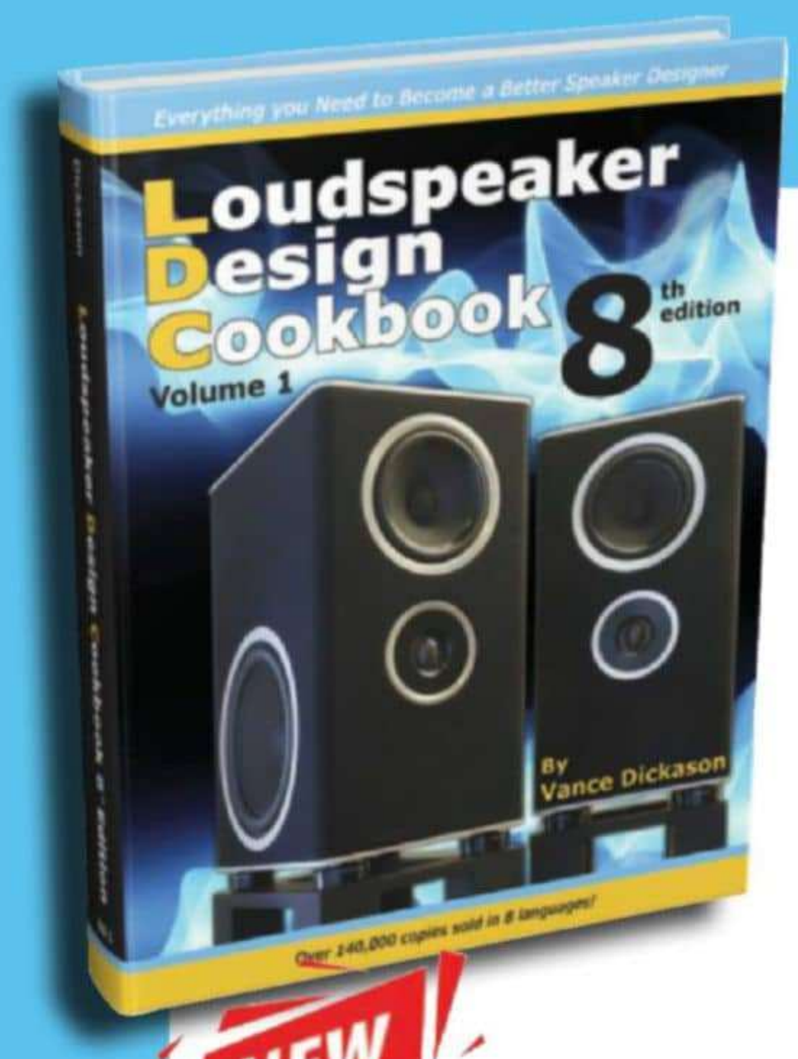
To make it easy and transparent to explore, evaluate, and realize engineered and tailored TPCD diaphragms, Composite Sound developed a three-step process:

- Sharing information: The first step is to provide information about the TPCD technology and answer all your questions. It also includes learning about what is important for you and what you are looking for. What would make a real difference for you?
- Pre-study analysis: We perform simulations on the diaphragm/driver you would like to explore with the potential of a Composite Sound TPCD. The simulations will show the potential performance of Composite Sound TPCD compared to your reference diaphragm.
- Prototype testing: Should you wish to test TPCD diaphragms in your headphone, driver, or loudspeaker, we will provide Composite Sound TPCDs tailored for you. Alternatively, we can arrange drivers using Composite Sound TPCD for you to test. Composite Sound cooperates with leading headphone and loudspeaker driver manufacturers.

When you are ready, we will optimize the diaphragms for performance, quality, and manufacturability, and start the production of your tailored Composite Sound TPCDs.

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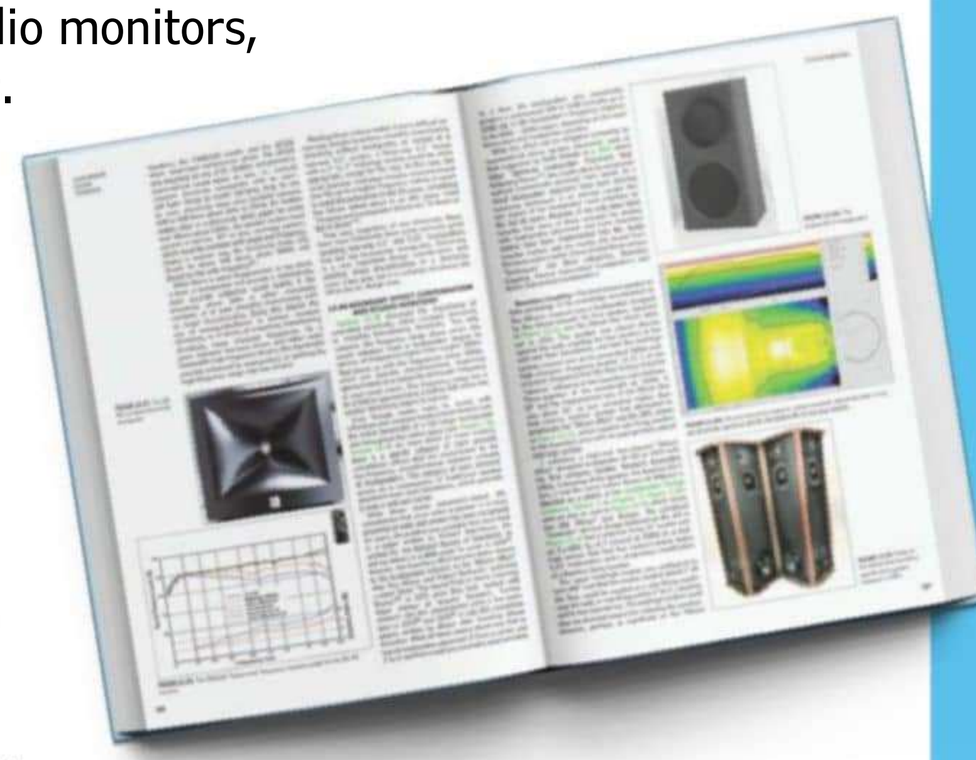
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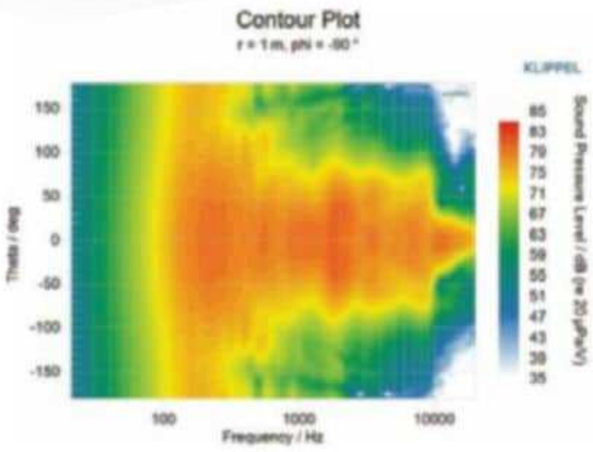
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- New Chapter 8 — "Loudspeaker Testing" has been substantially revised and includes specifics on the Fixed Mmd method for calculating cone assembly mass, a more complete definition of acoustic phase, an updated list of computer based TSP (Thiele Small Parameters) measurement hardware, plus a discussion of the various types of loudspeaker measurements and how they relate to driver quality and system design integrity.

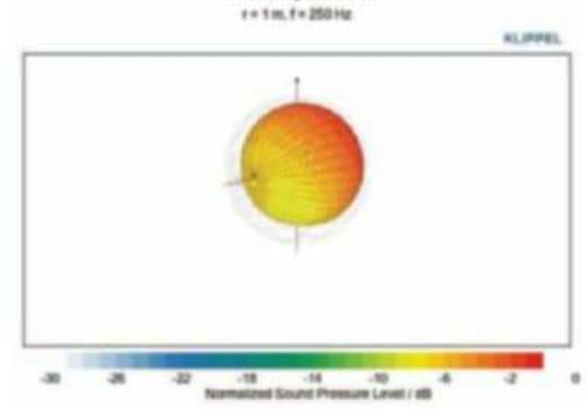
- New Chapter 9 — "Computer Based Tools for Loudspeaker Development" has been completely revised to include current offerings of computer based loudspeaker measurement analyzers, as well as loudspeaker enclosure and crossover simulation software.



- New Chapter 10 — "Room Measurement Equipment and Room Correction Solutions", and includes a survey of room measurement analyzers, plus a discussion on room acoustics, room treatment and digital room correction devices.

- New Chapter 12 — "Home Theater Loudspeakers" and has been extensively revised, including details on Dolby Atmos and Immersive Audio.

- New Chapter 13 — "Studio Monitor Loudspeakers". This new chapter explores the difference between HIFI loudspeakers and studio monitors, professional criteria for studio monitor design, the constant directivity aspect of studio monitors, and a dissertation on zero phase in studio monitors.



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An Afternoon with Robert Munnig Schmidt



What do motional feedback, silicon wafer steppers, voice coils, and Class-D amplifiers have in common? Everything, as far as Prof. Ir. Robert Munnig Schmidt is concerned. So, *audioXpress* Technical Editor Jan Didden decided to find out why and went to interview the man known as “mister motional feedback.”

By
Jan Didden

I met Robert Munnig Schmidt at Grimm Audio when he was working on the company’s original Digital Motion Feedback subwoofer. Only much later did I discover that he was also part of the initial design team at ASML, the famous IC fab manufacturing equipment company founded in the Netherlands in 1984 (ASML originally stood for Advanced Semiconductor Materials Lithography). He joined ASML after several R&D jobs at Philips Research… but of course, I needed to ask him about his initial trajectory and how he arrived there.

Jan Didden (JD): How did it all start?

Robert Munnig Schmidt (RMS): It’s been a convoluted road, but there’s a common thread. It started with me, as a kid, playing the flute in a band. A friend’s father happened to be a recording engineer with the famous Ariola brand. This guy built his own mixing console and amplifiers, and my friend and I used his older equipment for our hobby. We built tube amps, and I put so much time into it that I had to double one of my college years. But you really must get a diploma at some time, and I went to study at Delft University. People told me, if you are a guy who understands instantly how something mechanically works just by looking at it, you need to study mechanical engineering. But if you really like math, you should study electrical engineering. I wasn’t bad at math, but not a wizard, so I decided to study mechanical engineering and do electrical engineering as a hobby. And that really was the start of the theme for my life.

My graduation project was a portable medical infusion pump, driven by a voice coil. It is conceptually the same as you see in some Bulova Accutron watches, with a vibrating mechanical part. It worked beautifully at my graduation and was also a sign that next to designing mechatronics I also wanted to work with my hands. Which is logical since I really loved to play with coils and magnets in the audio area.

JD: And then it was time to get a job?

RMS: Hah! What seems unbelievable in this day and age: after graduation, nobody offered me a job, so I coasted on unemployment benefits. I met a professor in Delft, and he said, “Have you thought about Philips?” — “Don’t they just manufacture light bulbs?” I asked. Totally out of touch. But the professor dictated my CV and a letter to Philips’ HR, and I got invited and ended up in the famous NatLab in Eindhoven.

I got to work in the Optical group. This was around 1977. They had two major projects: optical recording and the Silicon Repeater, which turned out to be the precursor to what later would become ASML! Because of the strict hierarchical organization at the time, each employee would be attached to a university graduated person who he would report to. One guy, Gerard van Rosmalen, was attached to me. He was older than me and already famous as the designer of many patented designs of the motion systems that were used for the optical pickup of VLP and the compact disc.

I learned as much from him as I could. One important thing I learned was to have no fear of mass. He taught me that mass and bandwidth can be largely decoupled.

The wafer stage of the Silicon Repeater was hydraulically powered, and there were always leaks, stuff got oily, not a good idea in a CD cleanroom. I concluded we should be able to do that electronically,

and if you must move more mass, you just use more power!

JD: But if the mass is higher, it is more difficult to get it moving—and stopping—fast. That must have bandwidth repercussions.

RMS: It can be controlled separately. If you have a large mass to move, say a thick heavy rod, it takes some time from the movement of the actuator until the effect reaches the end of the rod, there's a wave traveling through the material. But it is still moving much faster than sound in air. If you actuate a heavy load with a big coil and magnet, it's pretty easy to get sufficient bandwidth to control the accuracy (**Photo 1**). The real limit was amplifier power.

In speakers at that time, the design was focused on efficiency. We had voice coils that had the same dimensions as the magnetic field, and they would move outside that field, but it was very efficient. Drive power was at a premium, also because all power amplifiers were limited by heat dissipation. Somewhere I heard about switching power converters, so I decided to build a switching amplifier! I realized that if you modulate the duty cycle of a square wave you could reconstruct the original signal with a simple low-pass filter. I also was interested in how this would affect crossover distortion.

I was never a big fan of feedback, as feedback is a reaction to an error so is always late. Feedback doesn't eliminate distortion, it can only decrease it, so it should only be used after you have done your best to make the amp as linear as possible! I thought, in a switching amp, the output switches between positive and negative supply so you wouldn't have crossover distortion. And I really designed that switched stepper drive amp with audio in mind, for 20kHz bandwidth. But to my chagrin, that amp also had crossover distortion!

JD: In a switching amp there's no gradual current takeover from one polarity to the other. So, what is the mechanism?

RMS: Depending on the current direction, the energy in the filter coil will try to keep the current flowing. As a result, a positive current will speed up the negative switching ramp from the input of the filter, while a negative current speeds up the positive ramp. At the crossover point, this causes a change in ramp speed and that manifested itself as crossover distortion. You can minimize it with circuit optimization and feedback after the low-pass filter—as Bruno Putzeys has done at Philips, and later at Hypex in the UcD amplifiers—but you can't get rid of it completely.

Nevertheless, the amp worked fine in the stepper and was still used in the early ASML equipment. This was, however, a voltage amplifier, but nowadays everything in wafer steppers and the modern wafer scanners is current driven with current output amplifiers to guarantee a driving force that is only dependent on the current and no other factors like motion feedback and self-inductance.

JD: Did Philips have audio motional feedback products on the market at that time?

RMS: Yes, they did, and it was then that I found out that—for cost reasons—there was a big discrepancy between what the designers at Philips delivered and what was finally put out to the customer under the adage: "a bit better is good enough."

I then contacted the factory in Dendermonde in Belgium and got some discarded motional feedback (MFB) drivers with intact sensors. I bought a pair of Isophon 12" woofers that had the magnet inside the voice coil, which I thought was a smart design. No stray fields. It was still ferrite, so the field wasn't particularly strong. I mounted the salvaged sensors in the Isophones, connected to some control electronics I made from the last schematic from the NatLab audio guys. This was my first MFB speaker and I still use it today!

JD: So that launched you on the audio path?

RMS: Not yet! I had a good job at Philips but had no time for hobbies, so I had to buy my stuff from the Philips personnel shop. I bought a pair of

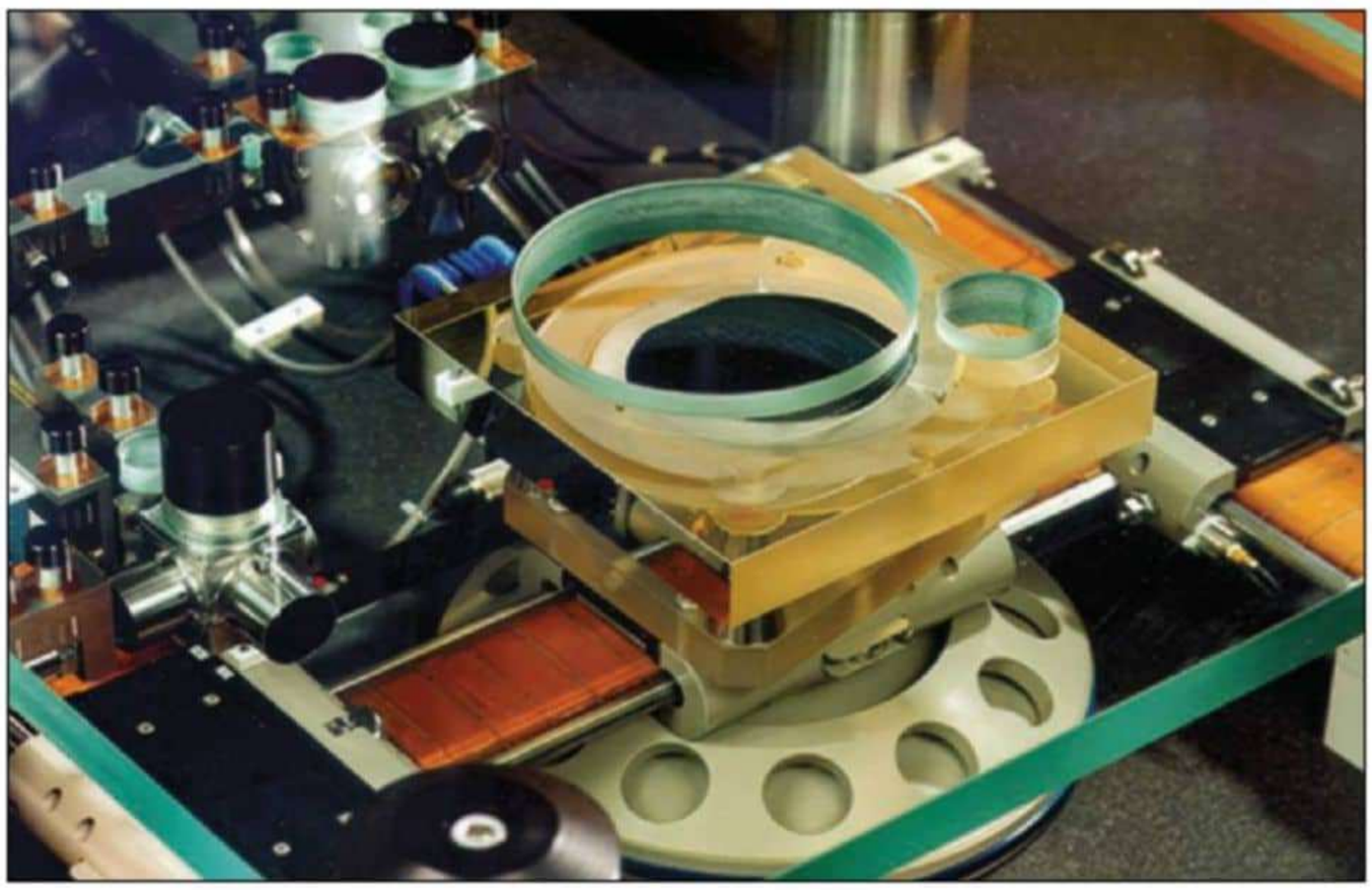


Photo 1: An early wafer stepper with X-Y voice coil drive

Philips Digital Speakers, they sounded horrible. I made some improvements, but it remained a dud. I gradually became more and more disappointed about the role of technology at Philips under the pressure of cost-driven marketing, so I decided to move (back!) to ASML where my technical skills were challenged more.

JD: But you left ASML; why?

RMS: At one time I was asked by my alma mater, Delft University, for a professor post. The stress and workload at ASML had started to take its toll, so I thought, why not, even with a pay cut. I took the jump, only to find out that I did not easily fit in the role of a good university professor. But I had to adapt in many more ways. It takes years of hard work and study before you get to be a professor, but once you have your seat, you're on your own. If you wanted to do some research, you had to find the money somewhere through mostly bureaucratic channels. And if there is one thing an ex-ASML director hates, it's bureaucratic channels! The contrast is immense.

At Philips you only went to your boss when the project was finished, and you had solved all issues. "Don't bother me with your problems." At ASML, you were asked: "Did you finish the project? If not, what can I (as your supervisor) do so you can finish it? What do you need, do bother me with your problems, because that's what I am here for!"

At the university, there was no common goal nor a demanding and supporting supervisor that I appeared to need for motivation. That was the motivation to write my book on mechatronic engineering [1].



Photo 2: The professor exercising his MFB woofers with 16Hz organ notes. In 2015 Robert Munnig Schmidt retired from the university to dedicate fully to mechatronic education and the design of active controlled low-frequency sound systems.

JD: It doesn't sound like you were happy at the university.

RMS: I loved teaching, but I hated examinations, and the research was not really satisfying as the work of one student per topic never goes fast enough when you are accustomed to the ASML environment.

I was saved when a student told me: "I want to work on motional feedback." I thought, isn't that old hat? We worked on that decades ago!

First, we would find out how the "plant" as we called it, worked, characterize all nonlinearities and resonances, and then tried to compensate for that. That's feedforward control. Ninety-nine percent of the work at ASML is to make sure the hardware works very linearly before the application of feedback. If you have 100 kilograms of mass banging around at nanometer scales, and you want to linearize it with negative feedback, it's a lost case, you get a bandwidth of maybe a few 100Hz if you're lucky. So, we always put a lot of effort into mathematically describing the plant and linearizing it, and then adding negative feedback.

When we started on the design of a MFB loudspeaker we approached it the same way, starting with feedforward control. And that didn't work. You can try to characterize the driver, and Wolfgang Klippel, for instance, has done important work on that, but a driver isn't a driver. Unlike an ASML item, which is characterized down to the smallest detail, a driver is a mass product and no two are exactly the same. You could characterize each individual unit and tune it, but due to external influences like temperature and wear this should be repeated on a regular basis, which becomes very costly. So, your model is never accurate enough. And the complexity of resonances and break-up, it's all three dimensional, you just can't correct them at a single point where the voice coil is attached…

JD: So, back to traditional negative feedback then?

RMS: Yes, but with a twist: implementing the control loop in the digital domain.

We started some experiments with a professional d-Space digital controller, with many options and degrees of freedom. I got a few ACH-01 acceleration sensors, and we were having lots of fun. But ultimately, we needed a low-cost digital control that could be manufactured and implemented in every driver.

A DSP solution needs A-D and D-A conversion that in most converters causes delays of several milliseconds latency, which drastically lowers the stable control bandwidth you can achieve. The same problem exists for noise cancelling headphones, and we discovered that these use special A-D and D-A converters with latency down to a few microseconds! It's not really hi-fi but we wanted to use it for the lower octaves only. You can make good tweeters, and as long as you stay away from the resonance frequency, a speaker can be very good. Speaker distortion is less irritating than amplifier distortion. At (very) low frequencies, however, you need to move a lot of air, so you need a large cone, which leads to a large enclosure, and you get a monster that nobody wants in the living room.

If you can reproduce down to 32Hz with full power and good linearity you can reproduce all music, even a 32' organ pipe (16Hz), which mainly has the 32Hz harmonics anyway (**Photo 2**). But if you put a large cone in a small box, you get resonance frequencies at 50Hz or 60Hz and below that distortion rises very quickly and is very audible due to the fact that at the lowest frequencies our ears are more sensitive to the harmonics than to the fundamental.



Photo 3: The original LS1s-DMF motional feedback subwoofer turned the Grimm Audio LS1 speakers into a powerful three-way system.

JD: And the feedback will help you lower that distortion?

RMS: Yes, and with acceleration feedback you really add virtual mass to the system, which means lowering the resonance frequency. That is the basic effect of motional feedback. And to lower distortion you need loop gain. Where? Not at those frequencies at 32Hz, because the harmonics are above that! You need loop gain at 60Hz, 80Hz, 120Hz! Not many people realize that. And bandwidth is just a consequence of the loop gain you need at higher frequencies. We started to experiment with an ADAU1772 and later the ADAU1777, which had





Photo 4: The Grimm Audio LS1 speakers with the SB1 was the successor of the original LS1s-DMF motional feedback subwoofer. The new design took the Motional Feedback approach to a whole new level with an improved driver, a significant evolution in sensors and DSP, creating an even better match for its LS1 series speakers.

even lower latency, with onboard A-D, D-A and a DSP engine. That DSP makes the design so much easier. For instance, when you go up in frequency you get to the point where the surround dynamically uncouples itself from the cone. That means that the radiating surface and mass has a jump, which you see in the frequency response. If you wanted to correct that with feedback, you need more loop gain at that higher frequency, which means more bandwidth and stability issues. With DSP I can set a notch at that surround decoupling frequency, and don't need loop gain there.

Another handy DSP function is a limiter, because when the system clips, the feedback only would make it worse. A bit of DSP limiting can prevent that.

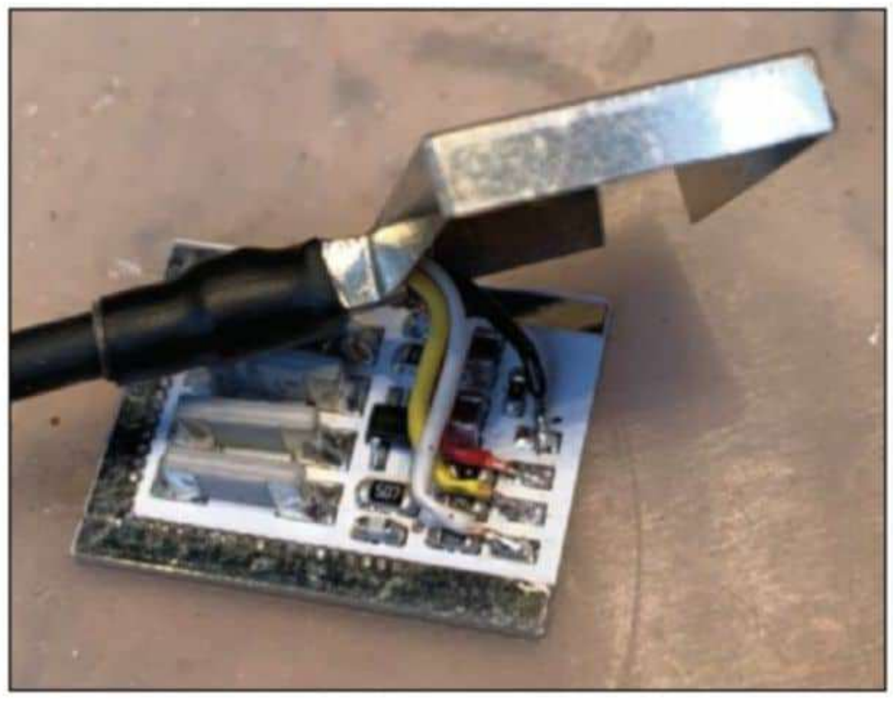


Photo 5: The digital motional feedback sensor showing the four paralleled MEMS sensors

JD: Did you commercialize that design in any way?

RMS: Well, at about that time, two things came together. One, I ran into my cousin Eelco Grimm and Guido Tent of Grimm Audio [2] who asked if I was interested in designing an MFB subwoofer for their LS1 speakers (**Photo 3** and **Photo 4**). To put it on a commercial footing, I founded my company in 2012 and started to give courses on "Mechatronics Design" and the design of mechatronics-based audio systems. Eelco and Grimm Audio became my first customers for MFB.

There was one particular problem that was difficult to solve. Due to the forces on the cone at high levels and low frequencies, the sensor started to deform and generate output that disturbed the control system. That lead to the "Guido (Tent) bolt." The sensor was mounted on a piece of carbon sheet attached to the cone. Guido's idea was to put a small conical-head bolt into the carbon sheet and glue the sensor on the bolt head. That worked great! The coupling between sensor and cone was good and deforming forces on the cone were no longer transmitted to the sensor.

JD: What exactly do you want to sense, the voice coil movement or the cone movement?

RMS: Both really, it's a matter of finding the best spot. And if you get to the frequency range where voice coil movement and cone movement start to decouple, you shouldn't use it anyway. That's the upper frequency limit where it can be used. But generally, it is best to mount the sensor as close as possible to the voice coil.

The early models I developed for Grimm had two bridged 400W UcD amplifiers with one 400W power supply. I finished two prototypes a month ahead of the Munich High End show. So we went to Munich, and it sounded all great, but at that show demos are always very loud. We tried to match that, and then intermittently we got silence as the power supply overloaded and had to be unplugged from the mains to be reset.

These prototypes were already current driven to avoid the distortion due to eddy currents in the magnet core and maximize the peak at the resonance frequency for higher loop gain. Because of the bridge configuration however, we needed a special IC to sense the current through a series resistor floating from ground. Unfortunately, that caused a lot of noise, requiring a too large sense resistor value. It would have to be a small resistor to avoid power loss; but this amplification needs to add as little noise as possible.

JD: Why is that noise such an issue? It's all low-level control, isn't it?

RMS: Noise is a real issue in motional feedback because it cannot be attenuated by volume control! You have to compete with amplifiers with 130dB dynamic range. With MFB the sensor determines the sound and its noise creeps into the control loop and ultimately into the drive signal. In production we settled on a single 700W Ncore amplifier with a 1200W power supply and a small sense resistor to ground, followed by a very low noise op-amp.

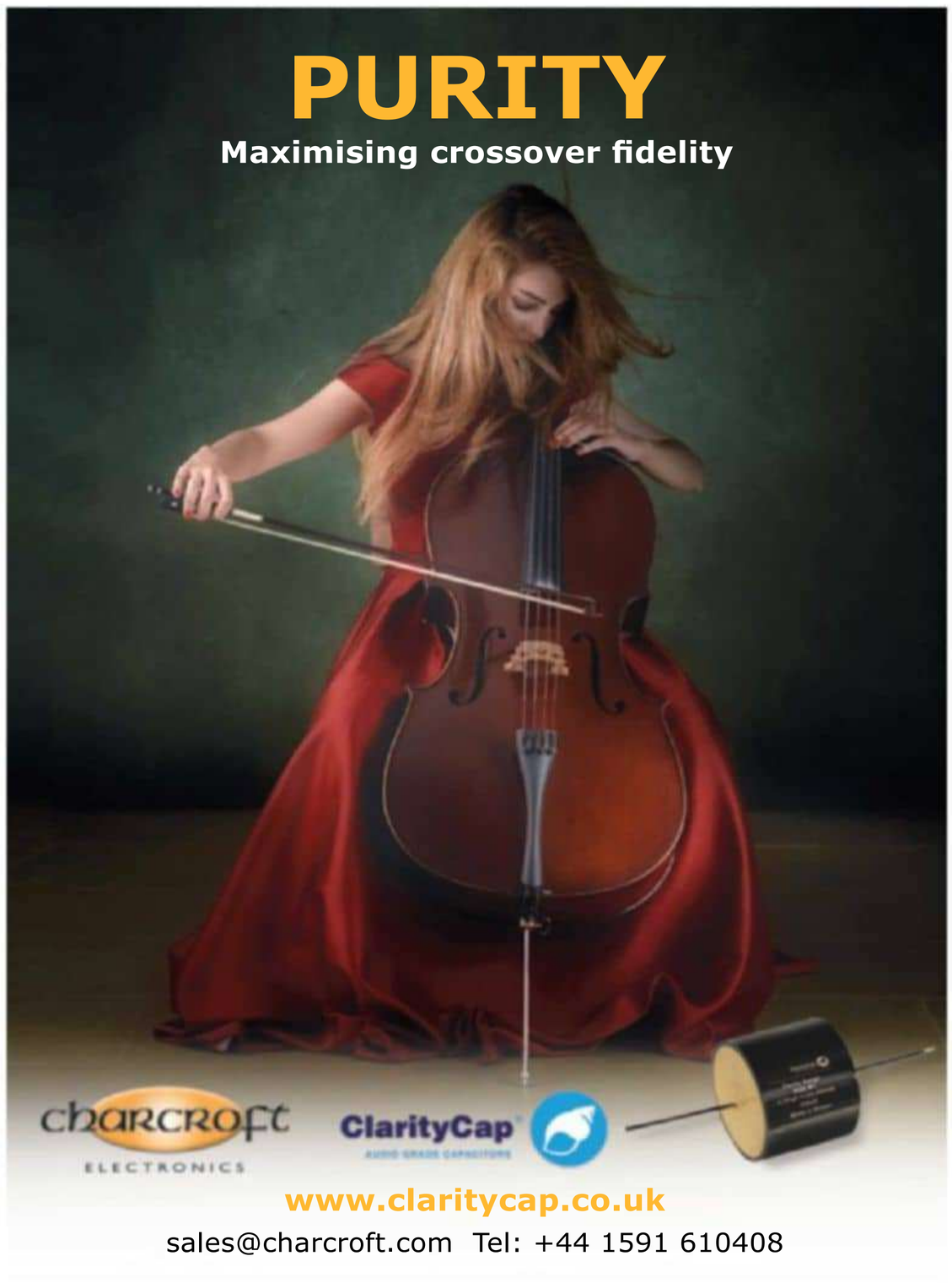
Now we get to the final act, the sensor. The ACH-01 still bothered me. I got a batch where some were pressure sensitive and reacted to the pressure inside the enclosure. At that point in time, I was actively involved in discussions on MFB in Hans Mons' forum MFBLabs.nl. It was there that Chris Camphuisen from Piratelogic.nl discovered acceleration sensors as used in Murata hard disks [3]. These are very small sensors, manufactured with MEMS technology in small SMD form factor (**Photo 5**).

In the end I decided to manufacture my own sensors, using four of those MEMS parts in parallel to get a better noise figure, in a charge amplifier setup. And these are so good that it doesn't really matter at what orientation they are mounted, which makes modification of a woofer much simpler: One can even glue the sensor at the back of the cone, close to the voice coil, and that's it (**Photo 6**)! The connection cable is very supple to avoid mechanical interference. As you see the sensor assembly is electrically shielded because of the high impedance capacitive nature of the sensor elements.

JD: I learned that Hypex was considering adding an MFB subwoofer to their arsenal. So, they have their work cut out for them!



Photo 6: The MFB sensor mounted on the back side of a woofer cone





RMS: Yes, and I give them a hand. It is still in the early stages; I don't know when or in what shape it will materialize.

JD: What about the stability of MFB systems? Some designers seem to struggle with the idea for that reason.

RMS: In the past, the main problem with MFB systems was that they could become unstable over time when driver parameters change. That is no longer an issue. In a competent design, the phase and amplitude margins in the feedback loop can be made sufficiently large there's never going to be a problem. I have systems here with 25-year-old sensors that play absolutely flawlessly.

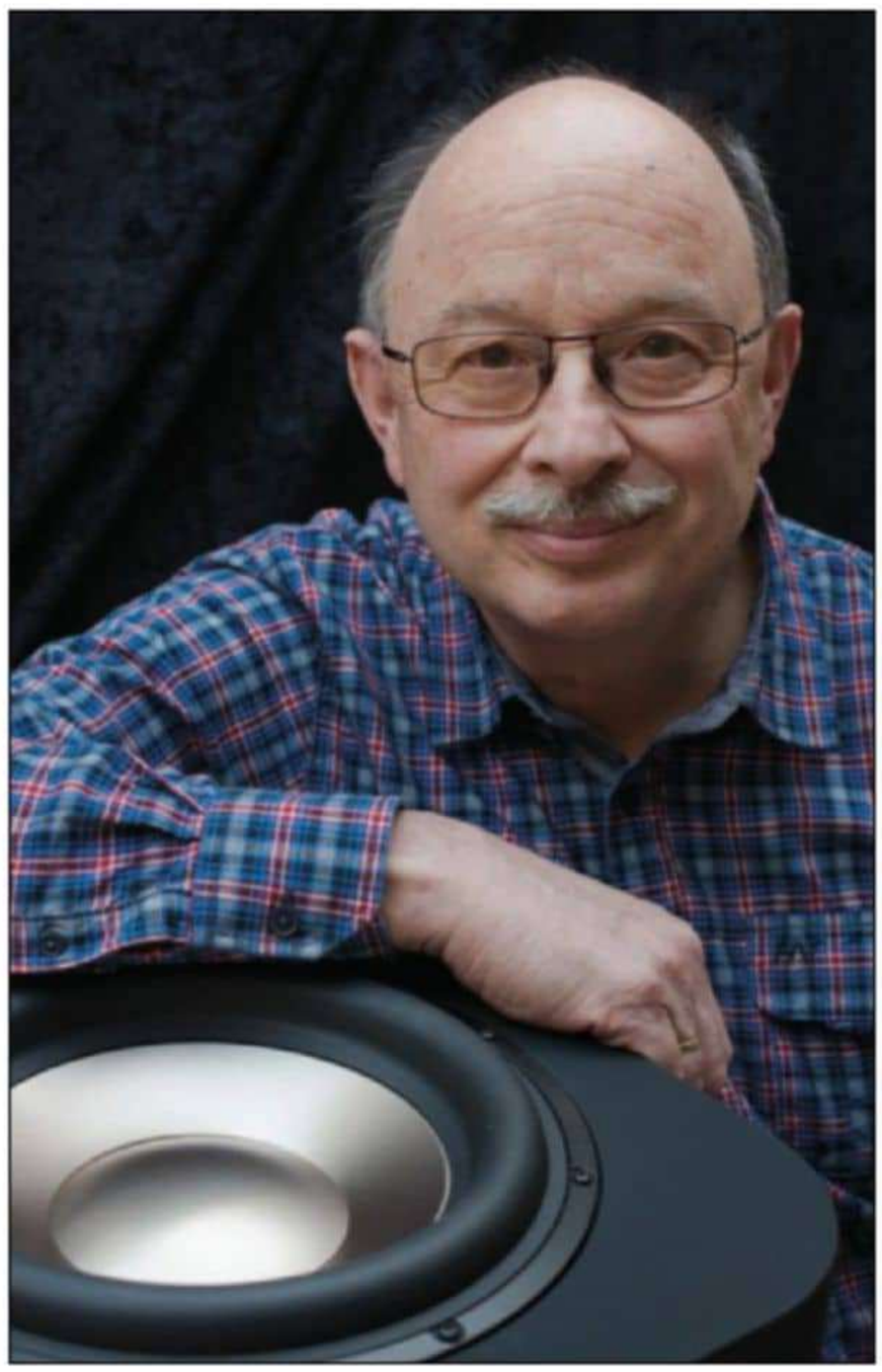


Photo 7: High-quality sound reproduction and controlled motion systems have always been a source of curiosity and inspiration for Robert Munnig Schmidt. He studied Mechanical Engineering at the Delft University of Technology, while simultaneously improving his skills in electronics and electromagnetics and making loudspeakers with amplifiers as a hobby. He retired from his business activities in 2021.

JD: On your website, you are quite critical about bass reflex enclosures. What's your take on that?

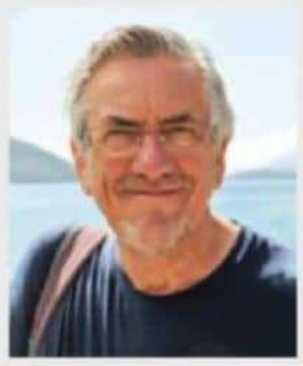
RMS: You know there's something strange going on in audio. We talk about frequency response, but that doesn't exist! It's a mathematical translation of the time domain response through techniques like Fast Fourier transformation. ASML's Martin van den Brink always called us "the Hertz mafia." Without mathematics, a frequency response can only be constructed by using a continuous sinusoidal signal (time domain!), then wait until everything is stabilized, write down the response, then change the period of the signal, again record the response, and so on. You then construct a response graph with the magnitude on the vertical axis and the inverse of the period on the horizontal axis, which is defined as the frequency of the signal, by connecting all the dots of those individual measurements. With mathematics, we can calculate how it looks even from noise or impulse responses. And then we think that if we have a flat frequency response, all is well. But look at a waterfall plot; if all was level, the response would be bounded by straight lines, but that's never the case! There's always ridges, peaks and throughs.

And that brings me to the bass reflex enclosure. A bass reflex enclosure causes (at a certain frequency) the cone to come to a standstill while the resonance in the reflex tube causes acoustic output. That is, however, a static situation. In reality, if you start a signal, the first thing that happens is that the cone starts to move with a very large excursion because there's no air resistance! Then the air starts to move, and the reflex tube starts to absorb energy and gets going, and eventually the cone comes to a standstill. When the signal dies out, the cone starts to move again and the cone and resonator both move, and you get oscillating air streams and ringing. With resonating systems, you store energy in mass, and that energy has to be released. You should listen to Victor Wooten & Marcus Miller's "Miller Time" on *Live in America* over an MFB system: you hear the strong, dry transients beautifully. Play it over a bass reflex and the sharpness of the transients is gone. Bass reflex is a principally wrong concept.

JD: You are what we would call a purist.

RMS: Everything should be measurable, and there are still factors we don't know yet how to measure. It reminds me of the discussion you had with Jan Lohstroh [4]. At the time, we hadn't discovered transient intermodulation distortion in audio but we knew "something" was going on; by measuring and testing we develop methods to quantify it.

About the Author



Jan Didden has written for *audioXpress* since the 1970s and he is the magazine's Technical Editor. He is retired following a career with the Netherlands Air Force and NATO. He worked in logistics, air defense, and information technology. Retirement has provided him with the time to finish all the audio projects that have piled up for decades. He sometimes writes about them on his website linearaudio.nl. Didden is also known as the publisher and managing editor of the twice-yearly bookzine *Linear Audio*, which he published until 2018. He is currently teaching audio electronics for enthusiast hobbyists and serious amateurs in the Belgian city of Hasselt. More information is available at linearaudio.net

JD: You see a trend to much better designed and manufactured drivers like from Purifi, which have much less parameter spread between units, so they can be characterized more accurately. Is there a future where you design MFB systems based on feedforward techniques?

RMS: If possible, yes, it would seem to be less costly than to turn a woofer into an MFB woofer. Then again, those Purifi drivers aren't cheap. It's a choice between a $200 driver with a $100 MFB sensor system, or a $500 driver with feedforward. So, it's not a clean-cut situation.

It is often suggested to use a real MEMS sensor for MFB as used in smartphones and such. They are cheap, include electronics and work from DC up to sufficiently high frequencies. Unfortunately, their sensor noise is still not solved, especially at higher frequencies. That's the advantage of a capacitive piezo sensor, where the noise rolls off with frequency.

If you work with smaller speakers, the cone displacement for the same SPL is larger, which means that the signal-to-noise ratio of a MEMS sensor might be just OK. But with large diameter subwoofers, sensor noise still remains an issue.

JD: Professor Munnig Schmidt (**Photo 7**), thank you very much for your time and the fascinating insight in motional feedback. I'm sure we will hear a lot more about it!

RMS: You can be sure about that. And thank you for the chance to talk about it!

For more information about RMS Acoustics & Mechatronics (https://rmsacoustics.nl).

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The Role of Resonance in Audio Clarity



State-of-the-art high-speed winding machine in clean room

By
Roger Tall
(ClarityCap)

Years of research have shown that resonance from the body of a capacitor in a loudspeaker crossover circuit has an audible impact on sound quality. The article details the extensive testing, measurements, and conclusions, which allowed perfecting a manufacturing process minimizing resonances in capacitors.

It all began with a two-year Knowledge Transfer Partnership (KTP) research program, by audio capacitor manufacturer, ClarityCap, and the UK-based Institute of Acoustics at the University of Salford, which assessed the effects of loudspeaker crossover components on sound quality. The conclusion was that there were no electrically measurable differences between standard industrial and audio-grade capacitors.

Although this result was surprising, the research continued and determined that it was capacitor resonance in the crossover circuit that had audible impact on sound quality.

The original research was comprehensive as well as thorough. Tests were performed on more than 350 metallized polypropylene film capacitors. These included a mix of audio and standard industrial capacitors that are available on the open market, as well as custom capacitors across a range of design parameters.

Each individual capacitor was subjected to a total of 819 electrical measurements at frequencies up to 40kHz. The goal of these tests was to determine how frequency changed the capacitor parameters, which included equivalent series resistance (ESR), capacitance and the tan delta loss factor. Yet these tests did not provide conclusive results.

The ESR of all the tested capacitors was measured at 5mR to 15mR at 1kHz, which is negligible in a typical loudspeaker crossover circuit and wiring. There were also no observable electrical anomalies or deviations from the predicted values, and the capacitance and tan delta also followed the predicted curves. Finally, there was no significant variation in the component aspect ratio, film thickness or even the cost.

Researching Resonance

After working closely with several loudspeaker manufacturers, ClarityCap and the Institute of Acoustics continued the research to understand how component properties in the loudspeaker crossover affected sound reproduction.

The crossover circuit is used to divide the incoming audio signal of a loudspeaker into sub-ranges of the audio spectrum. The different types of crossovers are determined by the slope and roll-off characteristic of each crossover output.

During the research it was discovered that the application of voltage, transients, and AC current through a capacitor can cause the windings to resonate.

This resonance was measured using a swept sine wave, and impulse excitation, with an instrumentation microphone and was seen in the upper audio frequency band between 10kHz and 30kHz. The resonance by frequency is shown in **Figure 1**. The mean amplitude varied depending on which capacitor was being tested and, for most capacitors, the emissions were clearly audible.

The research team then used a laser scanning vibrometer to scan multiple points over the surface of the capacitor to provide accurate measurement of surface displacement. The three-dimensional graphs that were generated represented the movement of the surface. By overlaying these graphs onto images taken from a video camera, it was possible to identify the parts of the capacitor that were experiencing surface disturbance.

The presence of a resonant mode was shown by the nodal point in the middle of the capacitor body, in **Figure 2**, and by the areas of the capacitor body that demonstrated maximum surface displacement, shown in **Figure 3**. While the earlier research demonstrated no difference in electrical or acoustic measurements, scanning with a vibrometer showed that mechanical resonance does vary across individual capacitors.

Transforming Resonance Control

The resonant signature of each capacitor is determined by the physical dimensions as well as by the properties of the material and the processes used to manufacture the capacitor. The next stage of the research was to assess whether the mechanical resonance delivered a change in the quality of the sound that was reproduced.

The difference in sound produced by two different metallized film capacitors in the high-frequency driver circuit of a first-order crossover is very slight. It is possible that this difference could only be detected by a listener trained to critically evaluate and develop audio systems. Many hours of listening tests were essential to verify the impact of mechanical resonance on the quality of sound reproduction.

For these tests, ClarityCap and the University of Salford Institute of Acoustics completed a total of 33 blind listening tests. The results showed that the capacitors optimized to reduce resonance reproduced sound with greater clarity and spatial information.

Experiments in design, as well as in manufacturing processes, identified two production variables that were differentiators in minimizing capacitor resonance. ClarityCap used the research to develop an optimized series of capacitors, which were designed

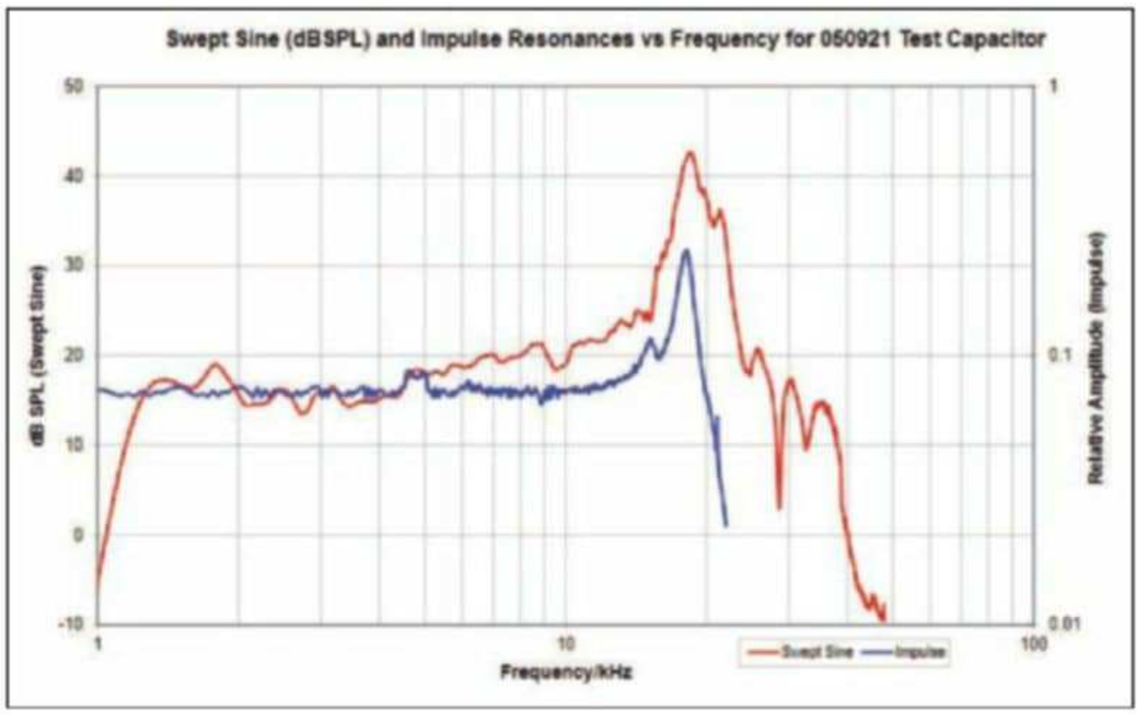


Figure 1: Swept sine (dBSPL) and impulse resonances by frequency

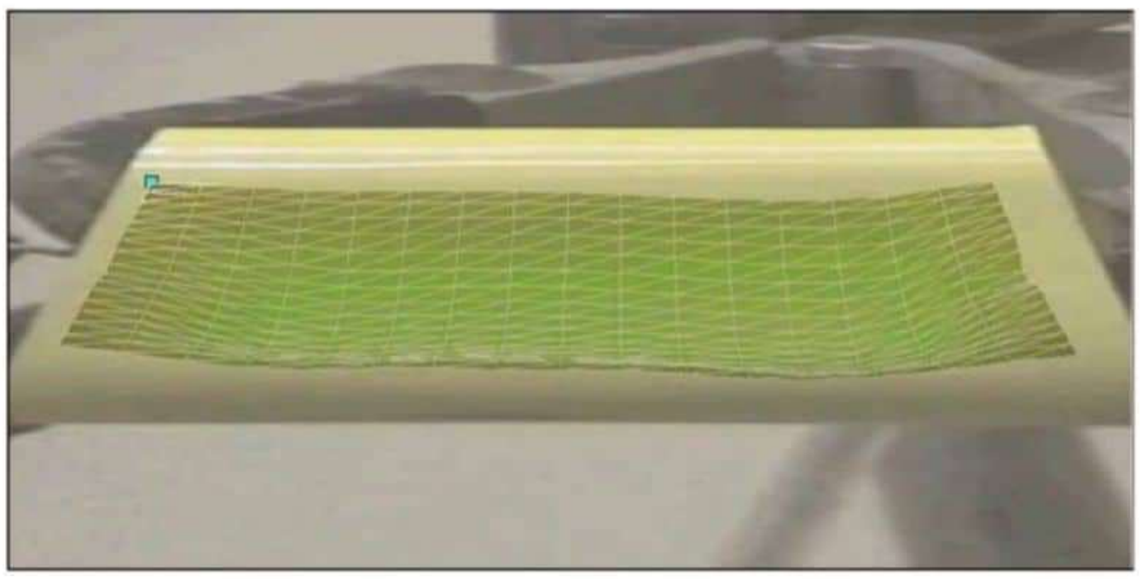


Figure 2: The nodal point in the middle of the capacitor body

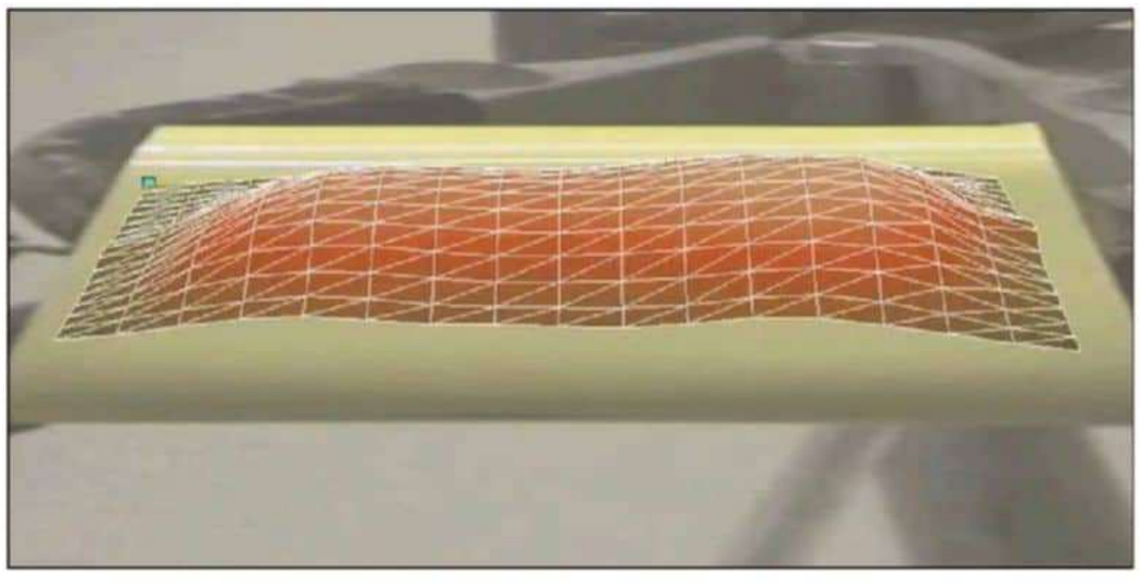


Figure 3: The area of maximum surface displacement on the capacitor body

About the Author



Roger Tall is a director of ClarityCap and Charcroft Electronics and has had an interest in audio and music from an early age. In the 1980s, he created mix-tapes, followed by sound-engineering live events, and producing optimized, well-mixed, and balanced sound using analogue and digital mixing desks. As well as attending local live-music concerts, Roger admits to having three hi-fi systems at home. The technical support that Roger gives to manufacturers of high-end audio equipment starts with selecting the right capacitor for each application, and adds lateral thinking to follow the development through to the launch and beyond.

to reduce resonance. The SA series of capacitors already exhibited low resonance, but this was further optimized to achieve a reduction in resonance of 4.3dB, and the introduction of the Enhanced SA (ESA) series of capacitors.

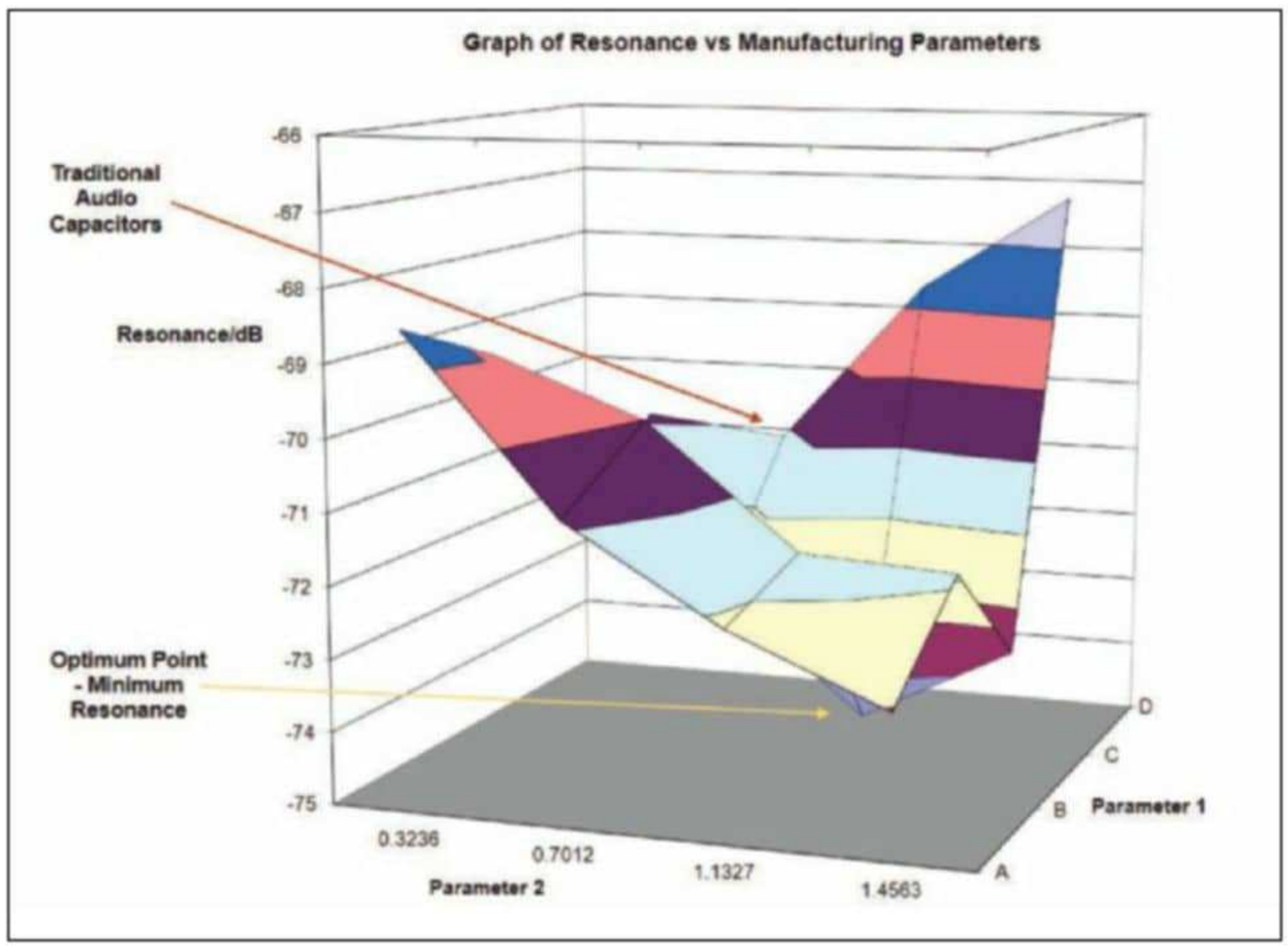


Figure 4: Comparison of resonance in a standard capacitor and an optimized ESA capacitor

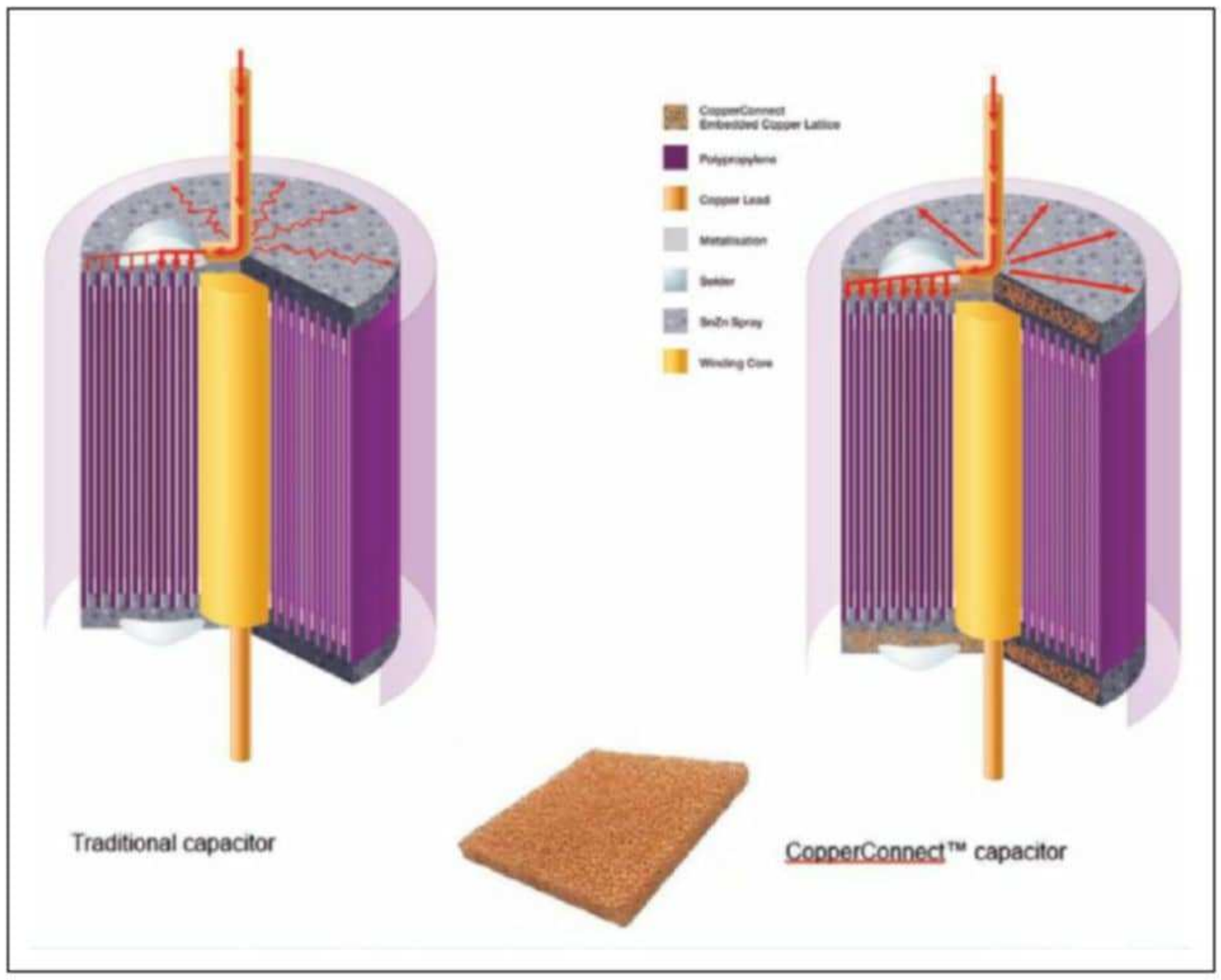


Figure 5: CopperConnect lattice technology

Resources

"Capacitor Resonance," Charcroft Electronics, Ltd., www.claritycap.co.uk/research

A standard manufacturing process was followed, with particular focus on the production variables. The pure aluminum metallization was applied to a roll of polypropylene film with a vacuum deposition system, and the roll was slit for winding on a high-speed winder in a 5µm filtered clean room. A metallic contact layer was applied, known as schoopage, before the capacitors were heat-treated to remove areas of stress in the winding and expel any remaining air-pockets, The process also hardened the winding once it has cooled back to room temperature.

Defects in the winding were removed by the clearing process in which high voltage is applied. This cleared areas that had an inherent issue in the film or metallization, or an impurity that had been picked up during the winding or building process. The clearing process was used to effectively vaporize a very small area of the metallization. The next stage was to select a metallization pattern that could minimize resonance.

The pattern of the metallization is typically adjusted for different voltages, current-ratings, or ESR, as well as to increase safety by functioning as a fuse. In a standard winding, one metallized film is wound on top of another to create two parallel conductors with a separate dielectric. However, ClarityCap used a metallization pattern that is widely used in industry, but not for manufacturing audio capacitors. This had the effect of breaking up the active sections of the capacitors into smaller sections, and reducing the overall resonance, as shown in **Figure 4**.

The metallization pattern was used to produce the Minimum Resonance (MR) series of capacitors and the result was that reduced resonance was delivered, with minimal impact on the ESR value.

The Role of Resin and Copper

While minimizing resonance was the start of transforming the delivery of audio quality, other technologies were added to improve signal propagation in addition to reducing distortion and phase shift.

Hardness testing in an Aeronautical Laboratory showed that a standard polyurethane resin can provide some damping but remains too soft to restrict the displacement in the capacitor. The use of an epoxy resin provided the best results, providing a lower level of damping but enabling the capacitor to increase audio quality by achieving excellent restriction of displacement.

Another new development was the addition of a 1mm copper lattice embedded at both ends of the capacitor. The CopperConnect technology improved

the electrical interface between the capacitor lead and the capacitor electrodes, which are Zinc sprayed. This delivered a reduction in the grain boundary crossings at the ends of the capacitor, and the introduction of narrower films delivered lower ESR. Another benefit was that the 1.0mm^2 copper leads were oxygen free.

The minimum-resonance metallization pattern and CopperConnect technology were initially combined in the CMR capacitors, and then in the latest Purity capacitors, which added more enhancements, as shown in **Figure 5**.

In addition to extending the CopperConnect lattice to a 10mm thickness, the Purity capacitors achieve voltage up to 800V DC and add a silver-coated multi-strand Van den Hul cable, see **Figure 6**. This cable is already used by many high-end speaker manufacturers and provides an easy path for high frequencies to reach the tweeter to deliver accurate audio reproduction.

The combination of these innovations means that Purity capacitors increase the tweeter drive and transform the reproduction into a more nuanced treble. In early listening tests it was assessed by audio engineers that Purity provides a significantly lower noise floor, with more "dark space" around the audio signal, and an improved dynamic range. These features provide greater clarity in the reproduction of sound as well as improved spatial separation of individual instruments and vocals. A standard ±3% tolerance also enables tight crossover control.



Figure 6: Purity capacitors combine minimized resonance with CopperConnect

The new capacitors are already being used in crossovers by several high-end loudspeaker manufacturers. Development is ongoing, and more innovations will follow to ensure that capacitors continue to play a vital role in delivering the highest achievable levels of audible performance.

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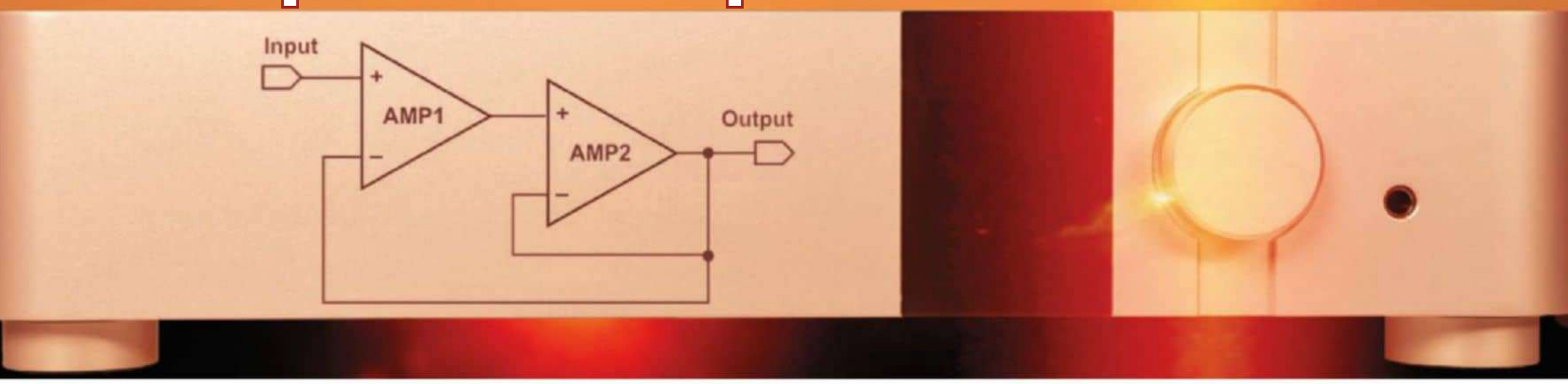
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Composite Amplifiers



This article presents a fairly in-depth analysis of composite amplifiers, including the effects of local as well as global feedback. In his exploration the author looks at stability in detail and debunks a few rules of thumb, before applying clever measurement techniques to measure the very low distortions.

By
By Paul Marchese

Any additional active circuitry on either the input or output of an operational amplifier (op-amp), or the addition of a second op-amp, either parallel or in series with the original one, is considered a composite amplifier. For the two-amplifier composite, in many cases this first amplifier is either a low offset or low distortion amplifier coupled into either a high voltage or high output current amplifier.

The composite amplifier's goal in this example will be to make a lower distortion, power audio amplifier. I decided to stick with linear amplifiers for the power amplifier, although the Class D, and other type amplifiers will probably also work. An immediate point to bring up is that of stability. Cascading amplifiers result in potentially enormous gains and hence the increased possibility of oscillations. A rule of thumb for a stable composite amplifier is that the second amplifier (A2) should be approximately three times "faster" than the first amplifier, A1, so not to add any significant additional phase shifts at the higher frequencies. This however is a general rule for global feedback that will be shown in the local feedback circuit to be a gross oversimplification. Also, the speed of the power amplifier is unfortunately usually less than the input amplifier. The way to compensate for this is to limit A1's gain bandwidth product (GBWP) via B1, but not in the in the global feedback B3.

While there are quite a selection of amplifiers that have the necessary qualifications for the first amplifier, A1, that is not the case for the power amplifier, A2. I chose the OPA1602 for A1 and the LM3886 for A2. The OPA1602 is a very low distortion audio amplifier. The LM3886 is a low distortion amplifier that can deliver approximately 50W of power into an 8Ω load. There are two negative aspects of A2 with respect to A1. One: A2 is slower than A1, with a gain bandwidth product (GBWP) of 8MHz versus 35MHz. And two: A2 is not unity gain stable. Both of these negative aspects can be overcome with the correct feedback.

Figure 1 shows this composite amplifier where the first stage is inverting, and the second stage is non-inverting and the minimum necessary feedback components. C1 is to limit the gain of A1 at higher frequencies and the R2s set the local gain of A2 to at least 5V/V. R3 is to set the overall composite amplifier gain and C2 is for transient tweaking.

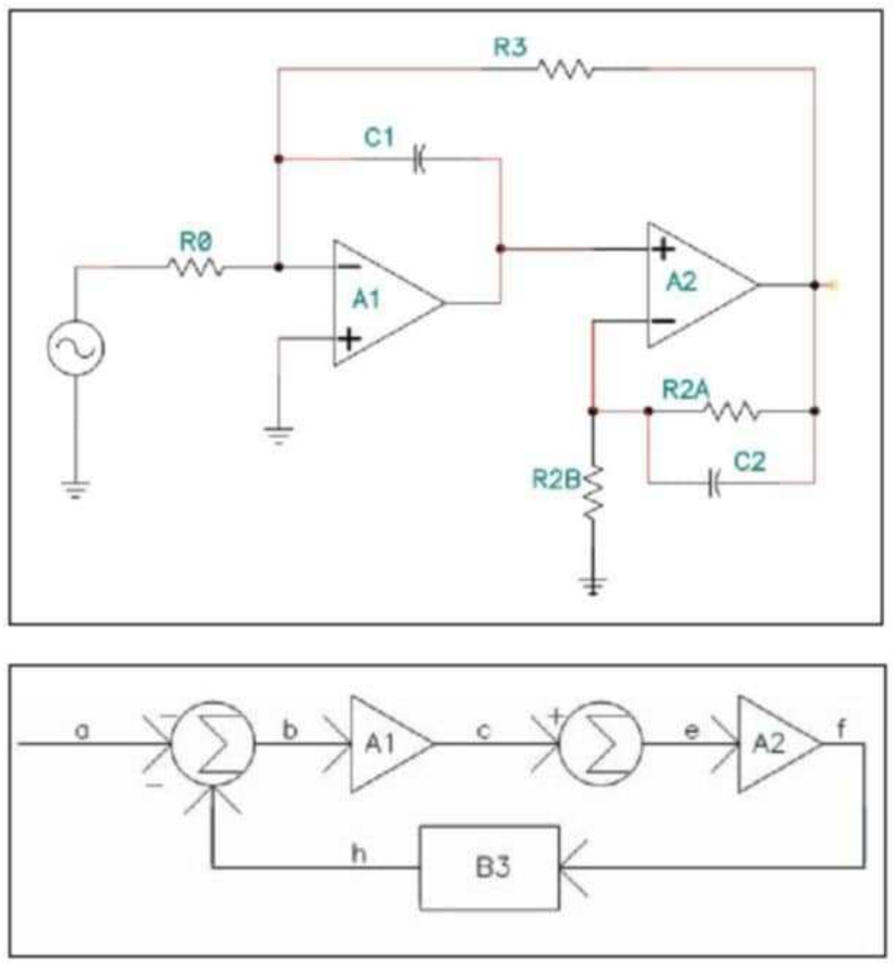


Figure 1: An inverting composite amplifier with minimum feedback components

Figure 2: The feedback block diagram of the composite amplifier

To get a feel of the overall operation of a composite amplifier, let's first look at the circuit with just the global feedback, without the local feedback. There are two amplifiers, A_1 and A_2, with their summing nodes (the two sigmas) and the global feedback of B_3. The analysis is the same as in Black's classic paper [1] but with two amplifiers versus one. **Figure 2** shows a block diagram of the circuit.

From Figure 2: **f**=A_2e; e=c; c=bA_1; b=-**a**-h; h=**f**B_3
Isolating **f** and **a**, the output vs. the input, **f/a** is:

$$\mathbf{f/a} = -(A_1A_2)/ [1 + A_1A_2B_3] \qquad [1]$$

where $A_1A_2B_3$ is the loop gain (see the Loop Gain sidebar). Note: since both the amplification (A) and the feedback factor (B) are dimensionless, usually V/V in both cases, the loop gain can have any number or combinations of A_s and B_s, as will be seen with the local feedback circuit, and still be dimensionally correct. This transfer function, the output with respect to input, f/a, can be looked at in (at least) three different ways:

One: the expression best suited to examine the stability, Equation 1

Two: the exact equation, Equation 2

Three: the low/mid frequency simplified expression, Equation 3

Equation 1 is in good form for stability analysis using loop gain. When the right side of the denominator approaches -1, then the denominator nears zero, and the expression becomes very large.

Dividing the numerator and denominator of Equation 1 by (A_1A_2) reduces the complexity of f/a, but it is still exact.

$$\mathbf{f/a} = -1/[(1/A_1A_2) + B_3] \qquad [2]$$

At the mid-frequency of 630Hz, A_1 is =~ 95dB and A_2 is =~ 80dB and B_3 =1/20 = -26dB, so ($1/A_1A_2$) is 149dB less than B_3 and therefore negligible. The low/mid band approximation of Equation 1 becomes:

$$\mathbf{f/a} = -1/B_3 \qquad [3]$$

which is the ideal gain of the composite amplifier.

Now back to Equation 1,
f/a = -(A_1A_2)/ [1 + $A_1A_2B_3$],
where $A_1A_2B_3$ is the loop gain.

One way to analyze stability is to look at the rate of closure between the amplification, the product

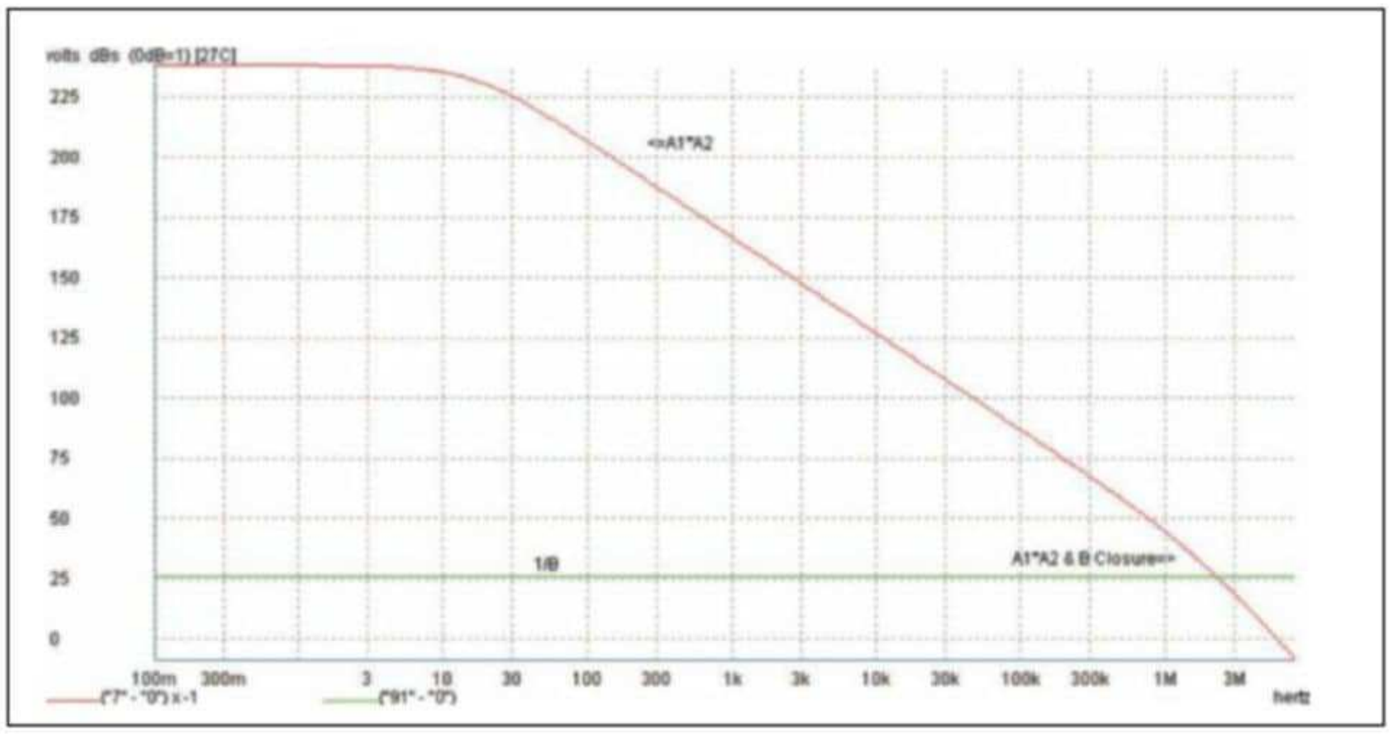


Figure 3: Open loop gain, A_1A_2, 1/Feedback Factor, $1/B_3$ vs. Frequency

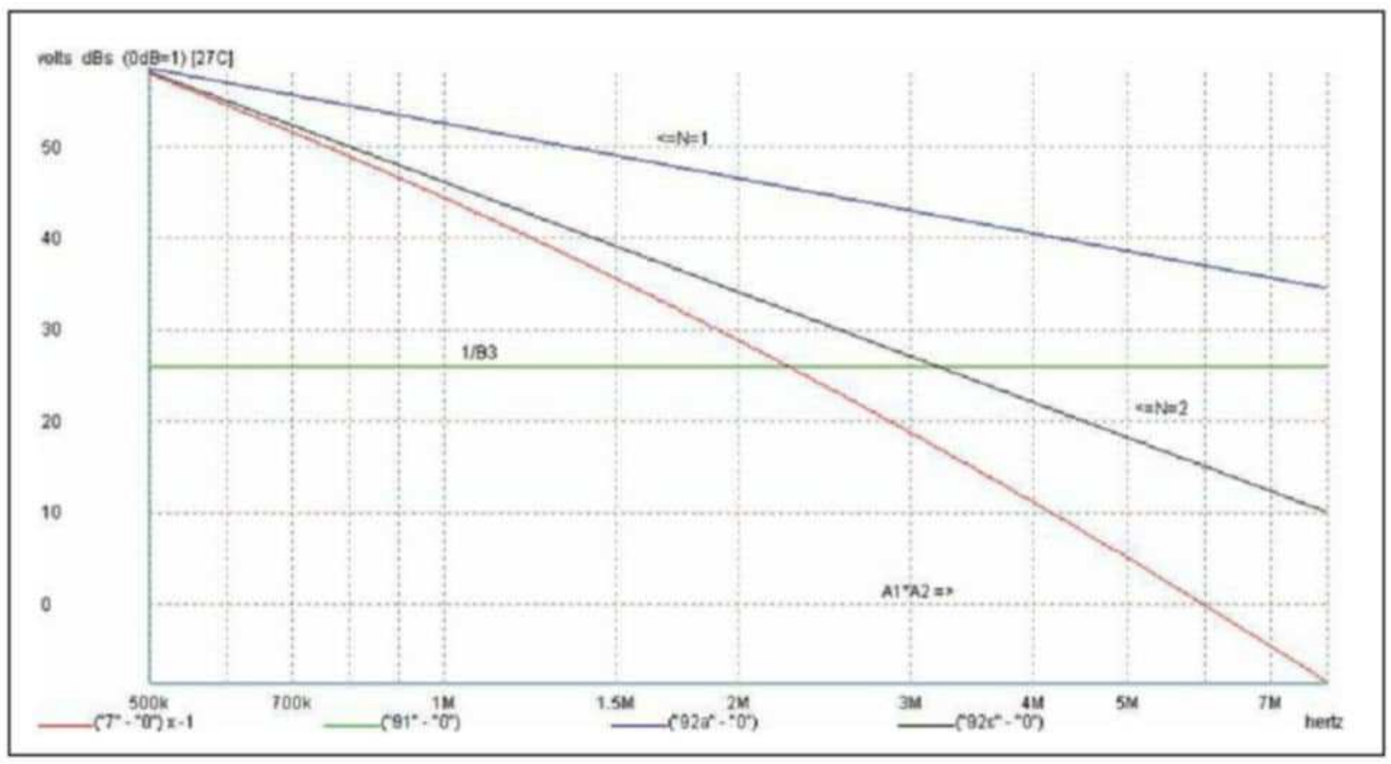


Figure 4: Open loop gain, 1/Feedback Factor, $1/B_3$ vs. Frequency

A Quick Look at Loop Gain

In a single amplifier with feedback, the loop gain is the difference between the open loop gain of the amplifier, A_{ol}, and the inverse of the feedback factor, B, as looked at on a Bode plot. Bode plots are logarithmic: decibel (a log) versus log frequency. The "difference" on the graph (**Figure A**) is a subtraction. Remembering that subtraction in logs is equivalent to division, so the difference between A_{ol} and 1/B on the Bode plot is really A_{ol} divided by 1/B, $A_{ol}/1/B$, which is $A_{ol}B$, hence the "loop gain."

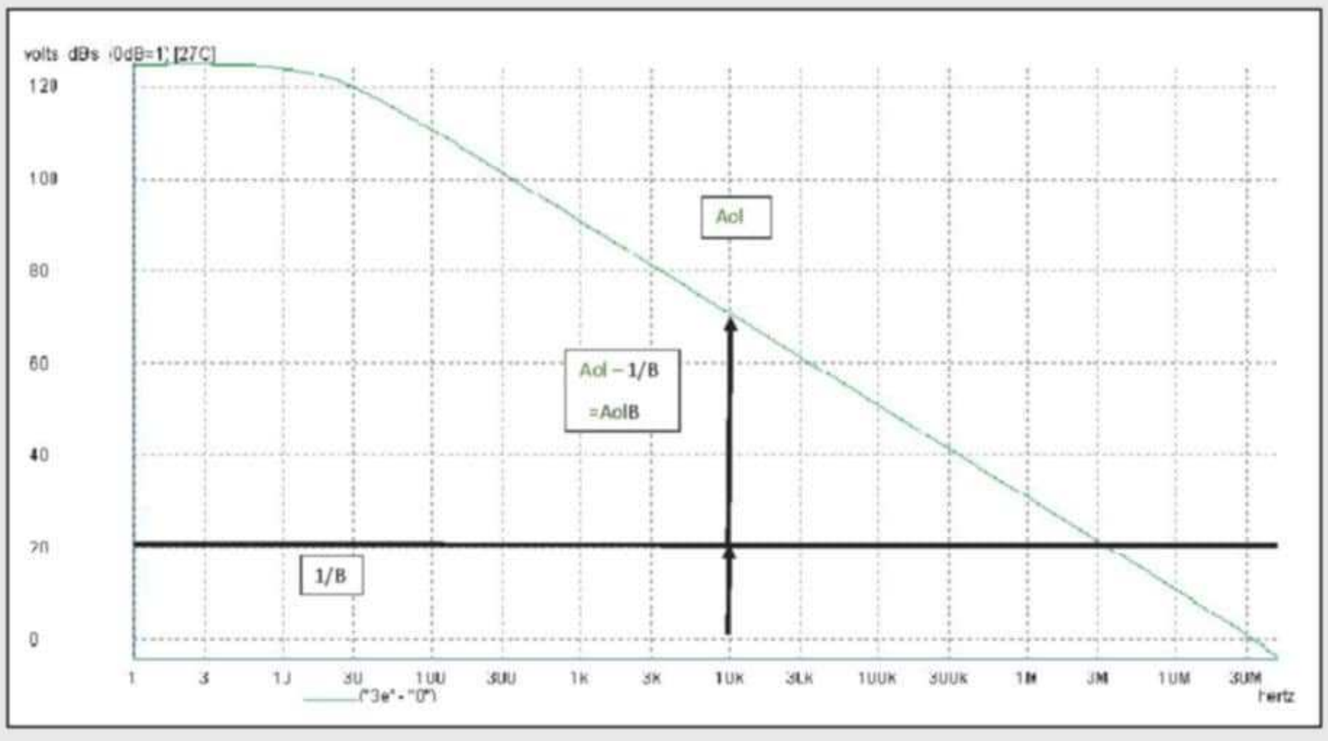


Figure A: A_{ol}:(Green), 1/B:(Black) and Loop Gain: A_{ol}-1/B (Black to Green)

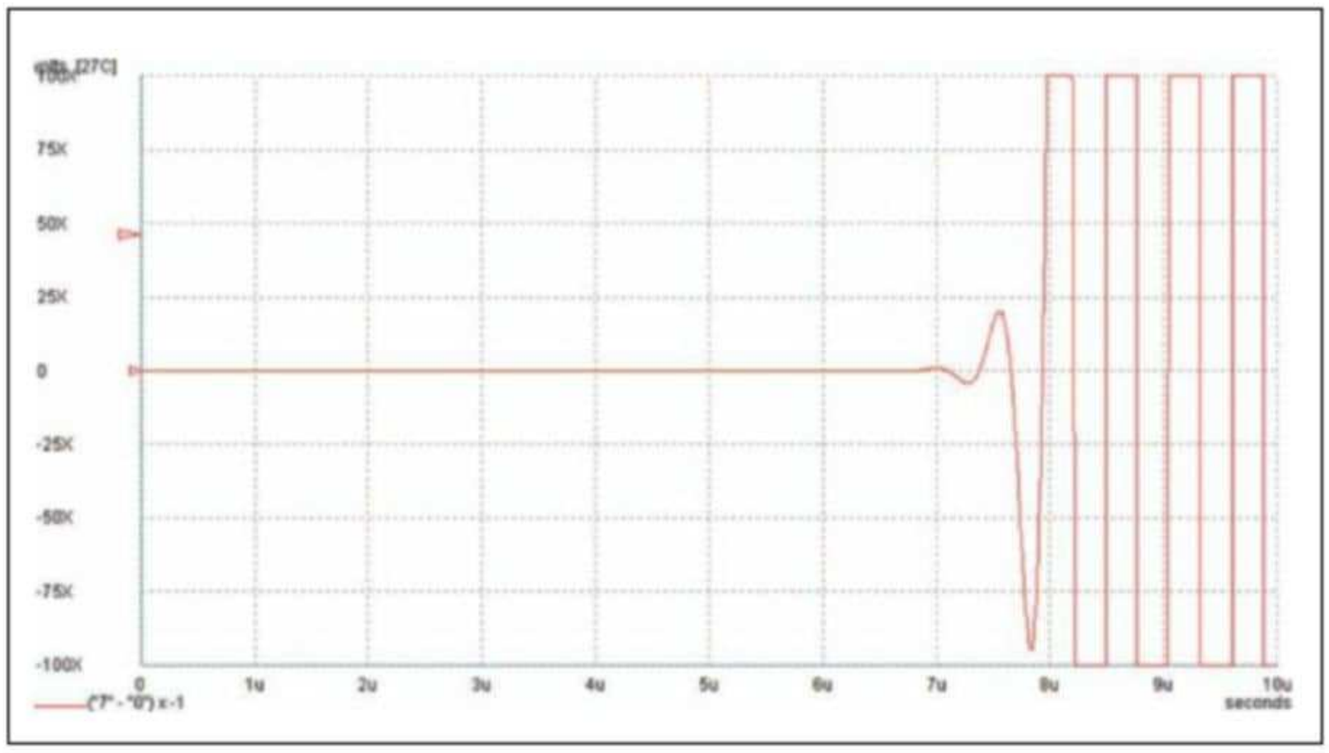


Figure 5: Circuit response to a step, Note: "X" = 10^{15}

Figure 4 is a blow up of the 500kHz to 8MHz frequency range with first order, n=1, and second order, n=2, slopes shown. As can be seen, the rate of closure between the amplification, A_1A_2, and $1/B_3$ is greater than second order.

The OPA1602 can be modeled with break frequencies at 19.7Hz and 94.7MHz. The LM3886 has break frequencies at 14.1Hz, 1.26MHz, and 7.96MHz. With 26dB of gain, the loop gain intersection point is ~ 2.25MHz, there is a combined total of 258 degrees of phase shift from the two amplifiers.

Figure 5 is the circuits response to a step. So even with a gain of 26dB, this circuit is unstable. Now let's look at the composite amplifier with both local and global feedback. **Figure 6** is a block diagram of that circuit.

Care must be taken not to get into an infinite feedback loop trap when "c" and "e" show up for the second time, so let's go through this step by step:

of A_1 and A_2 in this case, and the inverse of the feedback factor B_3. If this closure rate is second order, then the negative feedback has been shifted 180 degrees and is now positive feedback. Since both amplifiers have low break frequencies, 1/B will intersect at least second order, which is too high. There is a seven-page Application Note from Renesas [2], that explains in detail the operation of a composite amplifier with global feedback like this one, compensation strategies, with excellent graphs. Another good reference is the *IC Op-Amp Cookbook* [3].

From Equation 1, **Figure 3** shows the product of the open loop amplifier gains, A_1A_2, and the inverse of the feedback factor, $1/B_3$, and their rate of closure.

$\mathbf{f}=A_2e$ $e=c-g$ $c=bA_1$ $g=fB_2$ $b=-\mathbf{a}-d-h$ $d=cB_1$ $h=\mathbf{f}B_3$
$\mathbf{f}=cA_2 - gA_2$ $\mathbf{f}=bA_1A_2 - \mathbf{f}A_2B_2$ $\mathbf{f}=-\mathbf{a}A_1A_2 - dA_1A_2 - hA_1A_2 - \mathbf{f}A_2B_2$
$\mathbf{f}= -\mathbf{a}A_1A_2 - cA_1A_2B_1 - \mathbf{f}A_1A_2B_3 - \mathbf{f}A_2B_2$
Now, for the c term: c=e+g (avoids an infinite loop)
$\mathbf{f} = -aA_1A_2 - eA_1A_2B_1 - gA_1A_2B_1 - \mathbf{f}A_1A_2B_3 - \mathbf{f}A_2B_2$

Again, to avoid an infinite loop, $e=f/A_2$
$\mathbf{f} = -\mathbf{a}A_1A_2 - \mathbf{f}A_1A_2B_1]/A_2 - \mathbf{f}A_1A_2B_2B_1 - \mathbf{f}A_1A_2B_3 - \mathbf{f}A_2B_2$
$\mathbf{f}[1 + A_1B_1 + A_1A_2B_1B_2 + A_1A_2B_3 + A_2B_2] = -\mathbf{a}A_1A_2$

$$\mathbf{f/a} = -(A_1A_2)/ [1 + (A_1B_1 + A_1A_2B_1B_2 + A_1A_2B_3 + A_2B_2)] \qquad [4]$$

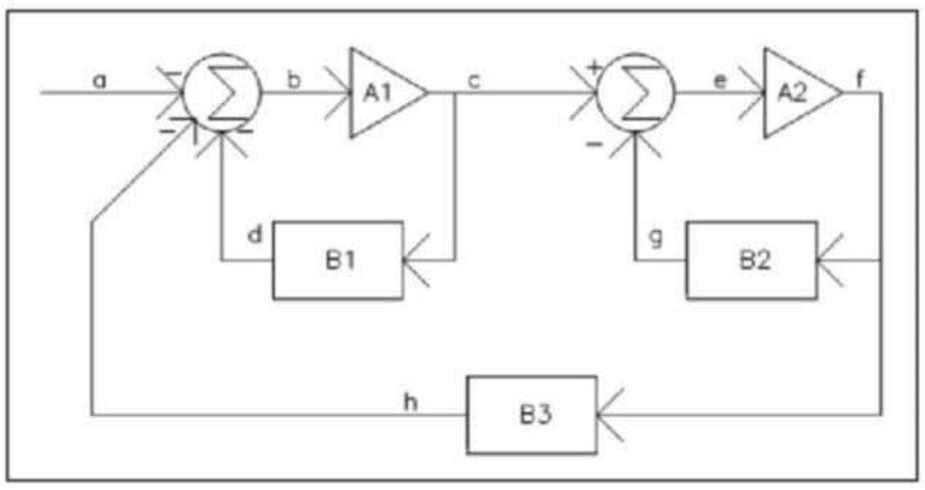


Figure 6: Feedback block diagram of composite amplifier

Again, this is the form of the equation that's useful in stability analysis. If the right side of the denominator is equal to minus one, the amplifier goes unstable. The loop gain, AB, is now made of four sections. This includes the two amplifiers with the global feedback as before, $A_1A_2B_3$, but also with the first amplifier with its feedback, A_1B_1, the second amplifier with its feedback, A_2B_2, and somewhat surprisingly the combined two amplifiers with their individual feedbacks without the global feedback, $A_1A_2B_1B_2$. This takes into account the path from (a) to (f) with local feedback but without global feedback. The loop gains now have 2, 2, 3, and 4 terms. So, the stability calculation goes from two multiplications with just the global feedback, to seven multiplications and three additions with the local feedbacks! Bottom line: having two "local" and one "global" feedback paths significantly complicates the operation of the composite amplifier.

Again, dividing the numerator and denominator of Equation 4 by (A_1A_2) reduces the complexity of f/a, but it is still exact.

$$\mathbf{f/a} = -1/[(1/A_1A_2) + B_1/A_2 + B_1B_2 + B_3 + B_2/A_1] \qquad [5]$$

One way to more easily analyze the exact expression [Equation 5] is to put the feedback terms in their order of significance with respect to their absolute value. You

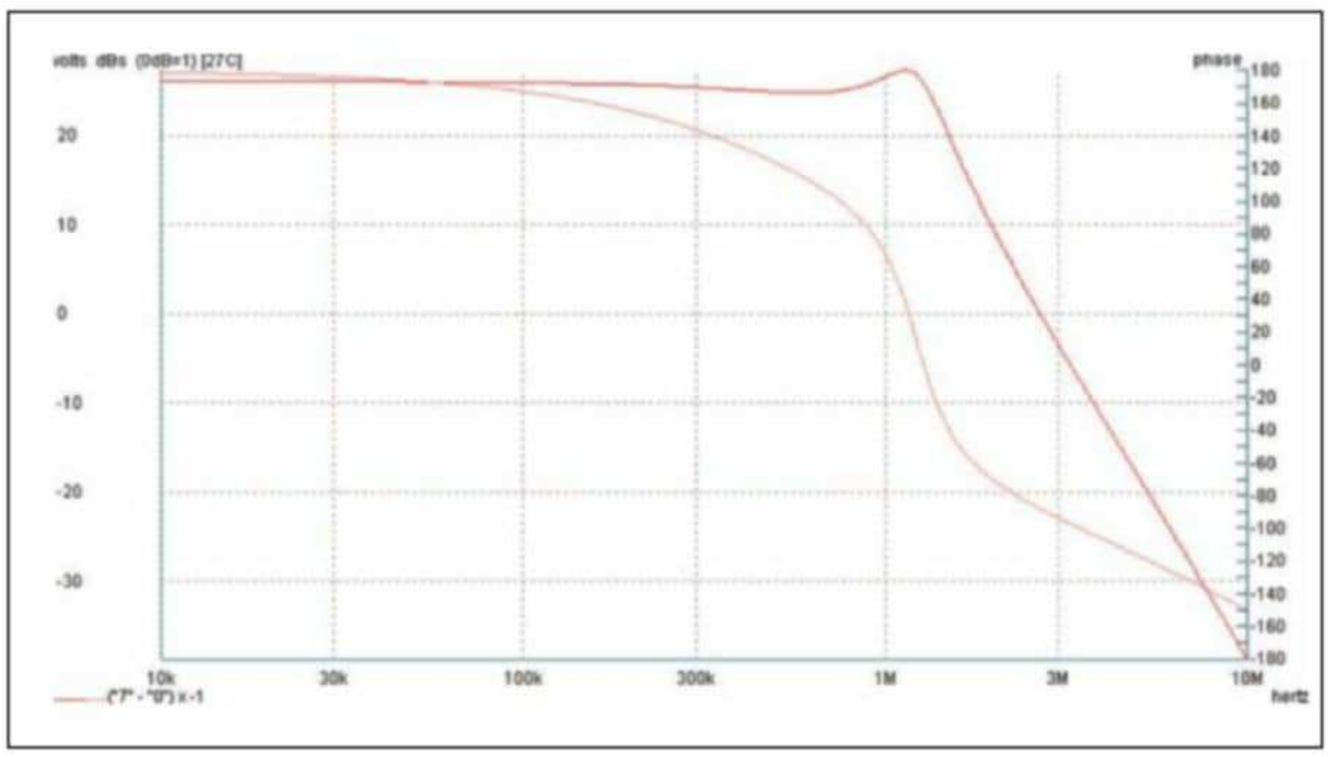


Figure 7: Gain in decibels (red) and Phase (orange) vs. Log Frequency from the Analog Simulation

can isolate which terms are significant by doing this but will lose the phase information. Using the mid-band frequency of 630Hz as a reference, in this example the two amplifiers gain will still be large, A_1=~ 95dB and A_2=~80dB, B_1 is very small, ~-69dB, B_2=1/5 (-14dB) and B_3= 1/20 (-26dB). So, the dominant terms are B_2 and B_3. But B_2 is either multiplied by B_1 (very small), or divided by A_2 (very large), leaving B_3 as the dominant feedback term.

$f/a = -1/[B_3 + B_1B_2 + B_2/A_1 + B_1/A_2 + 1/(A_1A_2)]$ [5a] in decreasing terms of significance
[-26 -83 -109 -140 -175] Values in decibels of the terms in [Equation 1a] at 630Hz

With respect to B_3, the next nearest contributor is down 57dB, so neglecting the smaller terms, $f/a = -1/B_3$ [6], which is the gain of an ideal composite amplifier. At higher frequencies, say 1MHz, the transfer function, f/a, is quite different than at 630Hz.

f/a = -1/[B3 + B1B2 + B2/A1 + B1/A2 + 1/(A1A2)] [5a]
[-26 -19 -45 -19 -62] Values in decibels. Now the second and fourth terms are dominant.
[-26 -83 -109 -140 -175] For comparison, values from 630Hz

Now the second term (both local feedbacks) and the fourth term (first amp's feedback and second amp's gain) are dominant.

In this example A_1, A_2 ,B_1 and B_2 are complex and a function of frequency. B_3 is real and constant. The overall gain was set to 20V/V (26dB) as in a typical power amplifier, but it can be set to almost anything above 5V/V, within reason. The amplifiers can be modeled as:

$$A_{ol}/[(1+jf/f_1) (1+jf/f_2) (1+jf/f_3)]$$

where A_{ol} is the open loop DC gain, f_1, f_2, and if needed f_3 are the break frequencies, $j = \sqrt{-1}$ and f is the test frequency. B_1 is $[j2\pi fR_0C_1]$. B_2 is $1/\{ 1+ 1/[(R_{2b}/R_{2a}) + j\ 2\pi f C_2] \}$.

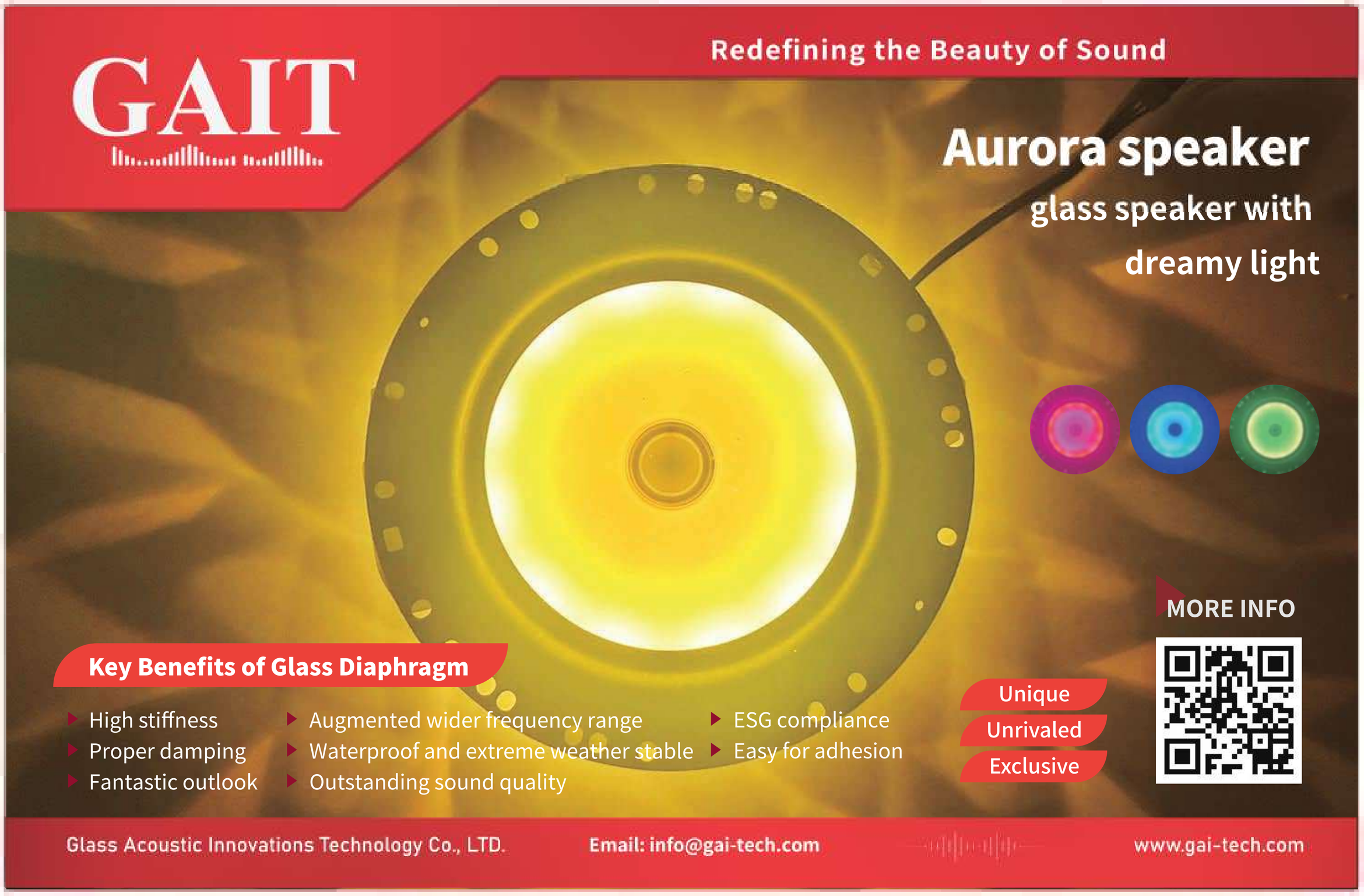


Figure 8: Gain in decibels vs. Log Frequency from Equation 5

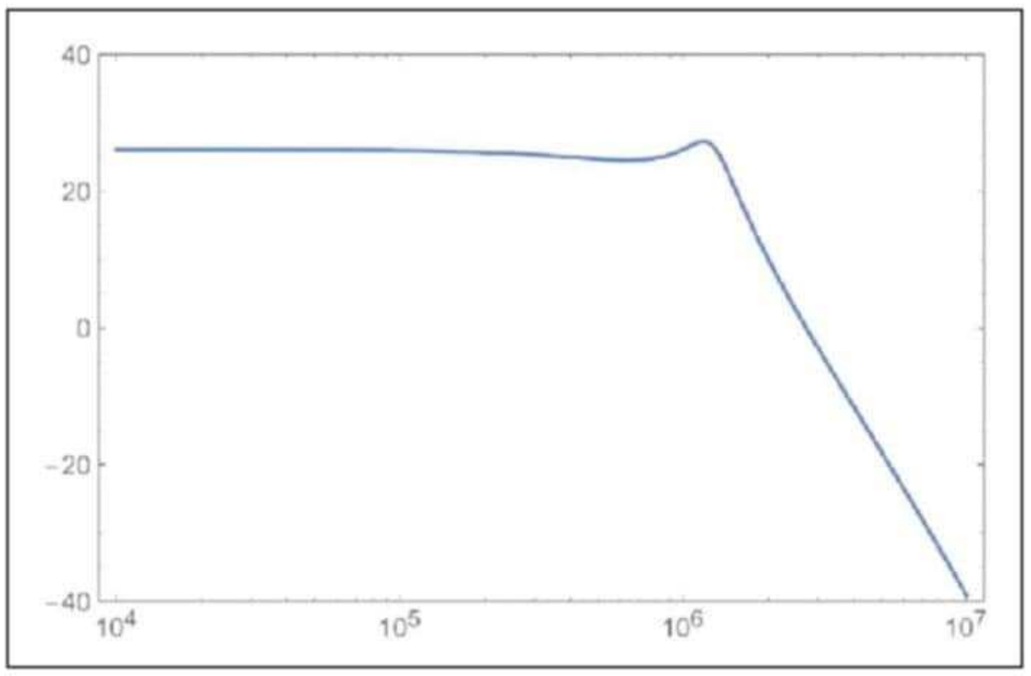
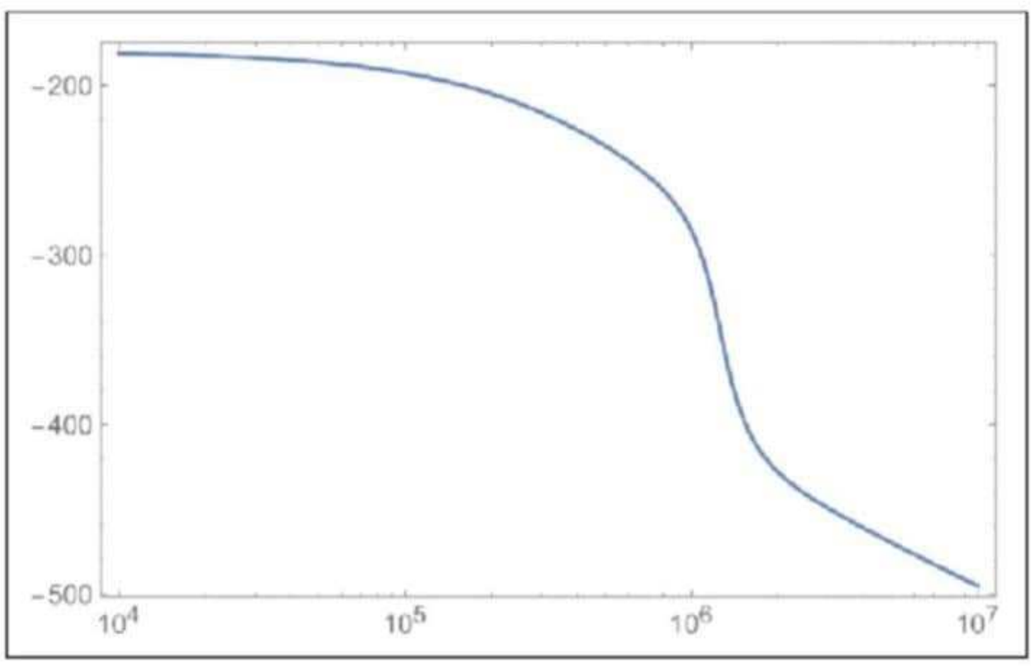


Figure 9: Phase vs. Log Frequency from Equation 5



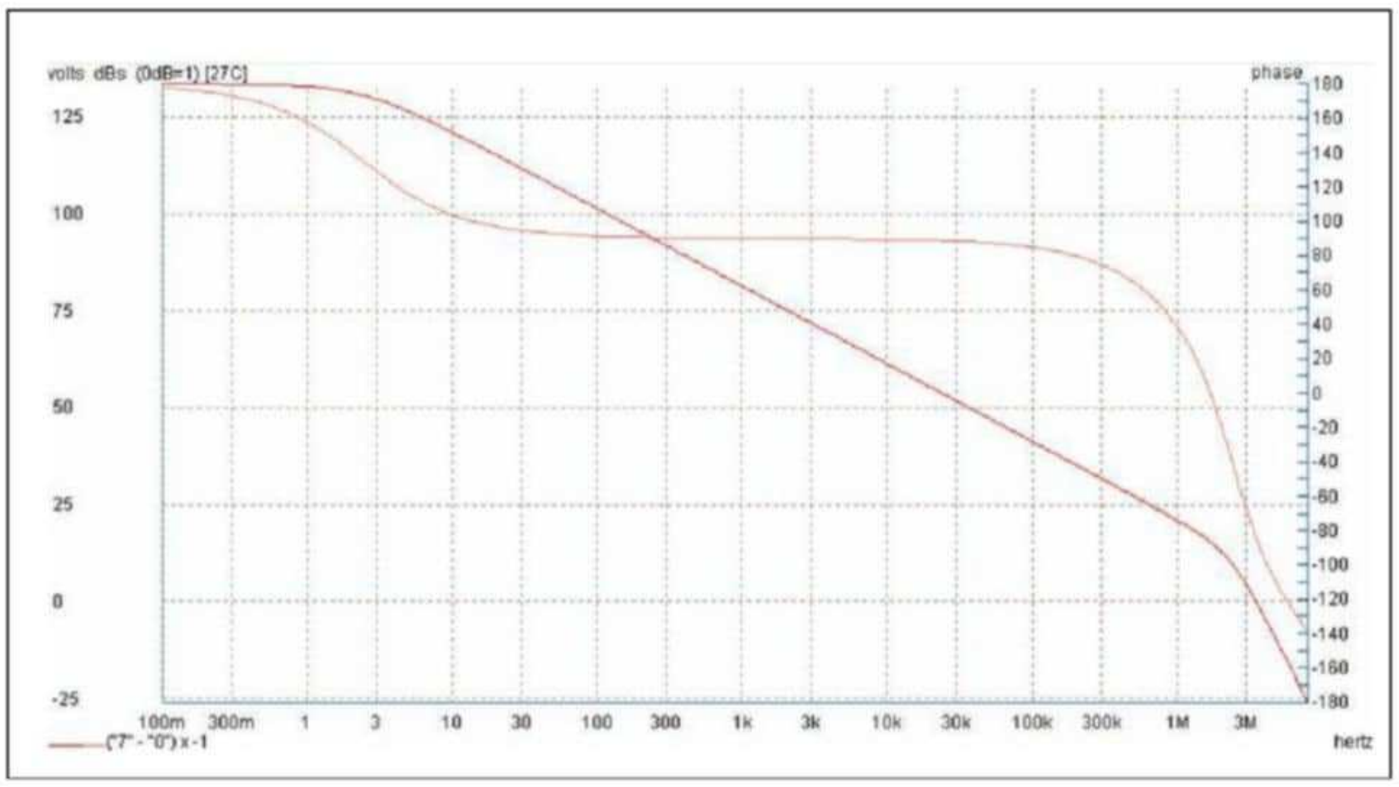


Figure 10: Gain and Phase vs. Frequency f/a after tweaking

About the Author



Paul Marchese served as an electrical technician in the military after high school and then got his BSEE from New Jersey Institute of Technology in 1980. At Westinghouse, he participated in its Advanced Science Program. His entire 37-year engineering career was in analog design, which included precision circuits, switched mode power supplies, low noise preamps, and active and passive analog filtering from sub-Hertz to gigahertz. His father was an electrical technician in the vacuum tube days. Recently retired, Paul infrequently does some contract work, but mostly passes his time trying to understand the physics of loudspeakers, climb and cut trees, doing ups (push, pull, and sit), and juggling. As far as electrical history, Paul is a big fan of Oliver Heaviside and the many contributions of Bell Labs.

Substituting the expressions for the five individual terms: A_1, A_2, B_1, B_2, and B_3, in the transfer function Equation 5, will result in a rather large equation that's best dealt with by using a mathematical manipulator (e.g., MATLAB or Mathematica) or at the very least, a programmable calculator. A more practical, less quantified approach is to use an analog circuit simulator where the outputs can be examined in both the frequency and time domains. Actually, the math model and the analog simulation are both math models, one using straight arithmetic the other using matrixes.

Figure 7 shows the Gain and Phase vs. Frequency of the Analog Simulation. **Figure 8** shows the Gain and **Figure 9** shows the Phase vs. Frequency of the mathematical expression Equation 5. They are almost in exact agreement. Although not a vigorous proof, this verifies that Equation 4 and Equation 5 are correct.

Tweaking the value of C1 and C2 can be done by looking at the frequency response and/or its response to a step (**Figure 10** and **Figure 11**). But don't forget that both the math expression and analog simulations are simplified approximations of a circuit that contains a significant number of semiconductors, with many parameters that are not in the model, or nonlinear, along with layout couplings. So, the final tweaking will have to be done on the actual circuit.

A common test is to look at the output response of a 10kHz square wave at the input. **Figure 12** shows this waveform. On the plus side, the slopes are very steep and the overshoot is very short. On the minus side, there is an overshoot. However, there are other ways to analyze stability, which won't be discussed here.

Measuring Distortion

The distortion levels of the LM3886 (A_2) are low and the levels of the OPA1602 (A_1) are very low. Trying to measure these low distortions can exceed the limits of the test equipment. In the OPA1602 (A_1) datasheet there is a circuit for just such a problem. The circuit (Figure 35 in the datasheet) has low output vs. input gain, but high amplifier gain. This will "gain up" the errors due to the amplifier. It appears to be from Jerald Graeme's article in *EDN* [4]. What's not clear is if the source impedances are considered in the calculations. So, I took the simplest circuit, and added a source resistance of 50Ω, like a signal generator will have. Although there are only three resistors in the circuit, it does require some analysis since both the source input and amplifier output have paths to both the inverting and non-inverting

amplifier inputs. **Figure 13** shows the circuit and **Figure 14** shows what I came up with for a block diagram of the circuit. There are no infinite feedback loop traps in the block diagram shown in Figure 14, so solving for **e/a** is straight forward:

e/a = $(B_1 - B_2)/[(1/A) + B_3 - B_4]$ [7]

where the B_s are the feedback factors, the voltage dividers (using super positioning) from the sources to their respective inputs:

$B_1 = (R_1+R_3)/(R_1+R_3+R_4)$
$B_2 = R_1/(R_1+R_3+R_4)$
$B_3 = (R_3+R_4)/(R_1+R_3+R_4)$
$B_4 = R_4/(R_1+R_3+R_4)$. Note that all denominators are $(R_1+R_3+R_4)$

Substituting the B_s into Equation 7 results in:

e/a = $R_3/[(\{R_1+R_3+R_4\}/A)+ R_3]$

This is just a unity gain version of Equation 7, R_3/R_3, along with the non-ideal effects of the amplifier increased by a factor of: $(R_1+R_3+R_4)$ with respect to R_3. This is the circuit's desired response.

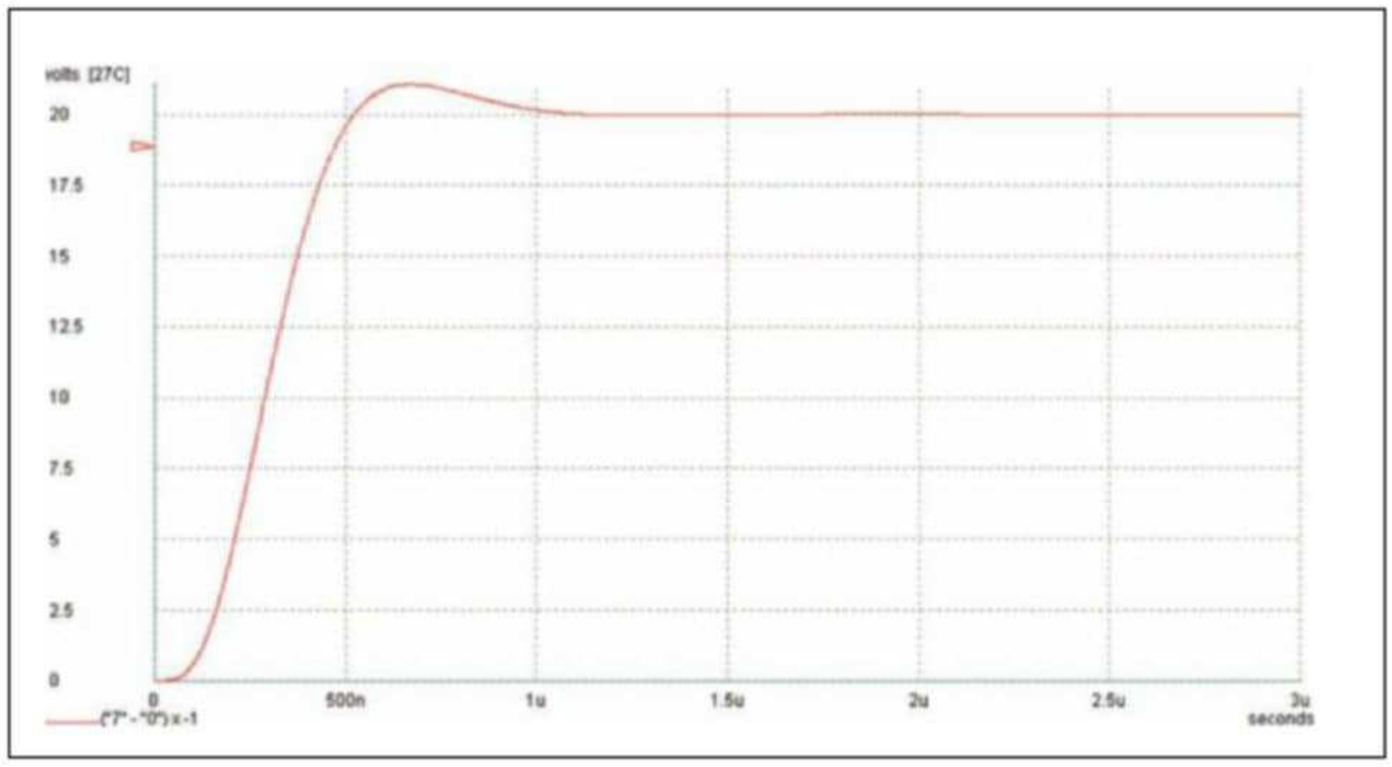


Figure 11: Response to a Step after tweaking

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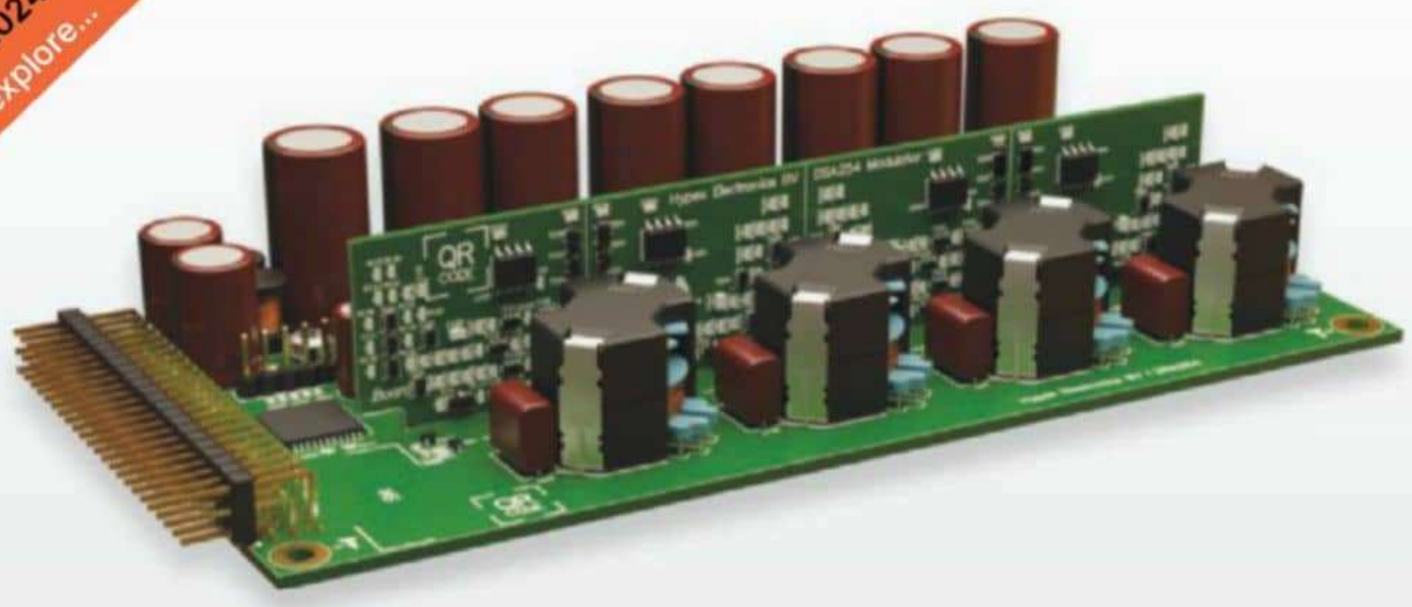
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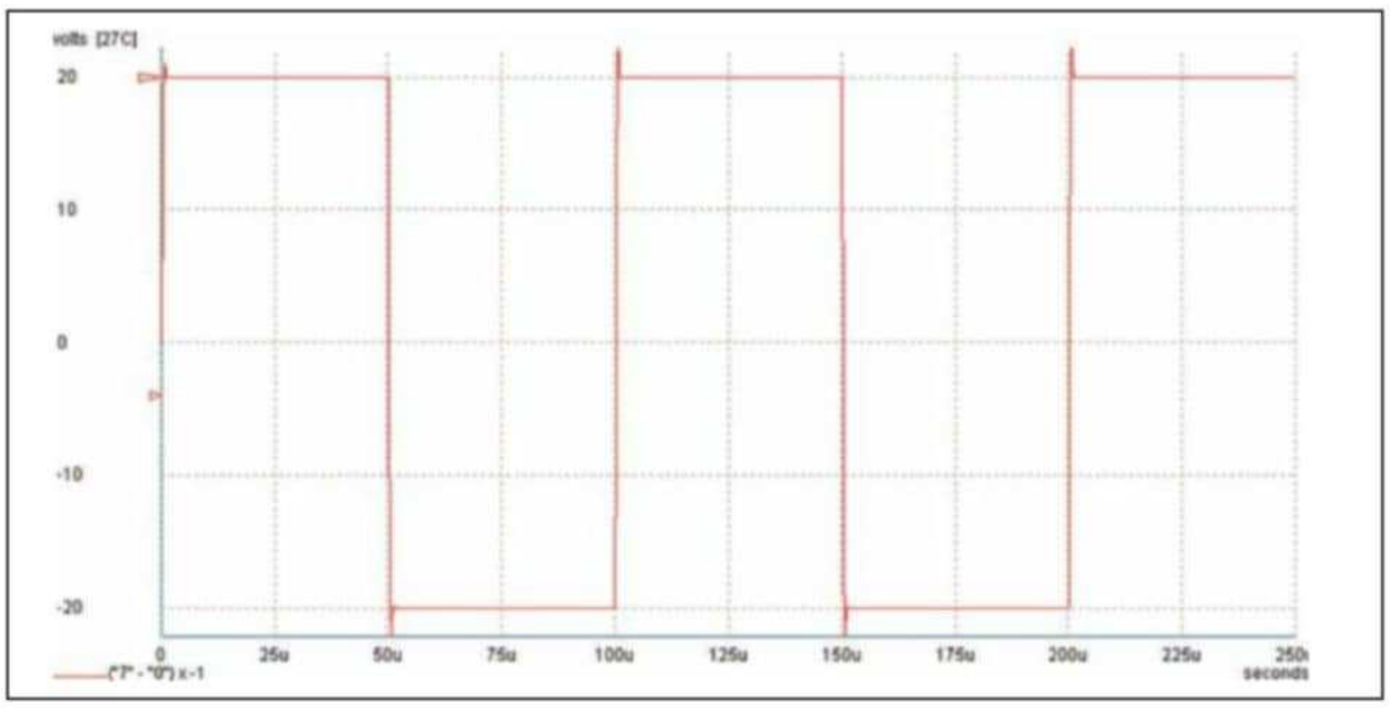


Figure 12: Output of the composite amp with a 10kHz square wave input

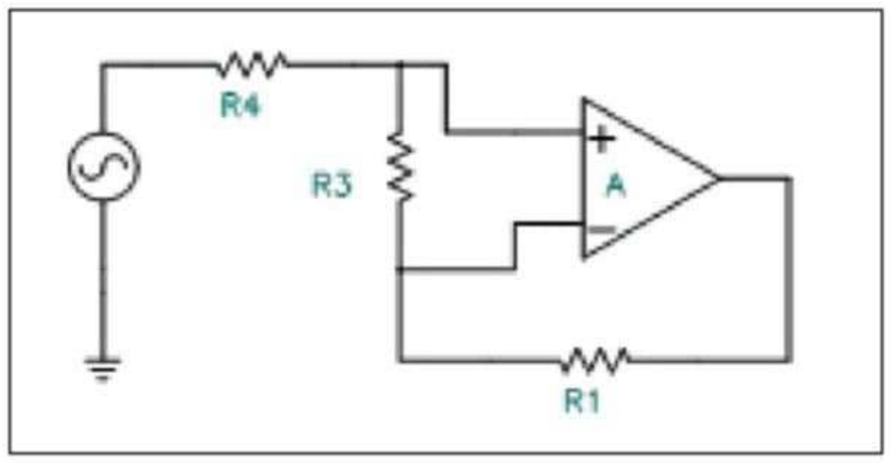


Figure 13: Unity gain, high amplifier gain circuit (when $R_1 >> R_3$, and $R_3 = R_4$)

If R_3 and R_4 are = 50Ω and R_1= 5kΩ, there is no gain, but the effects of the amplifier are gained up by a factor of 102 and the impedance of the source is included. Although this composite amplifier does need gain to be stable, the output to input gain of 1 shouldn't be a problem since the amplifier gain is approximately 40dB.

Since the distortion is gained up, the amplifier under test must have enough loop gain for an accurate measurement. For less than an 1% measurement error, the loop gain must be greater than 17dB (see the Gain Error vs. Loop Gain sidebar). With a "gain" of 102V/V, this will occur at approximately 5.5kHz. So this composite amplifier lacks the bandwidth in the last two octaves to maintain an error less than a 1%. If the "gain" was reduced to ~25V/V, the 1% accuracy can be maintained, at the expense of less output.

The common mode rejection ratio not being infinite is not a source of measurement error, but a source of amplifier error, and will be gained up with the others. Also, since the three-series resistors in the circuit are common to all the calculations, they either self-subtract to zero or self-divided to one, so the absolute values of the resistors are not

Are you interested in learning how guitar amps work, as well as how to fix and service an amp?

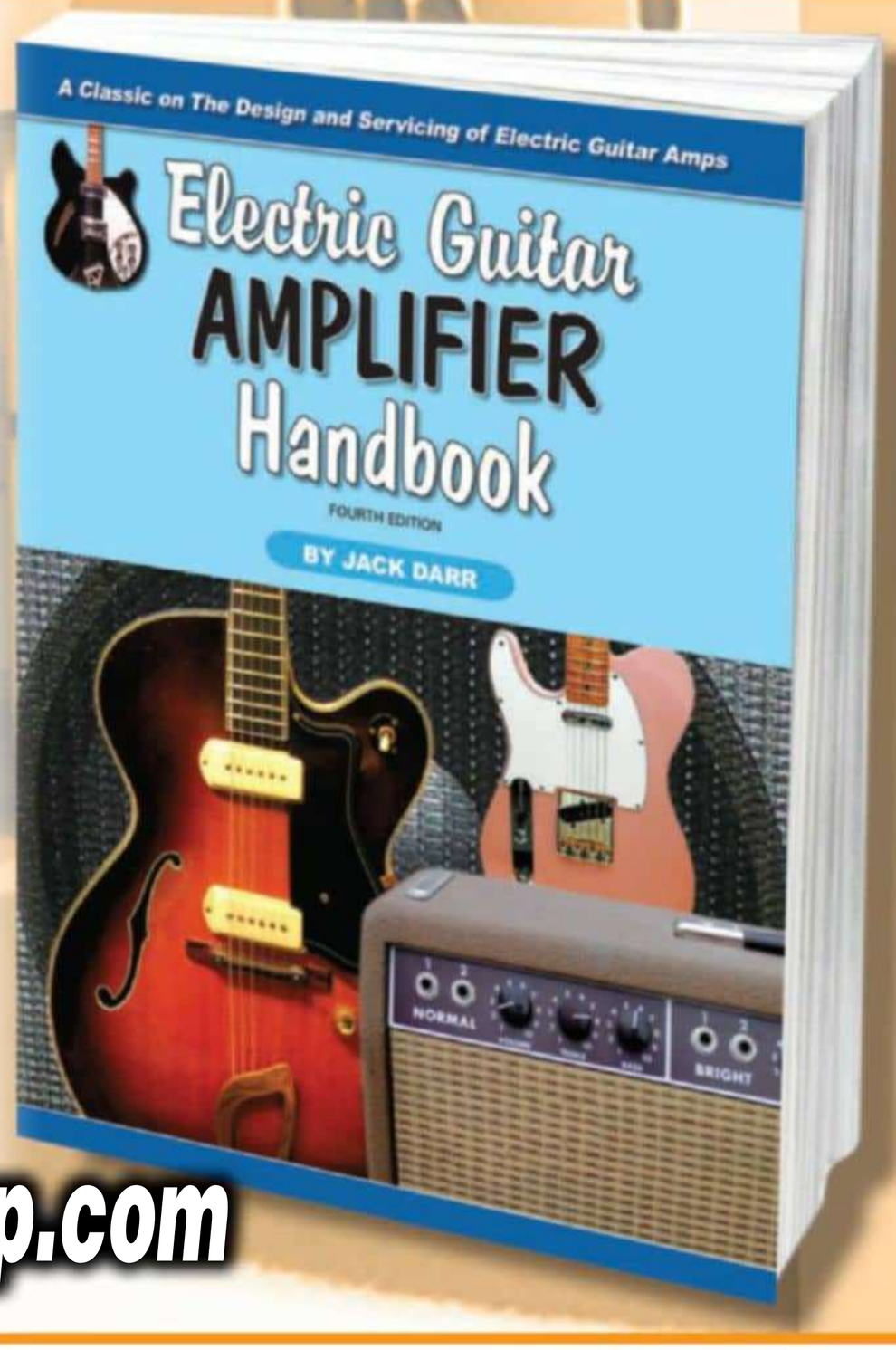
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Jack Darr's *Electric Guitar Amplifier Handbook* details the following:

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- How to properly and safely make amp repairs
- How to troubleshoot tube and transistor amps of all sizes
- Details on "typical" amp circuits

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critical and are only needed to calculate the amount that the amplifier is gained up.

Well, all this theory stuff is great, but an actual circuit is needed. Unfortunately, my old (but state-of-the-art at the time) schematic and layout program was bought by a larger company 17 years ago, and the old program hasn't been updated since. That coupled with operator problems and rust, makes a slow go. And my analog simulator has been gone for 13 years now too. The good news is that there are free simulators offered by Texas Instruments and Analog Devices including PSpice. And micro-Cap, a very powerful analog simulator, is also now a freebie [5], which I'm trying to get up to speed. The LM3886 (A_2) hasn't been available for almost a year now. I'm not sure what's going on with that, but it hasn't been officially discontinued.

This article turned out to be a bigger project than originally anticipated. I realize there's a lot of information presented, and some passed over fairly quickly. I viewed stability as a sub-set of control theory, but I see that it has its own branch of mathematics. This is where the mathematicians have a field day. The rocket landing on the moon is a classic control problem—velocity, gravity, mass, when to start, amount of thrust, fuel use, effects of changing weight, and more, but there are a lot of every day real-world control problems that are important—from an automobile automatic transmission to printing a newspaper, to name two very different examples. With local and global feedback, it has been shown that the composite amplifier becomes a complicated circuit that needs a significant amount of analysis.

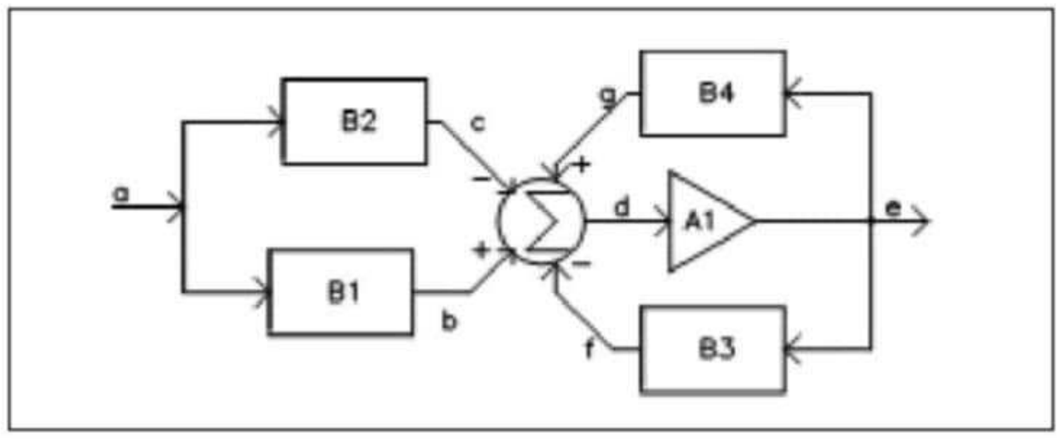


Figure 14: The block diagram of the amplifier and the feedback paths

Gain Error vs. Loop Gain

Let's debunk another rule of thumb, that 40dB (100 V/V) of loop gain is needed for a gain error of 1%. In a first-order approximation of the open loop gain (A_{OL}) of an op amp there is a 90-degree phase shift after the first break frequency, which is low, usually a few dozen Hertz. This means that if B is real (e.g., a resistor divider), then at higher frequencies the loop gain $A_{OL}B$ is at right angles to the real 1 in the denominator and adds Root Sum Squared (RSS). So what RSS value is needed to have a change of 1%? $\sqrt{[1^2+(1/x^2)]} = 1.01$, x = 7.05 =17dB. For example, the frequency limit for 1% gain accuracy of a 1MHz GBWP op-amp with a gain of 1 is not 10kHz as the 40dB of loop gain would dictate, but 142kHz. Non-believers can confirm this on an analog simulator. Note: this is an amplifier with a single pole.



Is Digital Jitter Really a Problem?

Since the dawn of the digital era decades ago, the timing error known as “jitter” in A/D/A converters has been accused of harming audio fidelity. Some people claim that jitter adds audible distortion and noise, affects bass fullness, and even harms stereo imaging width and depth. In this article Ethan Winer explains what jitter is, what causes it, and—most important—lets you hear jitter in controlled amounts so you can decide for yourself how damaging it really is.

By
Ethan Winer

In an earlier article, “Artifact Audibility” [1], I wrote about the audibility of various artifacts including distortion and noise. Several audio examples are provided so people can learn at what levels artifacts can be heard using their own stereo system or headphones. Jitter is another artifact that’s often complained about, but it’s difficult to generate artificially in controlled amounts so there haven’t been many tests.

What Is Jitter?

Jitter is a noise-like artifact caused by slight timing errors in the master clock that controls the transfers between analog and digital formats. Jitter occurs both during audio capture into an analog-to-digital converter (ADC) and output from the digital-to-analog converter (DAC). This article will focus mostly on DAC jitter because consumers and their devices have no control over jitter that’s already been recorded. Jitter is specified in units of time, typically nanoseconds (billionths of one second). This defines how early or late the clock signal might switch. **Figure 1** shows the square wave output of an ideal clock at top, then with some amount of jitter added below. In the first block the clock’s output turns on slightly late, then turns off a little earlier than it was supposed to. This type of jitter is called Random because the timing deviations are more or less random based on thermal noise variations.

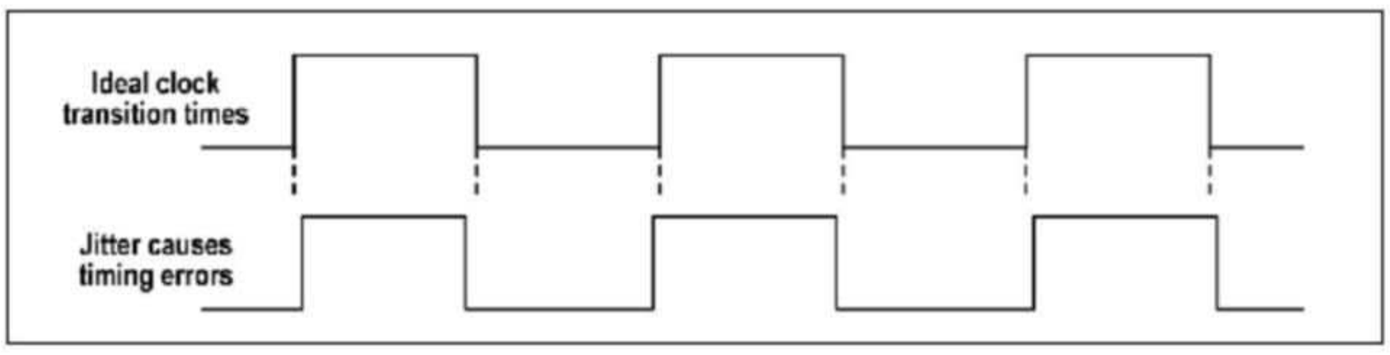


Figure 1: Jitter causes the clock to transition at incorrect times, which creates noise in the audio output.

Even though jitter is a timing issue it manifests as FM sidebands—noise-like artifacts that are added to the music. This is why it’s expressed as a time span rather than an amount of phase shift or frequency deviation. Depending on the nature of the jitter, the sidebands may be harmonically related or unrelated to the music. The spectral content can also vary. Both of these affect how audible the jitter will be when music is playing because of the masking effect. Noise containing frequencies similar to the music source will be less audible than noise at frequencies farther away. Absolute volume or SPL also affects audibility: The louder something is, the more detail we can hear clearly. In that earlier “Artifact Audibility” article I made the point that jitter noise in modern digital devices is typically below the noise floor of a CD, even for inexpensive consumer-grade gear. In my experience that is far too soft to be audible.

For a given amount of jitter, the level of noise added to the audio also varies with the source frequency. Jitter affects higher frequencies more simply because the timing error is a larger percentage of the total period. One microsecond of jitter is 0.1% of the period at 1kHz, but 1.0% at 10kHz. The graph shown in **Figure 2** from *The Art of Digital Audio* by John Watkinson [2] shows the

amount of jitter noise you can expect at different frequencies, with the noise floor of various bit depths as a reference. For comparison, jitter is typically under 0.5 nanoseconds (ns) even with modest consumer devices, so more than 100dB below the music. In various audibility tests people were unable to detect jitter unless it was greater than 30ns.

One possible exception is jitter in the audio stream of an HDMI connection. In February 2009 British magazine *hi-fi news* published jitter measurements for four popular receivers. All but one had substantially more jitter in their HDMI outputs compared to their SPDIF outputs. One was as high as 7.6ns through its HDMI output, compared to only 183 picoseconds (ps) through its SPDIF output. But still, that seems to fall below the level of audibility.

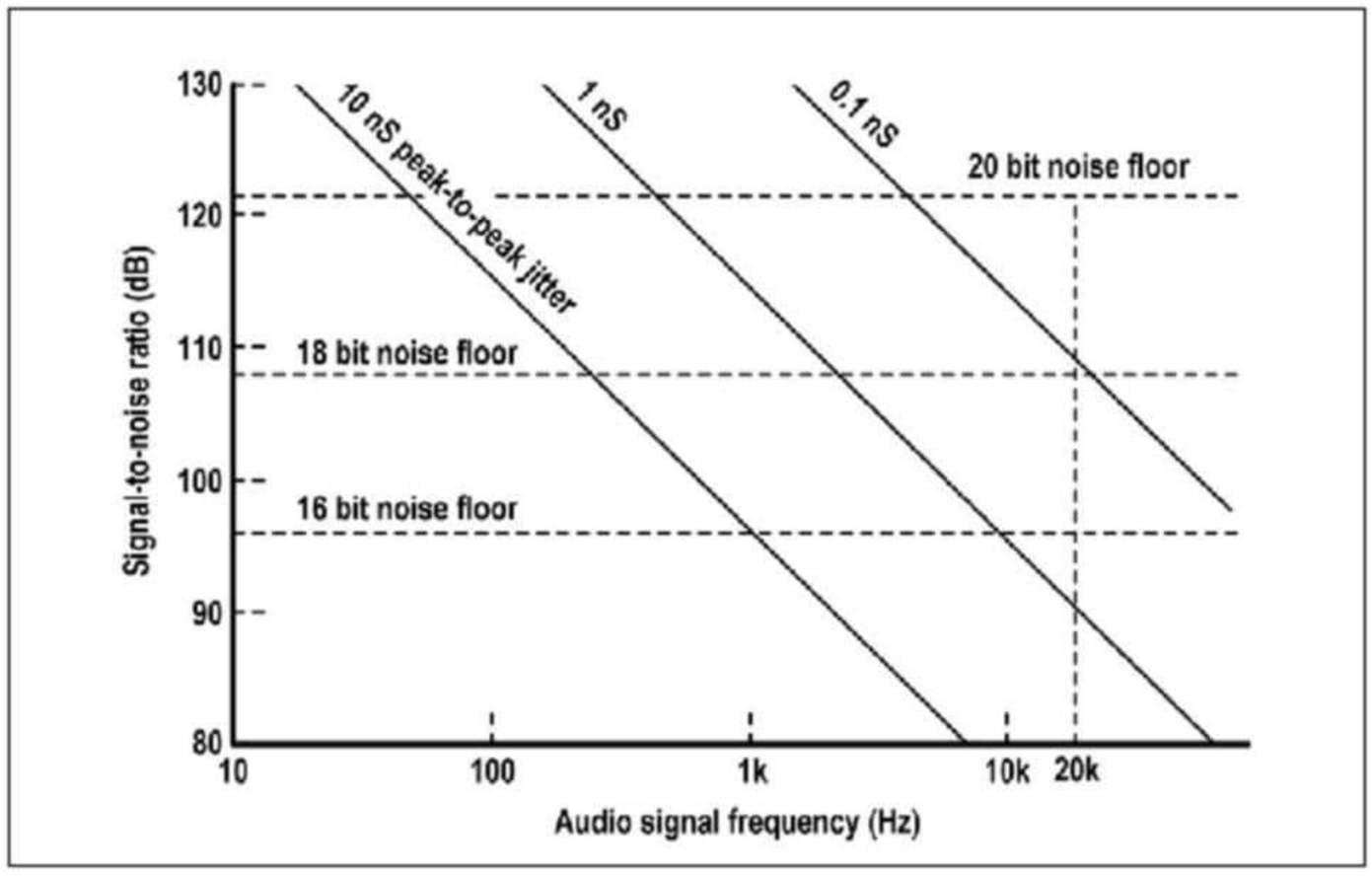


Figure 2: Effects of sample clock jitter on signal-to-noise ratio (SNR) at different frequencies, compared with theoretical noise floors of systems with different resolutions. (Image courtesy of *The Art of Digital Audio* by John Watkinson).

What Causes Jitter?

Every digital audio device that contains a clock starts with an oscillator that generates the clock frequency. The best oscillators use a thin slab of quartz crystal sandwiched between metal plates that's induced to oscillate by applying an electric current to the plates. When the highest accuracy isn't needed ceramic can be used instead of quartz. Both elements have a very narrow bandwidth (high Q), much like a tuning fork, and so vibrate at a single stable frequency. The resonant frequency of a quartz slab varies with ambient temperature, so when the highest accuracy is needed the oscillator circuit board is placed into a tiny oven whose temperature is tightly regulated to just above the ambient temperature. **Figure 3** shows the schematic for one type of crystal oscillator. Here the crystal is placed in the positive feedback loop of an op-amp thus fostering oscillation. At every cycle the op-amp's output reverses direction when its input reaches a specific "transition" voltage.

All electronic components possess some amount of thermal noise, which is a tiny voltage caused by random molecular motion. For example, a 10kΩ resistor at room temperature outputs 0.4μV (microvolts) RMS over a 10kHz bandwidth. Since resistors set the transition voltage, tiny amounts of their thermal noise can change the transition voltage by random amounts. Noise from the power supply can also get into the oscillator circuit. If that noise contains mains components at 60Hz or 50Hz, the resultant jitter can be modulated by that frequency.

Even when jitter is too minimal to affect perceived audio quality, it can still be measured. In June 2010 *Sound On Sound* (SOS) magazine (UK) published an article detailing its tests of seven external master clock products, all of which claimed that their low jitter amounts improve audio quality over the clocks built into typical converters and sound cards (see Resources).

External clocks are useful in complex audio systems that contain many digital devices that need to operate together, such as in radio and TV stations. But when SOS technical editor Hugh Robjohns compared the noise and distortion of several DACs, he found they all had less using their internal clocks than when connected to the external clock products.

From his conclusion: "Overall, it should be clear from these tests that employing an external master clock cannot and will not improve the sound quality of a digital audio system." So, while an outboard clock product is useful when multiple devices must be sync'd to the same word clock, an external clock won't improve audio quality in a typical setup with one sound card or DAC. [3]

Perhaps the most important point about jitter audibility was summed up in *The Audio Critic* where digital audio expert Robert W. Adams of Analog Devices, Inc. wrote: "Traditional THD+N versus frequency tests and FFT spectrum plots for input signals of various frequencies are adequate to cover the effects caused by jitter. There is no reason to single out distortion components caused by jitter as distinct from those caused by such other effects as D/A nonlinearity, op-amp distortion, etc." [4]

So clearly there's no basis for believing that jitter can affect bass fullness or stereo imaging as is commonly claimed. Fullness is a frequency response issue that's easily verified, and imaging

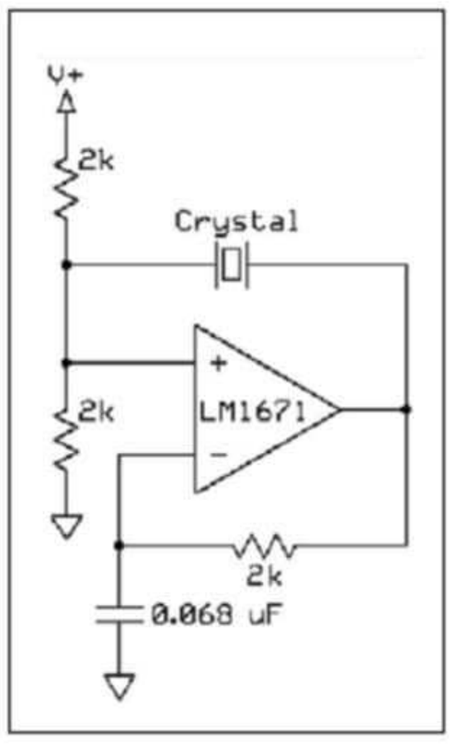


Figure 3: This oscillator puts a quartz crystal in the positive feedback loop of an op-amp.

changes would require much larger timing errors that also differ between the left and right channels.

Audio Examples

Earlier I mentioned that jitter is difficult to generate artificially in controlled amounts. Fortunately, I discovered the Distort program available for free at distortaudio.org. Distort accepts input from wave files, and lets you add various artifacts in controlled amounts such as harmonic distortion, dither, noise, hum and buzz, reduced bit depth (quantization noise), and of course jitter.

To let readers hear for themselves at what level jitter is audible I created four sets of example music files, each with increasing amounts of jitter. These music examples are mostly "gentle, open sounding" productions, rather than loud compressed pop tunes that might be too dense to notice small amounts of added noise or distortion. All the tunes were ripped from their original CDs as full-quality wave files. Note that I applied an aggressive amount of jitter to make it obvious. The smallest amount is 10ns, about 10-20 times more than even consumer-grade audio devices. At 10ns I can't hear a difference from the original source, but maybe younger listeners with better ears can. Then I added 10µs of jitter (1,000 times more) which brought the noise floor up to around -30dBFS, and then 100µs, which raised the noise to -15dBFS. At 100µs you can clearly hear the Tss Tss Tss percussive sound of the jitter artifacts as well as the generally gritty quality.

Before adding jitter, I normalized the files to around -4dBFS to guarantee that the audio will be well below any DAC's clipping level. Rather than require readers to juggle a large series of audio examples I created a five-minute video that plays and identifies each tune four times in a row with increasing amounts of jitter. I also extracted just the jitter from two of the songs, so you can hear what the jitter alone sounds like. YouTube videos apply lossy compression to the audio, so I rendered this video in both QuickTime and Video for Windows formats using CD-quality PCM encoding. Then I hosted them on my own website to retain their full audio quality. Use the links listed under Project Files to either stream the video or save the files to your computer.

Conclusion

Assuming you watched the video, I think the most important revelation is that even 10 times more jitter than is usual for inexpensive audio devices is innocuous and doesn't harm the music. The usual scapegoats "fullness and width" are clearly not affected by the addition of jitter noise. Even with the jitter at 10µs, which is huge and limits the signal-to-noise ratio (SNR) to only 35dB, you can appreciate the power of the masking effect. Unlike tape hiss and vinyl record crackles and pops, jitter noise is present only when the music plays. So, it's always masked by the music and never audible between songs. Then, why do so many people believe that bass response and imaging are affected by jitter?

Through my research in room acoustics, I believe the acoustic phenomena known as comb filtering is the most plausible explanation for many of the differences people claim to hear from cables, power conditioners, isolation devices, low-jitter external clocks, ultra-high sample rates, and so forth. Comb filtering is a type of frequency response error that occurs when direct sound from the loudspeakers combines in the air with reflections off the walls, floor, ceiling, and other nearby objects. This causes a change in the frequency response reaching your ears, even across surprisingly small distances.

My earlier article "Why We Believe," (see Resources), explains more about this phenomenon, and shows how moving your head even a few inches can change what you hear by a very large amount. The difference people hear may in fact be real, but it's not caused by changing wires or adding a "power" product or isolation device.

About the Author



Ethan Winer has been an audio engineer and professional musician for more than 45 years. His Cello Rondo music video has received nearly 2 million views on YouTube and other websites, and his book *The Audio Expert* published by Focal Press, now in its second edition, is available at amazon.com and Ethan's own website ethanwiner.com. Ethan is also a principal at RealTraps, a US manufacturer of high-quality acoustic treatment.

Project Files

Download the demo video from Ethan Winer's website in either format:
Video for Windows (57 MB): http://ethanwiner.com/jitter.avi
QuickTime (64 MB): http://ethanwiner.com/jitter.mov

References

[1] E. Winer, "Artifact Audibility," http://ethanwiner.com/audibility.html

[2] J. Watkinson, The Art of Digital Audio, Routledge, 2000.

[3] H. Robojohns, "Does Your Studio Need a Digital Master Clock?, *Sound On Sound*, June 2010, www.soundonsound.com/sos/jun10/articles/masterclocks.htm

[4] R. W. Adams, "Clock Jitter, D/A Converters, and Sample-Rate Conversion," Analog Devices, Inc., Wilmington, MA.

Resources

Distortion Audibility Tester, https://distortaudio.org

E. Winer, "Why We Believe," http://ethanwiner.com/believe.html

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How Can I Learn?

Learning about hollow-state electronics used to be as easy as DIY trial-and-error, getting information from a family member or friend, or reading technical books and magazines such as *Popular Electronics*. Even today with all the information available online, finding a knowledgeable and patient mentor and lots of practice still seem to be key factors.



Photo 1: Back in the 1950s and 1960s most drugstores had tube tester machines that enabled you to test your own tubes, buy the necessary new ones, and replace them.

By
Richard Honeycutt

When I was an electronics professor, one of my students asked me, “How can I learn about vacuum tubes and circuits”? This question led me to ponder how I had learned about hollow-state electronics. My dad grew up in the 1920s and early 30s in a farming community. Readers who know anything about farmers know that their motto is, “if it’s broke, fix it!” So, when Dad’s Model T Ford broke down as he was on the way home from college for a holiday, and the engine need rebuilding, he “thumbed a ride” to the nearest parts store, bought the needed replacement parts, and rebuilt the engine beside U.S. highway 64. In the early 1950s, when our radio/record player went dead, he removed the back panel, pulled out the tubes and checked them on a drug-store tube tester (**Photo 1**), bought the necessary new ones, and replaced them.

By observing Dad, I learned how to figure out how to go about repairing various things, although I picked up almost no theoretical knowledge about either cars or radios.

Once in the mid-1950s, our radio developed a problem that did not involve tube replacement. Dad had been a high school math teacher, and one of his students was the son of a professional radio repairman. So, we took the radio to his shop, and he let us watch as he repaired it. (**Photo 2** shows an example of what such a repair shop would have looked like back then.) The problem turned out to be a defective capacitor, and the repairman kindly told me what a capacitor was and what it does in a circuit.

By this time, Dad had answered enough of my third-grader questions that I had some idea how electricity worked, how AC and DC differed, and what radio waves were. But I was in no way an electronics hobbyist yet. That status awaited my buying a *Popular Electronics* magazine at a bus station in the early 1960s to learn about the transistorized microphone preamp from an article referenced on the cover. Reading the magazine cover-to-cover, I began to learn much more about electronics from both construction articles and tutorial articles, although with transistors costing hundreds of dollars each at the time, I could not afford to build the amplifier described in the article.

Tube equipment had several advantages. Parts cost much less than transistors at that time. I found a way to build up a “junk box” of parts scrounged from discarded radios and TV’s given to me by

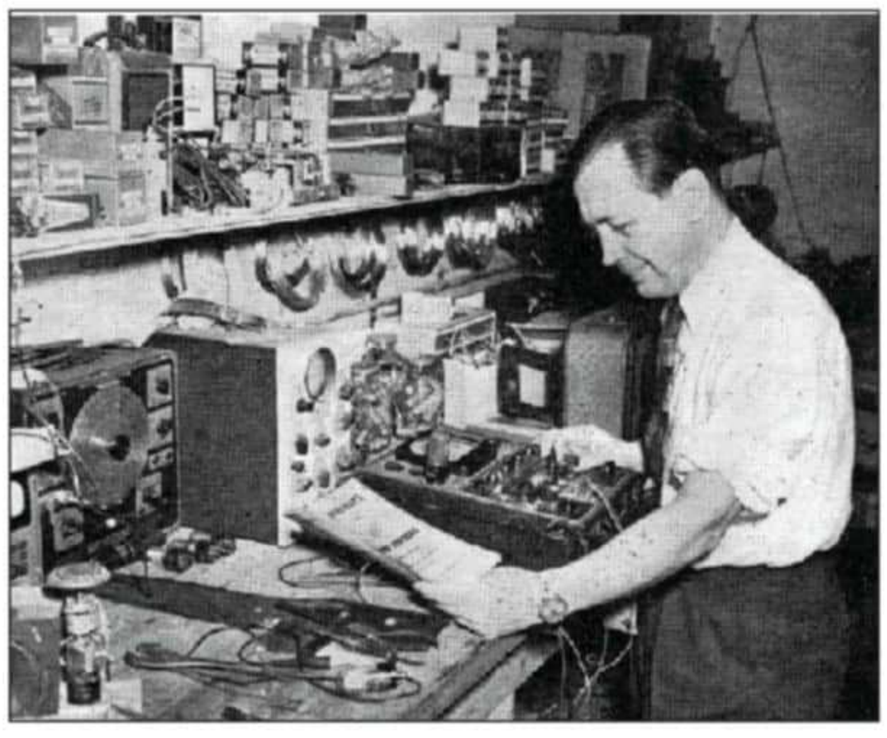


Photo 2: Here is an example of what such a repair shop would have looked like back then

relatives, and some projects could be built from these parts (**Photo 3**). And there were several good electronics magazines publishing articles about tube equipment design and operating principles.



Photo 3: I found a way to build up a "junk box" of parts scrounged from discarded radios and TVs given to me by relatives, and some projects could be built from these parts.

The Magazines

Popular Electronics was the first electronics magazine to which I subscribed. It included a broad coverage of electronics news, high fidelity (our name for audio back in the 1960s), amateur ("ham") radio, and even an occasional adventure story about two fictional teenaged electronics enthusiasts named Carl and Jerry. The contents were targeted toward hobbyists and service persons. Even the fictional stories were educational. Issues from 1954 through 2003 can be read online at www.worldradiohistory.com/Popular-Electronics-Guide.htm.

I also subscribed to other electronics magazines, including *Electronics Illustrated, Radio Electronics, Electronics World*, *Radio TV Experimenter*, and several others. Each of these magazines included informative and educational content.

Audio magazine, published from 1947-2000, focused mainly on hi-fi and stereo equipment, but occasionally included more technical articles, such as Richard Heyser's articles on audio equipment testing. Even the audio equipment articles—not to mention the annual review of audio equipment tests—were informative by nature of their giving readers an overview of the state of the art in audio reproduction equipment at any given time.

In 1970, Edward T. Dell, an ordained Episcopalian minister and avid electronics and audio hobbyist, began publishing The Audio Amateur or TAA (**Photo 4**). Later to be joined by *Audio Electronics*, *Speaker Builder,* and *Glass Audio*—the latter focusing on vacuum tubes— and *Multimedia Manufacturer*, TAA remained in publication until 1992, after which the four magazines were combined into *audioXpress,* which, along with *Voice Coil*, the trade journal for loudspeaker manufacturers, is now published by KCK Media Corp. At the risk of tooting my own horn, I naturally recommend *audioXpress* as an excellent source from which to learn about hollow-state electronics.

Wireless World (www.worldradiohistory.com/Wireless_World_Magazine.htm) published in the UK under various names from 1911-2008, covered radio, audio electronics, loudspeakers, and related topics at essentially an engineering level (www.worldradiohistory.com/Wireless_World_Magazine.htm).

Whether by reading back issues online or purchasing them at hamfests or from online sellers, enthusiasts can learn much about hollow-state electronics through the enjoyable time travel provided by these magazines.

Books

Having noticed my interest in electronics, Dad bought me a set of electronics training manuals written by Van Valkenburg, Nooger, and Neville, entitled Basic Electronics (**Photo 5**). This extremely well-written and -illustrated set of books provides a non-mathematical introduction to electronics for service technicians. It is now available online (see Resources). Though another volume—this one on solid-state devices and circuits—was added later, the original five-volume set focused on hollow-state electronics.

Dad also gave me a copy of the *Supreme Publications Radio Servicing Course Book*. This one was written for self-training of those who wanted to repair consumer radios and other similar equipment. This book is available from several online used-book sellers.



Photo 4: The first issue of *The Audio Amateur*, the predecessor to *audioXpress*, was published in 1970.

Hollow-State Electronics

During my sophomore year in college, my electronics professor gave me a copy of Paul M. Chirlian's excellent text, *Analysis and Design of Electronic Circuits*. Written for junior or senior level college students majoring in electronics engineering, this book goes into significant depth concerning the physics and analysis of both vacuum tube and transistor circuits.

Since graduation from college, I have been able to obtain several other highly respected books that include well-informed discussion of hollow-state devices and circuits.

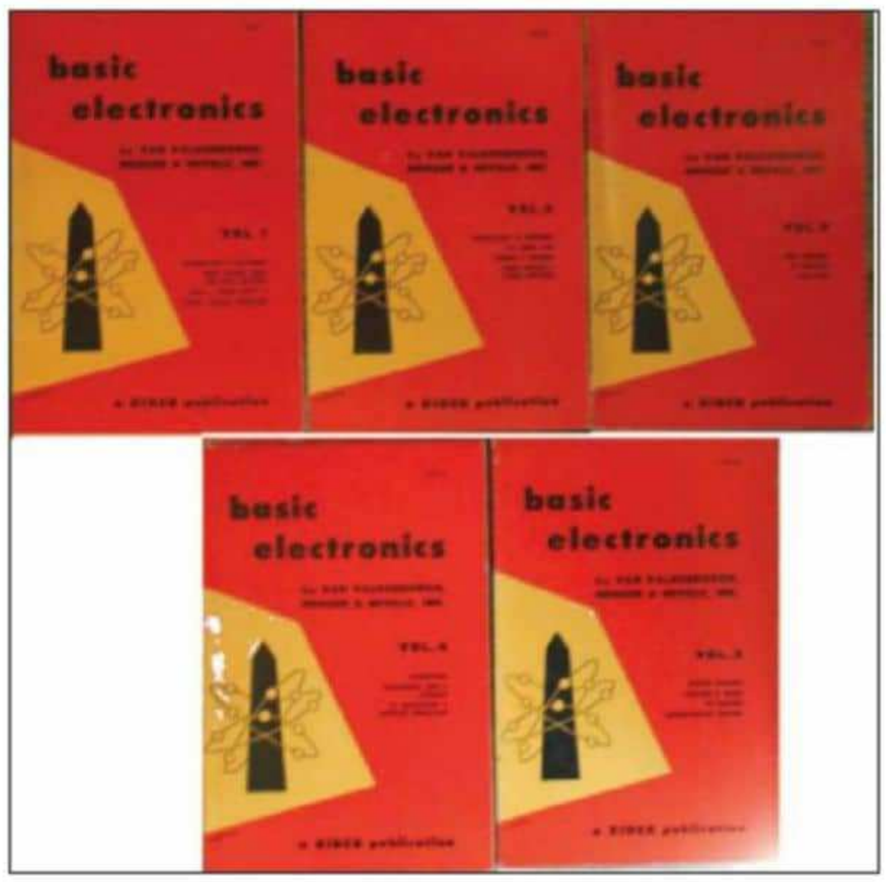


Photo 5: The Basic Electronic book series (SAMS Publications) provides an excellent introduction to hollow-state electronics.



Photo 6: Although modern electronic devices may not be designed to facilitate repairs, older models from friends, relatives, yard sales, and hamfests can provide valuable opportunities to gain experience.

The RCA Radiotron Designers' Handbook, a 1482-page 1952-vintage tome covering literally everything from tube design and manufacture through tube performance and circuit design. It even includes sections on phonograph reproduction, electrical network theory, radio-frequency design considerations, and design notes for iron-cored inductors and transformers.

The RCA Receiving Tube Manual, published for decades in succeeding new editions, contains not only technical specifications of RCA tubes, but also a tube application guide, sample professionally designed tube circuits for audio and ham equipment, and very abbreviated summary of the technical information contained in the *Radiotroon Designers' Handbook*. GE, Tung Sol, and other manufacturers also published tube manuals, but for breadth of content, the RCA manual has always been my favorite.

Although not primarily about vacuum tubes, Howard Tremaine's 1757-page (second edition) *Audio Cyclopedia* discusses many principles of audio electronics, including some topics impinging on tubes.

Jack Darr's electric *Guitar Amplifier Handbook* is written for amateur and professional guitar amp repair persons and contains a guide to amplifier repair methods for both solid- and hollow-state amps, as well as the schematics of many electrical instrument amps that have been on the market beginning in the 1940s.

Mullard's *Tube Circuts for Audio Amplifiers* is a collection of DIY designs for hollow-state audio equipment, dating from the 1950s.

Rudolf Moers' first edition of *Fundamental Amplifier Techniques with Electron Tubes* was published in 2010 and comprised 822 pages in an 8″x11″ format. Don't let the word fundamental in the title mislead you. The principles and techniques Moers discusses are not fundamental in the sense of being non-rigorous, but rather should be understood as covering the fundamentals upon which hollow-state devices and circuits operate. In my review of Moers' first edition in the April 2012 issue of *audioXpress*, I stated that the author "presents a 21st-century perspective on the science of hollow state design as applied to amplifiers and power supplies. Beginning with the principles of electron emission, the book progresses through standard vacuum tube varieties: diodes, triodes, tetrodes, and pentodes, after which it covers such general principles as frequency dependent behavior, nonlinear distortion, noise, and negative feedback. The book concludes with a chapter on the construction of electron tube amplifiers." In his

About the Author



Dr. Richard Honeycutt bought a *Popular Electronics* magazine from the rack at a bus station in 1960. It contained an article on building a transistorized audio amplifier, which captured his interest. He followed up by subscribing to several electronics magazines. To avoid the cost of parts, he built up a "junk box" by disassembling trade-in TVs from his uncle's furniture store. His dad bought him the Van Valkenburg, Neville, and Nooger Basic Electronics book set, so he learned about tube circuits as well as transistors. He started repairing electronic devices about 1965, earning his First-Class Commercial FCC license in 1969. He has worked part-time in broadcast radio engineering and audio electronics repair since 1968, in addition to full-time work in acoustical and audio system design, plus a 20-year stint teaching electronics at the college level.

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Hollow-State Electronics



Photo 7: Experienced ham radio operators are often glad to help you learn about hollow-state electronics.

second edition (908 pages in an 8½"×12" format), the same summary is true. Moers describes the aim of his second edition as being "to give you useful knowledge about electron tube technology in the application of audio amplifiers, including their power supplies, for the design and DIY construction of these electron tube amplifiers yourself."

Experience

An important part of learning hollow-state electronics is experience. It has been said, "You can't learn to drive in a parked car." I was fortunate in that my uncle, who owned a furniture store, gave me old TV sets on which I could experiment and from which I could scrounge parts for my junk box. I have also found equipment needing repair in yard sales and at hamfests. After I had gained some knowledge and experience, friends and relatives brought me radios, record players, and TVs for repair (**Photo 6**). After I started playing electric guitar, I began repairing my own amplifiers. During high school, I repaired and rewired the ancient sound system in the school auditorium.

I mentioned Ed Dell earlier (founder of the Audio Amateur and other magazines. Dell believed strongly that involvement in a hobby was very important to being fully educated and living a full life. During his 90 years, he strove to share the blessing of his audio electronics hobby with as many people as possible. In my years as an electronics professor, I noticed that my best students (and not coincidentally, the ones who landed the best jobs after graduation) were those who had practiced electronics as a hobby before entering our electronics technology program. It was no surprise when a group of these "best students" asked me to teach a one-semester course in vacuum tube electronics. I persuaded my department chair to approve the course on the basis that a nearby avionics repair facility who employed some of our best graduates still used (in the 1980s) exclusively hollow-state test equipment in their calibration lab. I used the Van Valkenburg, Nooger, and Neville books and the *RCA Receiving Tube Manual* as our textbooks. All these students found excellent employment after graduation, working at companies such as Analog Devices and Piedmont Aviation. Two of these graduates now design and build boutique guitar amplifiers: one, professionally; and the other, as a hobby.

The ham radio community has long been known for their eagerness to share their hobby with neophytes, and even coined a term for elder hams who voluntarily helped less experienced folk to learn about ham radio. These elders were (and still are) called "Elmers." The name seems to have arisen in a 1971 article in *QST* magazine. Even though I did not earn my ham license until the 1990s, I did have several respected Elmers helping me on my electronics journey. The first was the son of my father's radio-serviceman friend, who answered lots of questions while I was trying to repair my first (cheap!) guitar amplifier. The second was Charlie, a brilliant engineer afflicted with spastic paralysis who helped me attain my Boy Scout Radio merit badge, and later hired me to act as his hands in the TV repair shop while I was in high school and college. The third was Lewis (aka "Tink"), the chief engineer of WFDD-FM, where I served for three years as student engineer during college. All three Elmers taught me by words and example—things that could not be learned from books (**Photo 7**).

I hope this account of my experience will prove helpful to those who are now learning hollow-state or other specialties in electronics.

Resources

Electric Guitar Amplifier Handbook, CC-Webshop, https://cc-webshop.com/collections/books-audio/products/electric-guitar-amplifier-handbook

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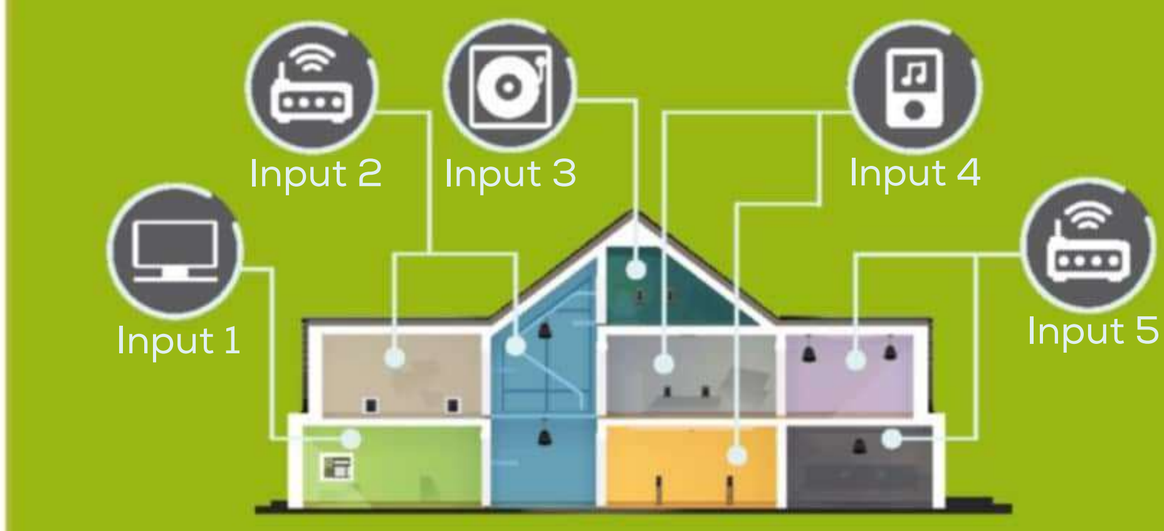
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