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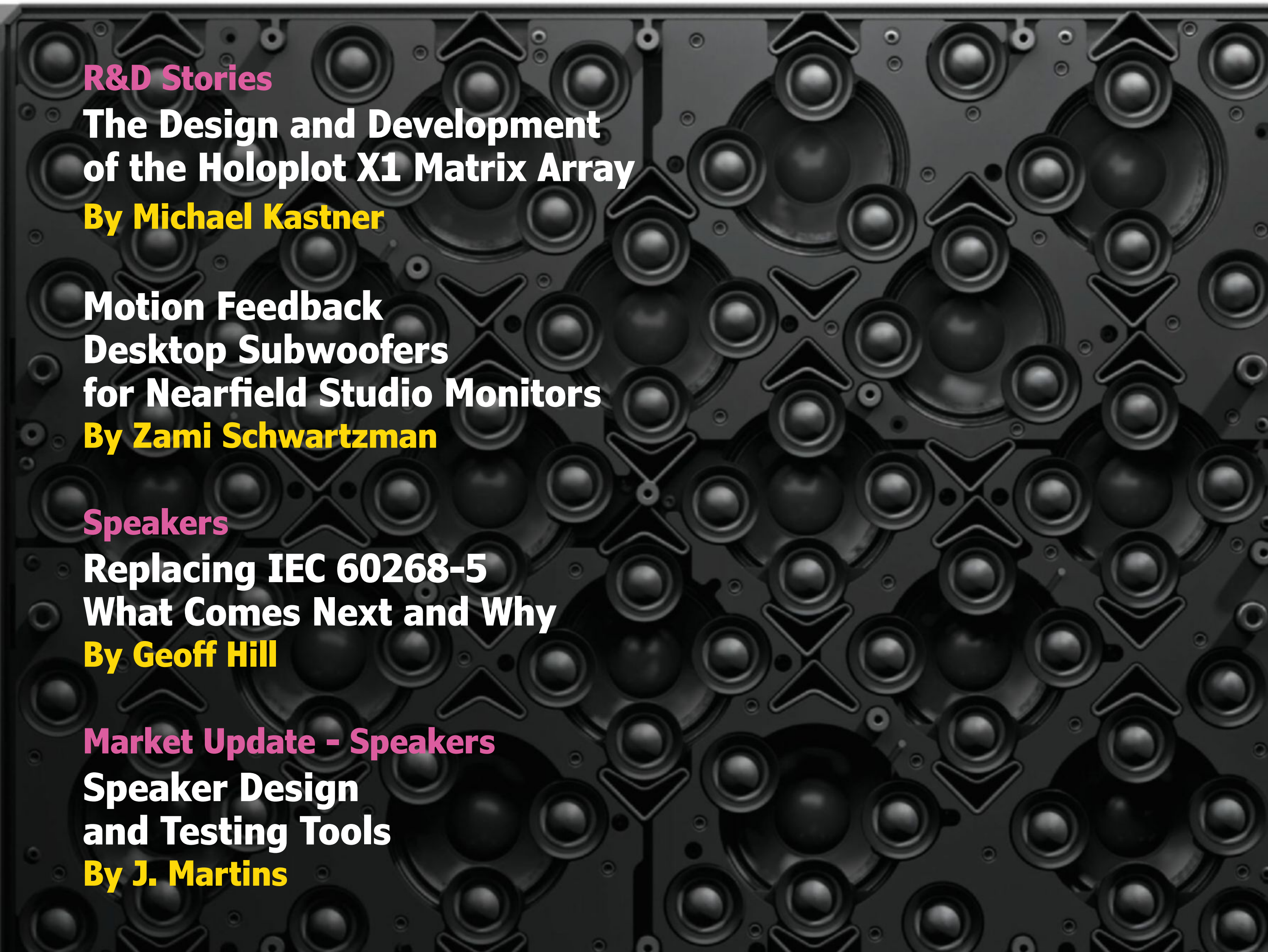

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The Better Mousetrap

A "better mousetrap" describes something that improves upon an existing object or concept. Typically used in the phrase "build a better mousetrap," whatever the intention was in the original expression, in the technology world it is frequently used with a sense of depreciation for a useless effort meant to improve upon something that ultimately awards the same exact result. I've heard it (too) many times in the audio industry as a way to detract people from investing time and money in ideas that could be excellent but that ultimately will not award the exponential noticeable improvements required to justify the investment. Unfortunately that seems to be a sort of rule in the speaker industry, where ultimately it is the end result that matters but sometimes it's just very hard to get there.

I've seen "inventors" being told that no manufacturer would be willing to invest in the "productization" or the new tooling and readjustments in production methods needed. And even if the manufacturing challenges seemed surmountable, many times investors or companies say no to supporting the "considerable marketing effort…"

More common to the majority of incremental advancements that can be realized in speakers is the realization that although a concept sounds promising and even demonstrates noticeable improvements, it would never be seen as "new," it would not result in significantly perceived improvements, nor would it award any cost savings, unless produced on a massive scale. Since new products in the speaker category are seldom perceived as "new," we are frequently restricted to satisfying the two other criteria of perceived benefits and cost savings to be "investable" in today's market environment. And that is the ruthless "Iron law" constrain of Innovation in Product Development. And yet, everyday we see examples of success in the speaker market—they just need to be evaluated using a different perspective or scale of success.

The revolutionary Dragonfire Acoustics planar magnetic desktop speakers were launched without strategic concerns of who would buy them and why. Translating decades of research, the Dragonfire speakers are absolute perfection—and yet remain largely ignored. But their creator, Dragoslav Colich is the successful co-founder and chief technology officer of Audeze, where his work in planar drivers is widely recognized and extremely successful.

I met Zoltán Bay in 2016, when he demonstrated his unique BRS tweeter and he was firmly convinced that large consumer brands would see it as innovative for desktop multimedia applications and would license the technology, reaching mass scale for mainstream audio applications. That didn't work out, but he did build a thriving high-end business with its exclusive Bayz Audio speakers for a select clientele who value extreme quality and exclusiveness.

That's why when *audioXpress* writes about speaker innovation, we value and recognize as much the journey and the strategy, as we value the technology, the patents, and the "success." In this issue of *audioXpress*, we have another one of those innovation stories. One that started like many others, with a bold idea that could easily have suffered the same luck as other "better mousetraps." Holoplot was born from the original research of Helmut Oellers and his "holophony" vision for true spatial audio reproduction. Its research and approach to Wave Field Synthesis, implemented in his earlier work since 2011, was successfully demonstrated and paved the way to a company that is now determined to disrupt the professional audio industry, by creating "an entirely new category of sound system."

I was privileged to attend an earlier demonstration, at Prolight+Sound 2016, in Frankfurt, and that is why I would like to share that moment with one of several photos I have in the *audioXpress* archive, where Helmut Oellers shows the first iteration of the Holoplot system.

J. Martins, Editor-in-Chief

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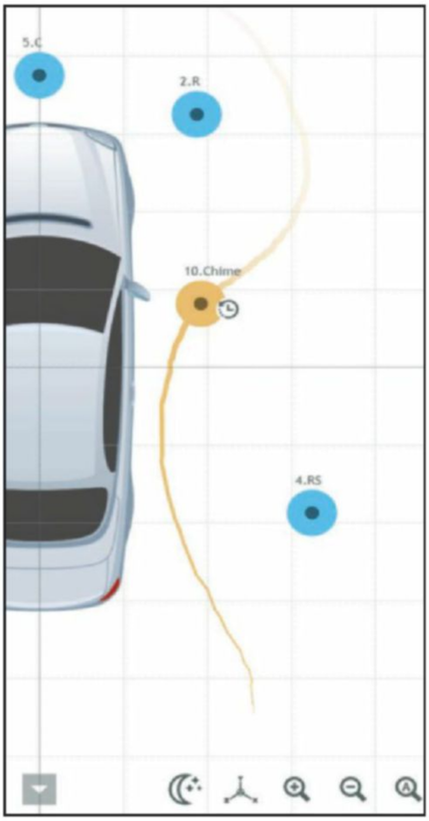

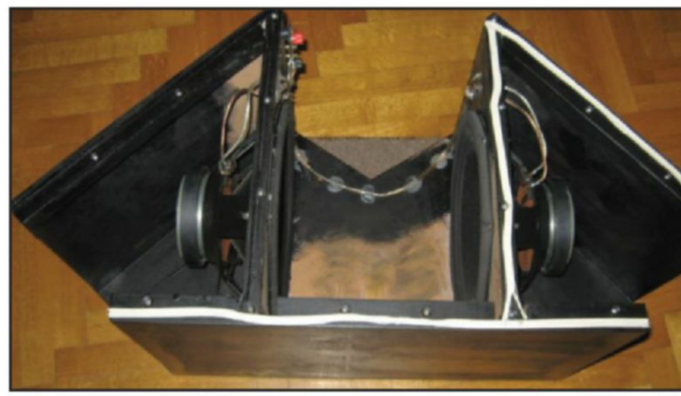

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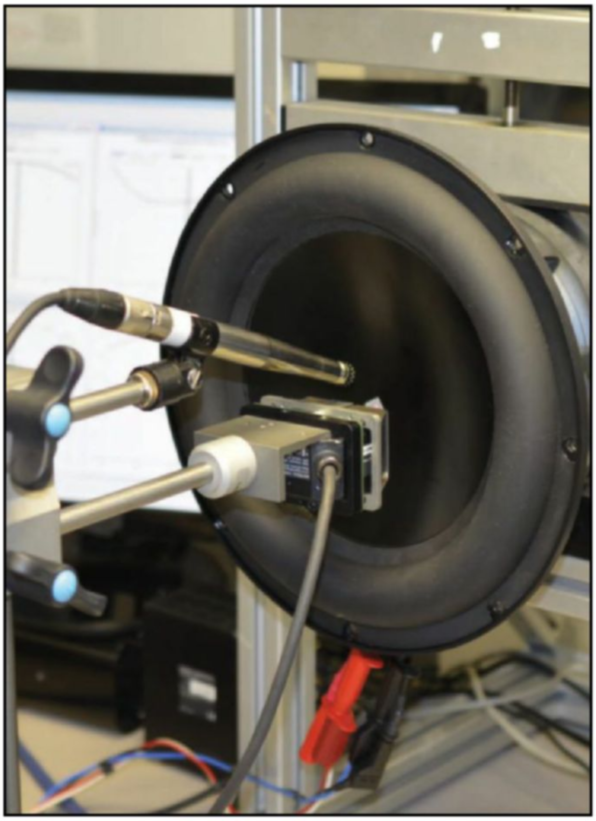

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Market Update—Speakers

Speaker Design and Testing Tools

By
J. Martins
(Editor-in-Chief)

In this article we revisit some of the tools that speaker designers, developers, and DIY builders have at their disposal, from free software to the latest hardware, from a home bench and R&D to production line testing.

Image courtesy of Klippel-Warkwyn

Revisiting a list of free or affordable speaker-related software design and measurement tools unfortunately feels like a trip back to the 1990s, the world of Windows 3.1 and 8-bit graphics. And that's one of the reasons why, although there's a lot of great tools out there—some of which remain in use by many designers, and even consultants, R&D centers, and manufacturing facilities—we prefer not to promote them any longer.

It has nothing to do with how good or unique those tools are. Most of that software was created by a single person, some living in exotic regions of the world, and it's hard to tell if they are alive because they existed prior to the creation of social media, so there aren't any personal pages to follow, and the only reason why that software remains valid is because an enthusiastic community in forums continues to offer support.

As Vance Dickason constantly reminds us, it's not easy to replace a tool we learn to use, we trust, and rely upon for years of work, when suddenly the sole creator and owner of the business disappears. That was the unfortunate situation with Chris Strahm (1956–2016) and all his work with LinearX in measurement tools that remain unique to this day. But software (and hardware) doesn't need to be done like that.

As another example, the extremely popular MLSSA Acoustical Measurement System from DRA Labs, founded by Doug Rife in 1986, was last updated in 2003 to support Windows XP(!). Why there are people still using it, is hard to understand.

I also revisited some of the lists shared in DIYaudio.com, and I could still see some MS-DOS references—which obviously don't even have websites to visit since the Internet hadn't been created then... It's great to recognize those things and their historical value, but no speaker enthusiast should waste their time trying to learn software that is long obsolete (and replaced).

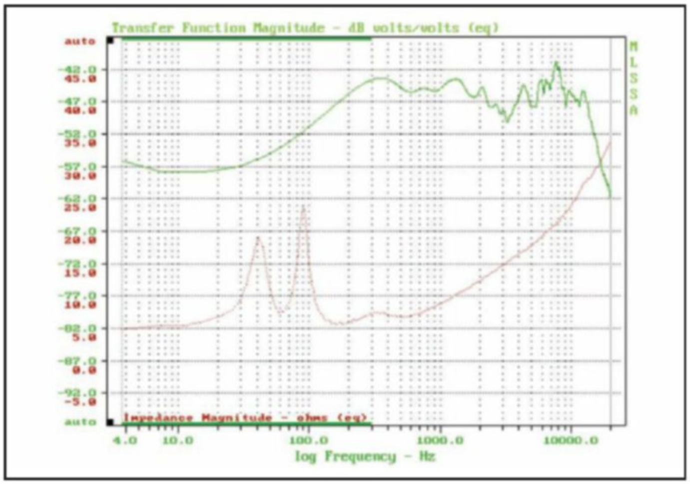
Hi, I'm a PC

A possible exception to this sad state of things is the popular ARTA (Audio Real Time Analysis) software, for audio measurements and analysis, which runs in the most recent version of Windows 11 and continues to be updated regularly by its creator Ivo Mateljan, from Croatia. Mateljan is now retired but still sells personal and commercial licenses of the software. Over the years, a significant number of users have devoted time and patience to create tutorials and even manuals in other languages. The interface is not something to write about, but for those who spend the time learning to use it, it's a valuable tool.

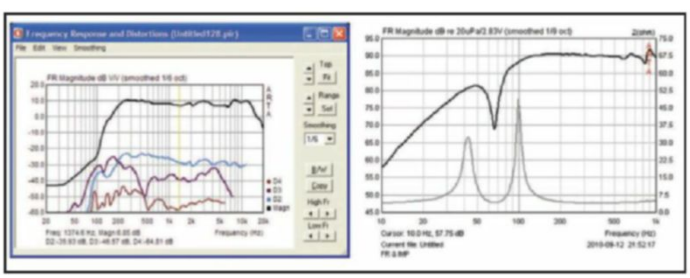
Likewise, TrueRTA, a remarkable audio spectrum analyzer software and WinSpeakerz, a popular speaker simulator, are two other recommended tools, available at www.trueaudio.com and created by John L. Murphy, who is also the author of the book *Introduction to Loudspeaker Design*. Unfortunately, even though keeping an old Windows PC just for software such as TrueRTA is perfectly fine, it's not very reassuring to depend upon tools that remain in the 32-bit realm and have not been updated for the last 8 years (14 years in the case of WinSpeakerz). And the reason seems to have to do with TrueAudio's involvement in the development of the hardware and software for the DATS (Dayton Audio Test System) speaker test and measurement solution from Dayton Audio, sold by Parts Express.

Until we start to see a new generation of developers willing to devote the time to create software for the popular Raspberry Pi,

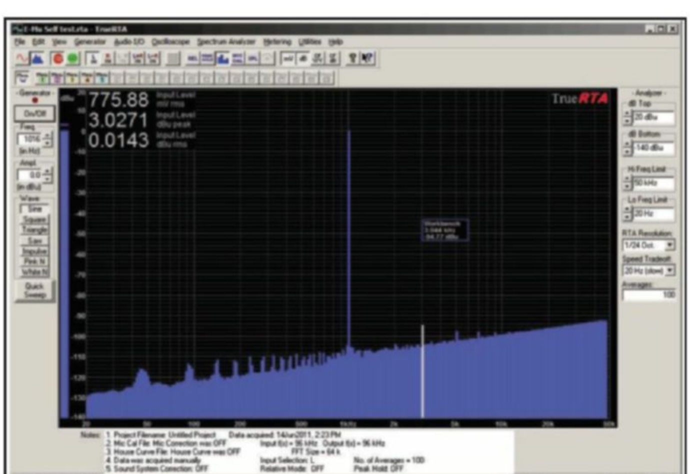
or for iOS and Android tablets—with the advantage that it would run much faster and with better graphics, audio measurement and speaker design tools will remain sadly restricted to old PC-based systems. Some of our readers will tell us that Windows XP is fine and that it does the job, but that's not the point. The world—and even Microsoft—is moving on and, as Apple has demonstrated with the migration to powerful ARM-based silicon, computers will never be the same, particularly when it comes to affordability and power for money. Also, the combination of modern embedded platforms and Linux have created an ideal environment for a new age of hardware tools—which is being enthusiastically embraced in the scientific instrumentation, data acquisition and test and measurement market application segments. And creating good quality hardware dedicated to audio test and measurements doesn't need to be expensive either, as largely demonstrated by the DIY community (see Jan Didden's Linear Audio Autoranger—https://audioxpress.com/article/fresh-from-the-bench-the-linear-audio-autoranger-mk-ii-interface), new companies such as QuantAsylum, or even the latest DATS system.



Simultaneous display of loudspeaker frequency response and impedance measured with MLSSA Acoustical Measurement software from DRA Labs. Unacceptable interface by today's standards.



It's a little bit better with ARTA on Windows, and at least the printed measurement results look a bit smoother.



TrueRTA Interface. A very powerful software full of valuable features and tools. Unfortunately, the graphic interface and tools are cryptographic and require an unnecessarily difficult learning curve.

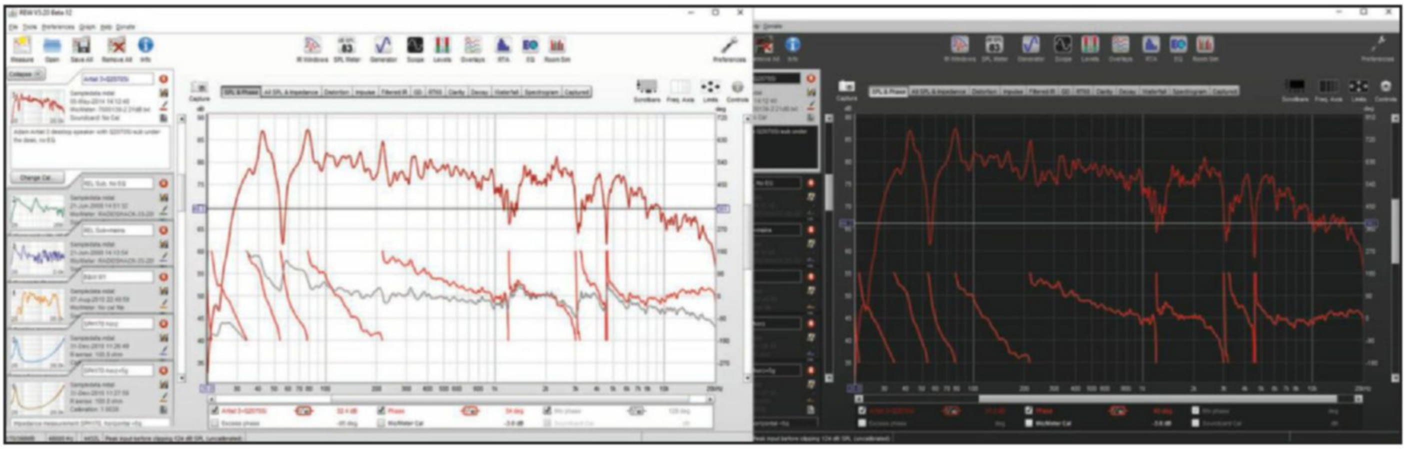
As demonstrated with the original DATS, which is now even better since the launch of the DATS V3 (reviewed by *audioXpress* in its March 2020 issue), while its software remains a Windows/PC application, the integrated solution is able to go beyond sound card based test systems, to combine an impedance analyzer, oscilloscope, LCR meter, signal analysis and generation, and satisfy all the requirements for speaker designers, including determining Thiele-Small (T-S) parameters, and much more.

For those looking for an affordable and relatively reliable solution, *audioXpress* strongly recommends the DATS V3 audio test system. It's one of the most complete and easy-to-use audio test systems available, allowing accurate measurements for any audio transducer or loudspeaker system. It combines USB interface with full-featured test software, and test leads. And it's been consistently updated. Our only wish would be to see a system like this with software for macOS and Native support for Apple Silicon—or a Linux version, for that matter.

Dayton Audio also offers the OmniMic V2 Precision Measurement System, an acoustic measurement system that includes a calibrated USB omnidirectional measurement microphone and intuitive software for the design and tuning of speaker systems and arrays. It's an affordable and reliable measurement solution, but we need to take into account that the OmniMic software only works with the serialized, calibrated OmniMic hardware.

For Serious Enthusiasts

And of course we need to mention the most popular free software for room acoustics, loudspeaker, and audio device measurements, which is REW (Room EQ Wizard—because it all started with room acoustics when it was created by John Mulcahy in 2004). And we mention it because the software is constantly updated, and because it was programmed using Java, it runs on Windows 64-bit, macOS on Intel, and Apple silicon (M1, M2, etc.), and even Linux. Over time, as Mulcahy admits, REW became primarily a measurement system. Combined with a calibrated USB microphone (e.g., the miniDSP UMIK), or an analog measurement microphone (e.g., the Dayton Audio EMM-6) and an affordable USB audio interface, users get a world of possibilities to explore. The audio measurement and analysis features of REW include audio test signals and many tools for loudspeaker measurements. The REW Pro upgrade even offers simultaneous measurement of multiple inputs with RMS averaging, adjustable weighting for each input, level alignment, and up to 16 input traces on the RTA in addition to the RMS average. And yes, there



Room EQ Wizard (REW) is by far the most complete software for acoustics and loudspeaker measurements. Although its donation-ware, unlike the vast majority of audio software, REW is constantly updated and, because of its Java foundation, it is available for Windows 64-bit, macOS for Intel, and Apple silicon, and even Linux.

are many forums where to ask questions to the most enthusiastic users, which explains why its user base continues to grow.

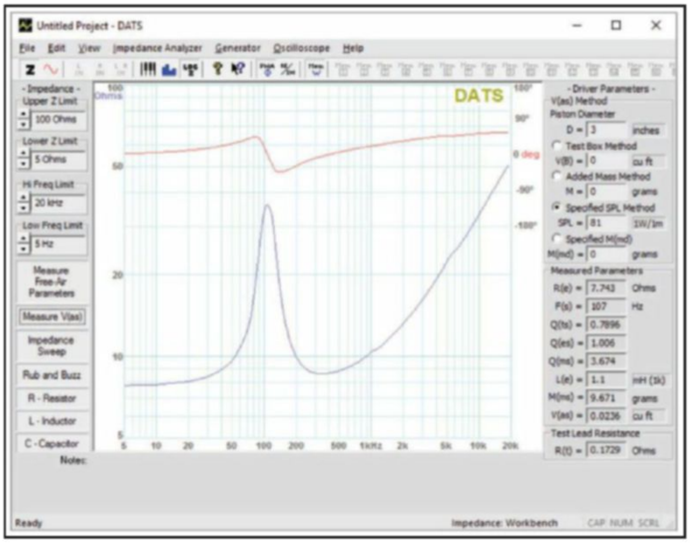
There are also countless speaker enclosure design calculators and simulation environments, some specializing in subwoofers, and others directly created by driver manufacturers for simulation with their own speaker catalog. Many of the most recent are already based on web tools that can simply be used from a browser—avoiding the problems of maintaining software packages for multiple operating systems. And of course, there are also countless applications and online calculators and simulators for crossover design—many of which, unfortunately, not only have scary websites, but are plagued by ads and adware. Caution is highly advised.

An updated alternative for speaker design—and a way to avoid those risks—is Speakerbench, a web-based application created by Jeff Candy (California) and Claus Futtrup (Denmark), combining several special-purpose modules aimed at analysis and design of loudspeakers. This online tool was designed for both the serious speaker designers and the loudspeaker DIY community in general.

While manufacturers rely on Klippel measurements, the vast majority of DIY and even many designers still rely on added-mass measurements for determining loudspeaker T-S parameters. The two authors of Speakerbench perfected a method of dual-added-mass, using three impedance measurements instead of two, allowing users to isolate the electrical from the pure motional impedance, and improving over the old-fashioned way of manually dialing in f_L, f_S, and f_H. Furthermore, with the three measurements, one can accurately compute a new model parameter that measures the level of driver viscoelasticity. And Speakerbench also offers the ability to create an advanced transducer model that is more accurate than the traditional Thiele-Small approach.

The two authors originally published an Audio Engineering Society (AES) paper in December 2017, which explained their novel dual-added-mass method, adding different masses to the cone. That is where the idea for the Speakerbench website was born, offering an approach that is perfectly valid for small-signal analysis, adding much higher precision.

The DATS V3 is one of the most complete and easy-to-use audio test systems available, allowing accurate measurements for any audio transducer or loudspeaker system.



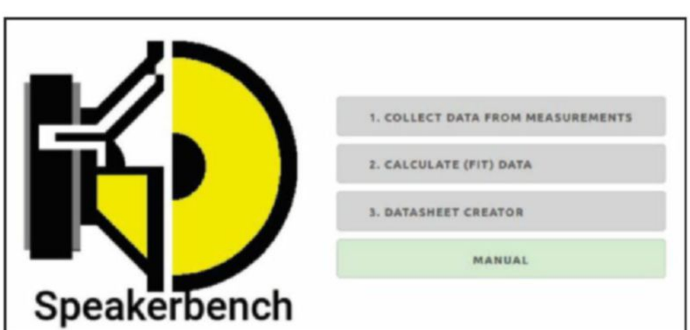
The DATS software interface evolves directly from TrueRTA and software created by TrueAudio's John L. Murphy.

The web application removes the math from the process and in the last major update Speakerbench gained a box calculation module that implements the proposed transducer model and enables users to export the object prepared from the Datasheet Creator module directly into the box module, automatically generating SPL, excursions, vent velocities, and other calculations. In this way, very high-quality transducer data can be generated even for DIY speaker builders. The Datasheet Creator offers a simple way to load one of a few predefined drivers into the creator or introduce our own measurements. A new complete dataset may be filed and shared with others. Loudspeaker manufacturers should actually make an effort to include their own data to the Datasheet Creator, so that they are offered in the predefined list.

Being a web-app, Speakerbench.com works on any computer, tablet, or smartphone with a browser without any software installation required. And the online version is always up-to-date.

Our readers should refer to the two articles published in *audioXpress'* August and September 2021 issues. An earlier article about Speakerbench, describing the methods and the research, including a comparison with existing measurement tools and industry practices, was also included in the 2020 edition of the *Loudspeaker Industry Sourcebook*. The two authors have also created a series of video tutorials on a dedicated Speakerbench YouTube channel.

Available from Dayton Audio and Parts Express, the OmniMic V2 Measurement System was developed in collaboration with Liberty Instruments, inventors of the Praxis analysis software, now discontinued. The OmniMic V2 uses a calibrated USB microphone and provides quick and accurate acoustic measurements including SPL, spectrum analyzer (FFT, or RTA), frequency response with phase and impulse response, harmonic distortion, RT60, polar plots, and much more.



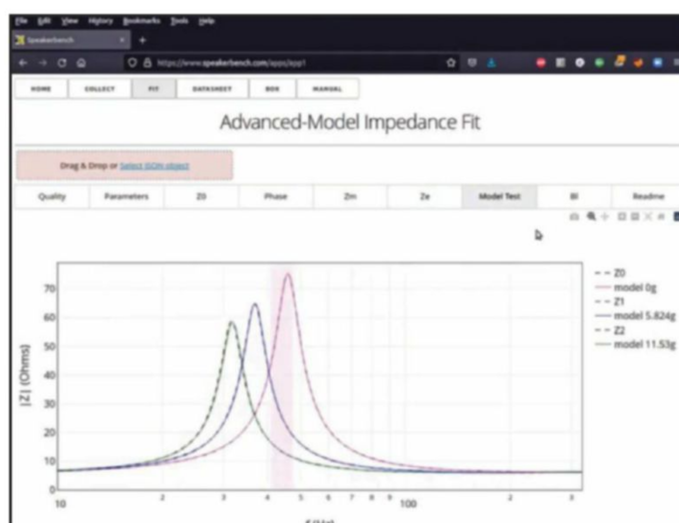
Speakerbench is a web simulation tool and calculator that takes things further for speaker designers, based on a dual-added-mass method to determine small-signal parameters from three impedance measurements, adding much higher precision.

For Development

A step up to more professional environments—and 64-bit software—implies learning to use tools such as those from Ahnert Feistel Media Group (AFMG), which are more oriented for manufacturers and particularly pro audio applications. AFMG was founded from the collaboration between Dr. Wolfgang Ahnert and Dr. Rainer Feistel (since 1982), starting to develop and perfect a wide range of measurement and simulation software solutions for applications in pro audio and room acoustics of which EASE (Electro-Acoustic Simulator for Engineers) is by far the most widely recognized.

Today, AFMG produces advanced tools, such as EASE SpeakerLab, for loudspeaker data analysis and GLL Creation; EASE Focus, an acoustic simulation program for 3D modeling of line arrays, sub arrays, digitally steered columns and conventional loudspeakers; AFMG SysTune for professional audio measurements and live sound system adjustments; FIRmaker for sound system design and optimization (now including beamshaping); and many other tools for acoustics measurements.

AFMG also offers Electronic and Acoustic System Evaluation and Response Analysis (EASERA), a complete software for electronic and acoustic analysis, including a signal generator and real-time analyzer. EASERA provides both data acquisition with a variety of stimulus signals including Time Delay Spectrometry, sweeps MLS or noise excitation signals and a post processing engine to calculate all acoustic functions or measurements, combining all the features required for acoustic analysis in one unified package.



Speakerbench generates a key performance test, in which model predictions of impedance are plotted on original data from measurements. The difference in peak height between the three measurements is due to frequency-dependent damping and is a consequence of viscoelasticity.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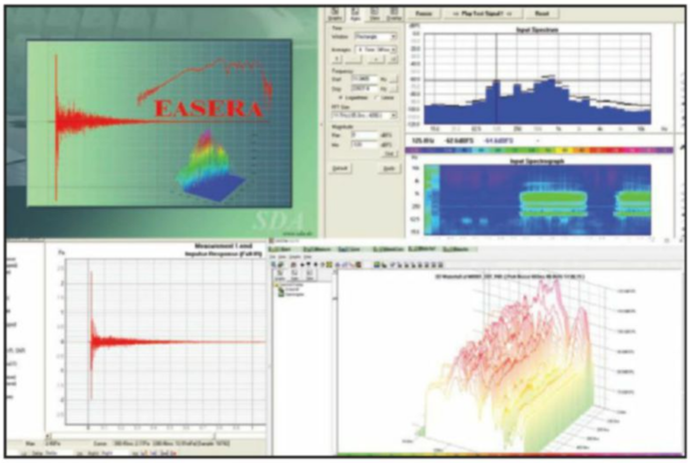
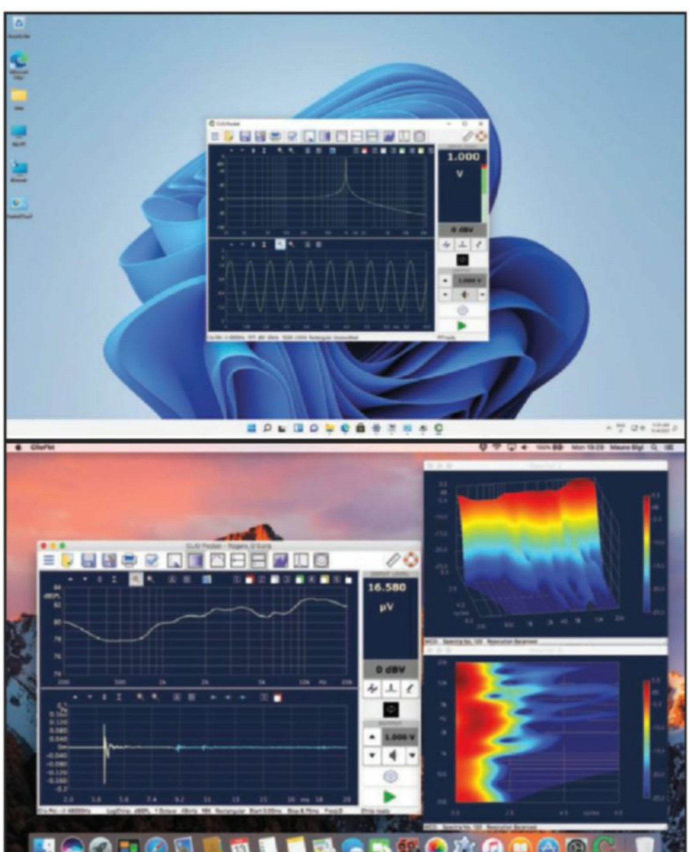
Also of note, apart from supporting the modern versions of Windows and macOS, AFMG is now introducing support for Apple computers with M1/M2 processors in all its tools.

And speaking of Apple, Italian audio test and measurement company Audiomatica, continues to update is popular CLIO software to run on the most recent versions of macOS and make the best use of Apple silicon configurations, while also supporting Windows 11.



AFMG's Electronic and Acoustic System Evaluation and Response Analysis (EASERA) is a powerful audio an acoustics test and measurement platform that is in urgent need of an interface update for the 64-bit era.



Audiomatica CLIO Pocket software version 2.11 runs in Windows 11 without any issue, and 2.12 ×64 for macOS already runs on macOS 11 and Apple silicon.

Audiomatica CLIO and CLIO Pocket systems continue to be an excellent option for full system identification and characterization, combining different measurement techniques, including Maximum Length Sequence (MLS) and LogChirp analysis, sinusoidal sweeps, FFT, RTA and "Live" Transfer Function. CLIO allows measuring frequency response, impedance, or other parameters, combining multiple measurements to assess its consistency, offers full Thiele-Small parameters for loudspeaker characterization, directivity analysis for loudspeakers, such as 2D color maps, circular or waterfall-like plots, 3D balloons analysis for complete spatial characterization, and much more.

The Professional's Tools

Before there were dedicated audio analyzers, speaker designers learned to rely on scientific test and measurement instrumentation, and fortunately for audio designers, the accuracy and sophistication of hardware and software solutions continues to evolve and improve. As noted in our latest Test and Measurement Market Update published in the March issue, there are clear trends for decoupling the required software from hardware interfaces and dedicated audio analyzers, while the software is becoming more sophisticated, modular, and affordable. Online and cloud-based tools are also starting to be integrated—following the general trend in the test and measurement segment in general.

Since 2019, Audio Precision has offered its APx500 Flex, a completely new class of test and measurement solution designed to support the use of APx audio measurement software with ASIO-capable third-party audio interfaces or sound cards. This allows manufacturers and developers to cost-effectively deploy the measurement capabilities they need for their task at a much lower price. And in 2021, Audio Precision also introduced software subscriptions as an alternative to perpetual software licenses for its APx500 software. This offers another flexible option to access all the power of this incredibly versatile software, which continues to expand with every new release.

Audio Precision's Electroacoustic test suite software options form a comprehensive solution from low-noise analog-to-digital processing to loudspeaker output performance. Measurements, results, reports, and automation can be easily shared among APx analyzer models, enabling designers and production engineers to collaborate and ensure quality, even when separated by great distances.

As an additional trend resulting from the separate investments in hardware and software by the leading vendors in audio test and simulation, it is becoming easier to find hardware options that are adjustable to repeated processes in product lines at lower cost, while calibrated tools and methods to load calibrations including Transducer Electronic Data Sheet (TEDS) calibration, are becoming common in professional applications. There's the hope that with affordable audio analyzers on the market, such as those from QuantAsylum, the need to rely on Frankenstein setups using audio interfaces designed for home studios, will become a thing of the past—even for DIY enthusiasts.

Of course there are numerous companies that rely on building boxed audio analyzers mostly for manufacturing applications. An example is K&K International, a small company from Denmark that inherited the instrumentation business formerly from Ortofon. The company offers test systems and QC for the loudspeaker industry, also covering headphones testing with its P630 and P900 acoustic analyzers. The systems are 19" boxes using PC cards inside and are built for high-volume production testing. The original Ortofon P400 that was introduced in 1983 was the inspiration for this K&K range of products. The most recent product from the company is the P900, which offers faster test time for fully automated environments and comes with dedicated software for Windows 10 and 11 that offers visualization, logging, and statistical analysis.

Also from Denmark, but with a growing focus on software is Loudsoft, the company founded in 2000 by Peter Larsen, translating his vast experience working with the world's top speaker companies. Loudsoft offers loudspeaker and headphone design software, test equipment, and consulting services but the company has been paying particular attention to its FINE range of design and simulation software of magnets, voice coils, domes and cones, box design, and more recently suspensions and surrounds (FINEMotor, FINECone, FINEBox, and FINEsuspension, respectively). Gradually, Loudsoft has added important new features and updates for this FINE family of software tools.

Of course the most important product from Loudsoft—significantly updated in 2021—is the FINE R+D software, which among other things, gained a signal generator with pure sine wave outputs and user set sweeps. The new signal generator enables users to select both frequency (Hz) and level (Vrms), as well as a continuous (repeating) sweep with a start and stop frequency, or duration (s) and level (Vrms). By selecting a limited range (say 100Hz to 500Hz at 2.7s) users get a very detailed sweep, which is ideal for listening and detecting problems.

When discussing new alternatives in hardware for audio test and measurement, and speakers in particular, it's also important to highlight Echo Test + Measurement, which is not a new company, but is effectively relaunching as a company fully focused on audio test and measurement. The company headquartered in Santa Barbara, CA, has roots going back to 1978 as Street Electronics, building text-to-speech cards for the Apple II and licensing custom DSP hardware and software to large computer and semiconductor companies. From the 1990s through the first two decades of the 21st century, Echo became a leading manufacturer of pro audio sound cards for computer music and recording, which is today the foundation for its current test and measurement range.

Since 2014, when Echo designed the original AIO modular Test System, it sold tens of thousands of units to manufacturing plants all over the world, leading to a complete transition to a test and measurement business exclusively, and the company being renamed to Echo Test + Measurement.

The reference to Echo in the context of loudspeaker testing is now mandatory, since it offers and affordable, and proven range of audio interface solutions that work with the most popular software packages in the industry, including Audio Precision's APx, Listen SoundCheck, LabVIEW, and ARTA.

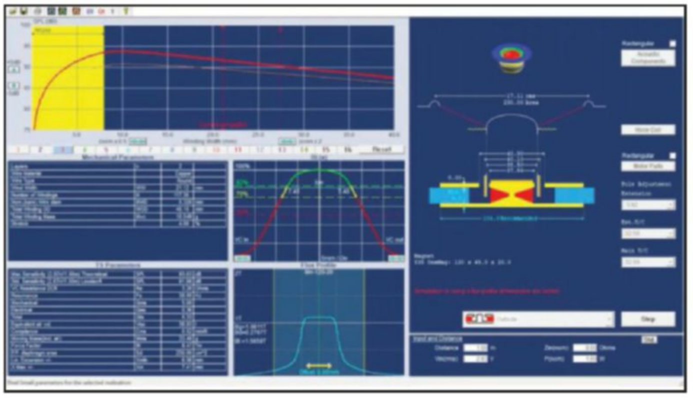
Expanding its AIO range already this year, Echo Test + Measurement now offers a very complete family of modules that are suitable to both R&D and production line testing, all using test industry standard connections and interfaces, such as IEPE microphone connections on BNC connectors with CCP power, and TEDS readers built into each channel. Output options include line level, headphone, and power amplifiers on EuroBlock connectors with added sense resistors built in for making impedance measurements. There is even a 10-channel TDM bus module for making chip level (I^2S) measurements. Other available options include battery simulation, GPIO inputs and outputs, and pressure, temperature, and humidity sensors (unique to Echo) in the C range. All these features that come standard on Echo AIO are costly add-ons to traditional sound card-based test setups.

Any AIO Test System can be controlled using a simple control app, the AIO Command-Line application, enabling the creation of test sequences, as an example, and finally using the AIO Interface Library, with a standard C-language interface. All the modules are cross-platform USB 2.0 audio class compliant, and support WASAPI and CoreAudio drivers in addition to ASIO.

With the release of APx500 software version 7.0, Audio Precision adds support for GRAS' SysCheck2-capable microphones, and introduced Fast Sweep signals in open-loop testing scenarios, such as the testing of smart speakers.

Listen's SoundCheck

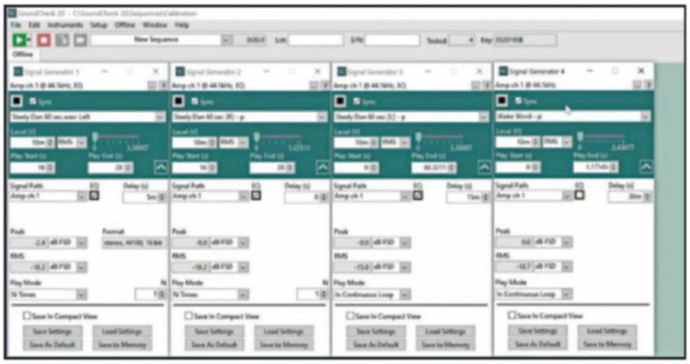
When it comes to the tools that are widely embraced by audio manufacturers and technology companies dealing with audio



Loudsoft's FINE family of software tools for speaker simulation and design contains tens of thousands of FEA simulations, offering calculations and prediction of all relevant parameters for designing the magnet system, voice coil, cones, spider, and surrounds for all kinds of speakers and microspeakers.

The Echo Test + Measurement AIO range of modular USB interfaces includes the AIO-AC multichannel model for production line testing and QA/QC verification with two amplifier outputs, two balanced line-level/headphone outputs, four mic/line analog inputs, eight GPIO inputs, eight GPIO outputs, 5V DC power supply, and variable 2V DC power supply for battery simulation in a single unit. The model on top is the Echo AIO-S2, with four-channel mic/line analog inputs in a single unit. The latest C range models even include pressure, temperature, and humidity sensors, which are essential in production environments.

SoundCheck 20, Listen's flagship audio analysis software, introduces a new state-of-the-art perceptual distortion algorithm for end-of-line testing—Enhanced Perceptual Rub & Buzz. This algorithm performs like a human ear under normal listening conditions with the background noise of a manufacturing environment.



Two of SoundCheck's virtual instruments, the multichannel Real Time Analyzer (RTA) and the Signal Generator, have been significantly upgraded in version 20. The multichannel RTA now includes expanded functionality for real-time observation of audio signals. Multiple signal generators can be synchronized or delayed for phase control during playback.

development, some of the systems available are in a league of their own. Based in Boston, MA, Listen is one of the most experienced and recognized companies in audio test and measurement. Its core product is the SoundCheck software, which has been continuously improved with cutting-edge features that directly reflect much of the research conducted internally by the company (and its founder and president, Steve Temme), but also the latest industry academic and scientific research, combined with a deep knowledge of the unique requirements of the industry.

Listen serves organizations working in traditional speaker and audio product design and manufacturing, as well as headphones, automotive audio, and technology companies working in the field of voice and connected (smart) systems.

All that is reflected by the deep level of advanced features integrated in Listen's SoundCheck audio test software, which was just recently updated to its v20 release, available for both Windows and macOS.

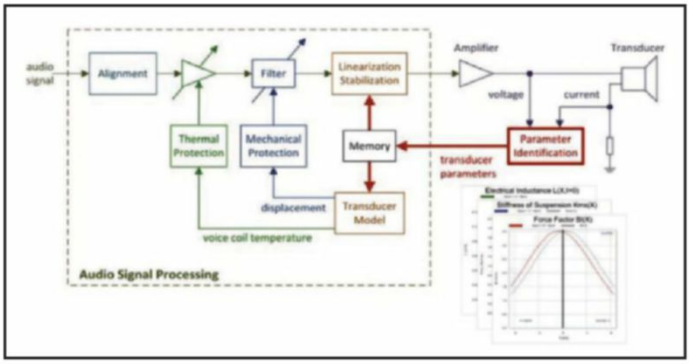
SoundCheck 20 was released a year after SoundCheck 19, which already introduced powerful advancements for users making multichannel, multifunction, and high-resolution audio measurements, including a new multichannel Real-Time Analyzer (RTA) and new room acoustics measurements. With the SoundCheck 19 announcement in 2021, Listen also introduced its new high-resolution multichannel interface, AmpConnect 621. Another important SoundCheck 19 enhancement was the option to use WASAPI-exclusive mode with Windows 10 audio devices. This advanced interface offers superior communication with audio devices, allowing SoundCheck to bypass the Windows audio engine, and send audio streams directly to the device. This delivers lower latency and ensures that sample rate conversion from the built-in Windows audio engine will not occur.

SoundCheck 20 introduced Enhanced Perceptual Rub & Buzz, Listen's state-of-the-art perceptual distortion algorithm, which promises to be a complete game changer for end-of-line testing and production line quality control. And this latest update expands even more the features required for audio measurement and full characterization of smart speakers, infotainment systems, headphones and hearables, and other voice-activated and multichannel devices, including communications testing. The multichannel RTA was upgraded with real-time calculations, the signal generator was significantly improved, allowing also users to play wav files and complex waveforms created by the stimulus editor, and this version introduces a new optional module for Perceptual Objective Listening Quality Analysis (POLQA). This new module may be used to assess the impact of noise reduction algorithms, evaluate Bluetooth degradation due to packet loss, or to analyze distortions introduced into the audio path. Naturally, it can be used within sequences to accelerate and simplify test procedures. A very significant software expansion that deserves attention from the industry and any speaker designer.

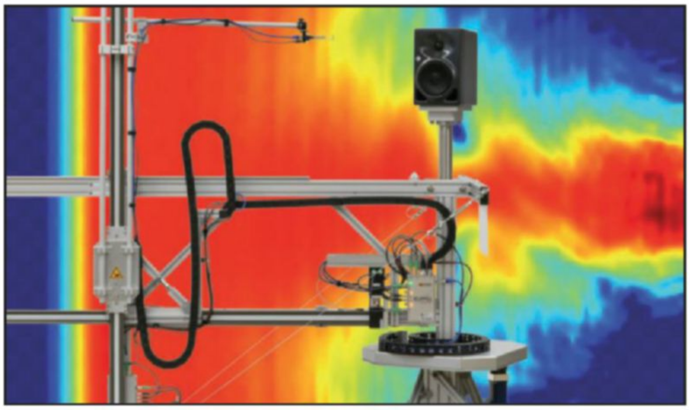
Klippel

When it comes to speaker test and measurement—and much more—the company that took things to a whole other level is Klippel GmbH. Founded in 1997 in Dresden, Germany, Klippel has been constantly expanding its portfolio of products and solutions based on a consistent research effort, and working closely with international standards organizations and industry associations. Dr. Wolfgang Klippel, the company's founder, remains the company's president and director of research and development while working closely with the Technical University Dresden, where he is also honorary professor.

The company's relentless pace of innovation is also the result of a constantly expanding team of talented young engineers, now led by Benjamin Barth, who was appointed general manager in May 2020, allowing Wolfgang Klippel to focus more on research and education. Barth has led Klippel's R&D projects since 2015 and maintained Klippel consistently at the forefront of leading technical projects in test equipment for electro-acoustical transducers and audio systems, allowing a unique perspective on the root causes of signal distortion and defects, and giving practical indications for improvements in design and manufacturing of audio products.



Presentation of Klippel Controlled Sound (KCS) technology, introducing a new paradigm in active loudspeaker control and correction in order to prove more sound pressure output, active protection against overload, cancellation of nonlinear distortion, and linear targeted performance, allowing manufacturers to deal with production variance, and create more efficient speakers with lower weight and size.



Klippel's Near-Field-Scanner 3D (NFS) offers a fully automated acoustic measurement of direct sound radiated from the source in the 3D space, enabling precise measurement of directivity, sound power, SPL response, and many more key parameters of a loudspeaker or audio system in near-field applications. Utilizing a minimum of measurement points, a comprehensive data set is generated containing the loudspeaker's high-resolution, free-field sound radiation in the near and far field. Due to the holographic nearfield measurement approach no anechoic room is required for precise measurements.

The company currently employs 45 people and is working on novel techniques developed for simulation, control, and measurement of loudspeakers, headphone drivers, and microspeakers.

Examples of Klippel leadership include the introduction of Multi-Tone Measurement (MTON) and Multi-Tone Distortion Task (MTD) measurements in its dB-Lab Software, which enabled among many other things to pinpoint the SPLmax according to IEC 60268-21, as well as the continuous max SPL, according to ANSI/CEA-2010-B and ANSI/CEA-2034 standard measurements.

The Klippel Near-Field Scanner (NFS) is another major contribution to measurements outside of anechoic chambers, with the corresponding Near-Field Scanning Software, making it easier to perform asynchronous testing of Bluetooth audio streaming systems and offering a fully automated acoustic measurement of direct sound radiated from the source under test. Directivity, sound power, SPL response, and many more key figures are obtained for any kind of loudspeaker and audio system in near-field applications (e.g., studio monitors, mobile devices) as well as far field applications (e.g., professional audio systems). Utilizing a minimum of measurement points, a comprehensive data set is generated containing the loudspeaker's high resolution, free field sound radiation in the near and far field.

While Klippel's reputation in the industry was built on the Klippel Analyzer System for R&D applications and QC (end-of-line testing), more recently, the company introduced the Klippel Controlled Sound (KCS) technology as a new paradigm in active loudspeaker control and correction. KCS uses the transducer itself as the sensor to identify transducer parameters by monitoring voltage and current at the speaker terminals. The adaptive control structure is based on electro-acoustical modeling and combines real-time monitoring with active protection against thermal and mechanical overload, nonlinear distortion cancellation, system alignment and stabilization of the voice coil position.

Klippel is combining this game-changing KCS approach with the Klippel Linear Simulation Module (LSIM), a software that allows the calculation of loudspeaker efficiency within seconds, allowing designers to improving the power-efficiency of their speakers.

Wolfgang Klippel also had a decisive role in the latest IEC 60268-21 measurement standards and expanding the industry practices to incorporate testing of active systems with DSP— work that is now bringing revolutionary advancements to the industry.

Resources

Ahnert Feistel Media Group (AFMG) |www.afmg.eu/en

Arta Labs | https://artalabs.hr

Audiomatica | www.audiomatica.com/wp

Dayton Audio | www.daytonaudio.com

Echo Test + Measurement | www.echotm.com

K&K Development | www.kk-int.dk

Klippel GmbH | www.klippel.de

Listen, Inc. | www.listeninc.com

Loudsoft | www.loudsoft.com

Speakerbench | https://speakerbench.com

True Audio | www.trueaudio.com

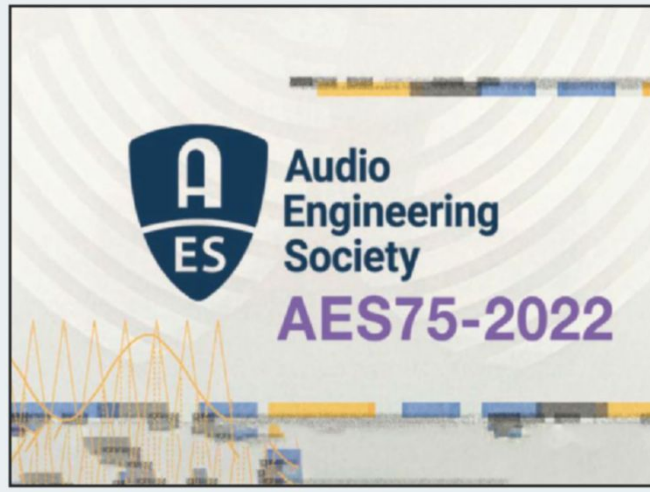
New AES75 Standard for Loudspeaker Performance Measurement

More than three years past since M-Noise was proposed as a new test signal for standardized measurement of a loudspeaker system's maximum linear output, in March 2022, the Audio Engineering Society (AES) has announced the adoption and publication of AES75-2022, "AES standard for acoustics—Measuring loudspeaker maximum linear sound levels using noise." The new standard addresses the need for a practical and cohesive procedure for the prediction of loudspeaker performance.

First introduced in 2019, M-Noise was developed as a means for determining the maximum linear output levels of loudspeaker systems when used in music reinforcement applications. To assure consistent repeatability, a test procedure was developed that specified test parameters and described requirements for associated microphones, signal generators and analyzers.

AES75 details a procedure for measuring maximum linear sound levels of a loudspeaker system or individual driver using the M-Noise test signal. Mathematically derived from analysis of hundreds of music selections spanning all genres, M-Noise uniquely exhibits a crest factor characteristic of music program signals. The specified test procedure determines maximum linear sound levels by incrementally increasing playback levels until reaching a stop condition: an unacceptable change in the transfer function's magnitude or coherence. Specific procedural steps are outlined to accommodate both self-powered and externally powered loudspeakers.

AES75 was the result of extensive work by the AES Standards Committee's SC-04-03-A Task Group, co-chaired by Merlijn van Veen (senior technical support and education specialist at Meyer Sound) and Roger Schwenke, Ph.D. (Meyer Sound senior scientist and innovation steward). Use of the AES75 procedure will eliminate uncertainties caused by published loudspeaker data that state



a maximum SPL output but often are ambiguous regarding the test signal used or other relevant test parameters.

Predictive evaluation of loudspeaker performance based on published specifications has traditionally been challenging due to inconsistencies in both measurement procedures and in how measured parameters are reported. "Until now," explains Schwenke, "reading an SPL number on a datasheet often inspired more questions than answers regarding test signals used and procedures for measurement. Most important to the end user is how the loudspeaker will perform with typical audio signals and whether the numbers can be compared apples-to-apples with numbers from one datasheet to another."

"AES75 addresses these issues," Schwenke continues, "by providing a detailed procedure as well as a specific test signal, M-Noise, whose RMS and peak levels as functions of frequency have been shown to better represent typical program material. Furthermore, AES75 is designed to be independently verifiable, using analyzers and microphones typically used by audio professionals. By being independently verifiable, AES75 provides system specifiers and uses a much more enforceable metric to use in quotes and architectural specs."

The M-Noise test signal is based on Meyer Sound's analysis of hundreds of music selections spanning all genres. The procedures documented in AES75 provide measurement of maximum linear sound levels by incrementally increasing playback levels until the magnitude or coherence of a loudspeaker's acoustic reproduction of the M-Noise test signal reaches an unacceptable state. The AES75 test procedures cover performance measurements of both self-powered and externally powered loudspeakers.

The AES Standards Committee's SC-04-03-A Task Group was co-chaired by Roger Schwenke, Ph.D. (Meyer Sound senior scientist and innovation steward at left), and Merlijn van Veen (senior technical support and education specialist at Meyer Sound at right).

The AES75 standard as well as the M-Noise signals and coherence test tracks are also available for download.

www.aes.org

Design and Development of the Holoplot X1 Matrix Array

By
Michael Kastner
(Head of Product, Holoplot)

The last few years have been ones of frustrated innovation for much of the pro audio industry. Many product designers and manufacturers in the audio industry the world over have been making huge leaps forward in hardware, software, and interoperability that they have been largely unable to show off thanks to a disruption to live events of all kinds. Following the return of trade shows in 2022, the lid has been lifted on a whole host of innovations in the pro audio space, and among them Holoplot has emerged as a standout with its X1 Matrix Array sound system. This article is an inside account about how the Holoplot team has been pushing the boundaries of what is possible in the pro audio space.

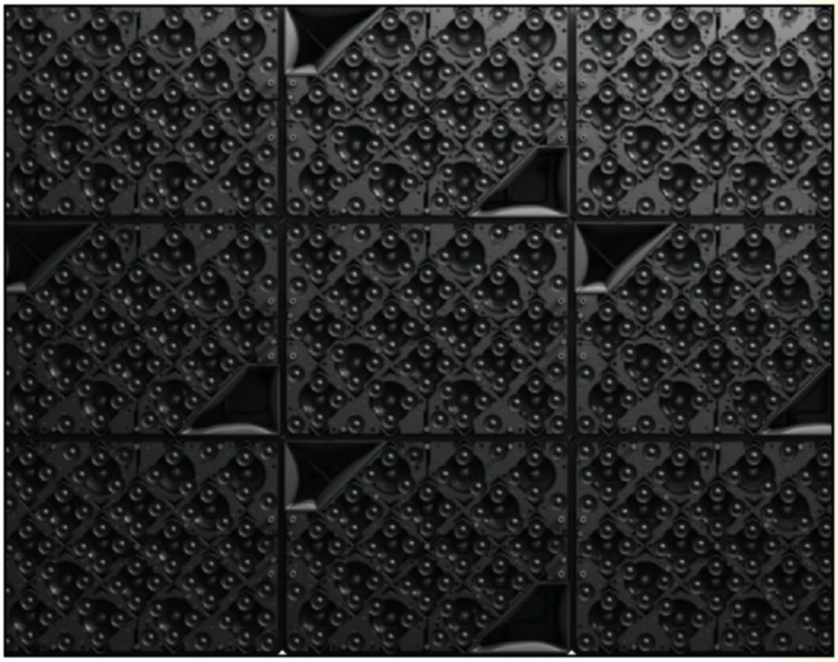
Headquartered in Tempelhof, Berlin, Holoplot is a company on a mission to transform the audio industry. In 2021, it launched the X1 Matrix Array—a product that has introduced a whole new category of sound system. The Matrix Array is the logical continuation of two already existing landmark innovations in pro audio—the line array and electronic beamsteering—and provides control of sound in both horizontal and vertical axes through a matrix of loudspeaker drivers.

Conventional high-performance line arrays are by now a commodity in our industry, but high-performance has been hard to achieve for beamforming loudspeakers historically. The prevailing preconception in the industry is that beamforming loudspeakers cannot perform well in full-range applications—or that they are very costly, and very complex to design and manufacture. With X1, Holoplot and our team set out to prove these assertions wrong.

The X1 product series features two audio modules. The Modul 96 (MD96) is a two-way audio module integrating 96 drivers in a two-layered matrix design. The Modul 80-S (MD80-S) is a three-way audio module with 80 drivers in its first two matrix layers and a sensor-controlled subwoofer driver in its third layer. This matrix arrangement of loudspeaker drivers of different sizes—in a unique multi-layered configuration—enables control over sound propagation across a large frequency range, while providing the SPL and quality desired for high-performance sound reproduction.

With this configuration, X1 offers a wide range of previously inaccessible sound control capabilities. Each module can create up to 12 individual and independent beams, able to reproduce differing content at the same time. With just one centralized Matrix Array, sound propagation can be tailored to cover an audience area and precisely target one or multiple zones.

The X1 concept did not emerge fully formed in some kind of "Birth of Athena" moment. Quite the contrary, the concept and creation of X1 is the result of tireless work of multiple specialists drawn together from across the music and computer technology, consumer electronics and pro audio space.



Holoplot's X1 series features the Modul 96, a two-way matrix module, and the Modul 80-S, a three-way matrix module integrating a sensor-controlled subwoofer. The MD96 and MD80-S can be arrayed horizontally and vertically to create differently sized X1 Matrix Arrays. The configuration of a Matrix Array is defined based on the required beamforming capabilities (such as areas to cover homogeneously and desired control over low frequencies), sub-frequency integration, and maximum SPL requirements.

The Holophonic Approach

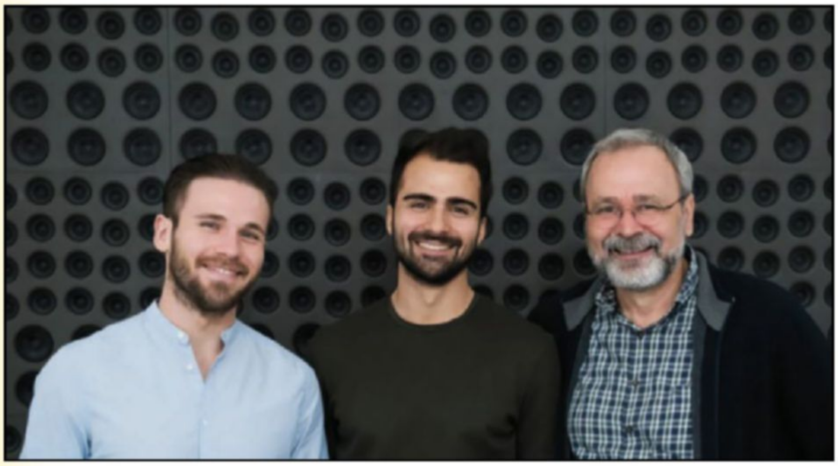
Beginning its journey back in 2011, up until 2016 Holoplot was primarily a research company run by Tonmeister Helmut Oellers, and DSP engineer Adrián Lara Moreno. The original vision for a loudspeaker in Matrix Array format came from Helmut—with his earliest ideas still published and publicly accessible at www.syntheticwave.de. Helmut has a long history of trying to understand existing audio reproduction methods and the drawbacks they present, such as the limitations of phantom sources or the acoustic overlap between the room acoustics present in a recording and room acoustics of the space where the recording is being reproduced. To overcome the limitation of those methods, Helmut came up with what he named "holophony." His holophonic approach involves the use of Wave Field Synthesis and two-dimensional speaker arrays, which was the foundation for his first patent back in 2004. Out of this patent, Holoplot was born to turn that theoretical vision into reality.

Researching the Matrix Array concepts was the main objective of Helmut and Adrián during the initial years. The arrival of Roman Sick as Holoplot CEO in 2016 enabled them to turn concepts and prototypes into a commercially viable product that could address the needs of the pro audio market.

"I met the Holoplot team when they were just four people working from a garage outside of Berlin, but I very quickly realized the game-changing potential of what they were trying to do. When the opportunity presented itself to lead the company I decided to commit and signed on with the original engineers on board. Growing the team, setting up our headquarters in Berlin, and commercializing the product was an intense but extremely exciting time. I was intrigued by the potential, as well as by the challenge of developing a radically new technology, which I'm convinced will change the world," recalls Roman Sick.

"X1 is a powerful toolset, which offers a wide range of benefits for a variety of use cases across multipurpose arenas, concert halls, immersive venues, conferences, places of worship, and more. The unprecedented control over sound X1 can achieve allows us to minimize the impact of the architecture and acoustics of the room, and the system just sounds phenomenal," he adds.

X1 was preceded by Holoplot's original Orion system. Orion wasn't designed to be a fully-fledged solution for a multitude of applications, but rather as a product that could demonstrate the fundamental ideas of the combination of 3D Audio Beamforming and Wave Field Synthesis (WFS). It was designed for niche applications that aren't serviceable with "standard" audio technology and consisted of the Io Audio Module and Eta Core processor. Utilizing the knowledge gained from more than six years of research, the Orion system was capable of



Roman Sick (center) joined Holoplot as CEO in 2016 to expand the efforts of Tonmeister Helmut Oellers (right) and DSP engineer Adrián Lara Moreno (on the left).

generating multiple highly efficient audio wave fronts, ranging from spherical to cylindrical and planar, and could control them precisely in the vertical and horizontal axis.

As Adrián Lara Moreno, Head of Research at Holoplot explains, "Orion is the genesis of X1, but already then was the culmination of many years of research turned into a product designed for specific use cases in the pro audio space. When we transitioned from Orion to X1, we kept the knowledge and experience from the past, and added new perspectives and invaluable input from notable industry experts to continue the journey."

Assembly of the first beta prototype of X1 at our offices, before Holoplot had a dedicated production site. The first beta prototype was milled out of plastic, and played for the first time in December 2018.

"The design criteria for Orion were very loose originally, but Helmut and I were in agreement it had to be a scalable, two-dimensional system. Essentially Orion combined loudspeaker array technology and electronic beamsteering in a way that no one else had attempted before. Holoplot's approach to Wave Field Synthesis was different from the traditional approach—most had combined signal processing with traditional point source loudspeakers, and required the installation of those loudspeakers all around the listener to create an immersive effect. Holoplot's proprietary loudspeaker and processing technology allowed us to handle the wave propagation in a completely new and more finely controlled way. We were for example able to create virtual sound sources through room reflections—meaning it was for the first time possible to expand traditional WFS and create a fully immersive audio experience utilizing just a single Matrix Array."

With the Orion system Holoplot completed a variety of significant tests and projects. The company successfully deployed a centralized WFS immersive soundscape of the Vienna Philharmonic Orchestra and created sound reinforcement systems with minimum SPL loss across many hundreds of meters.

Orion was also deployed to deliver immersive audio content for Nintendo at the 2019 Electronic Entertainment Expo in Los Angeles, CA. The well-known gaming company called upon Holoplot to deliver targeted sound effects utilizing Holoplot's multiple content zoning capabilities for a trade show gaming installation marking the launch of their action-adventure game *Luigi's Mansion 3*.

Another highlight for Orion was scoring some of the highest Speech Transmission Index (STI) measurements ever recorded on a train station platform. They were so incredibly high, that the acoustic consultant wondered whether his measuring device was malfunctioning and re-ran the test—with the same unbelievable results of 0.61 STI over a 180m distance. For reference, an STI value of 0.5 would have already represented an enormous improvement for the station compared to the performance of its installed system.

"Orion gave us the chance to develop, experience, and iterate Wave Field Synthesis and 3D Audio Beamforming. We see the combination of these technologies as a logical consequence of previous industry advancements: a natural evolution of existing technology—but a revolution in its potential applications, on par with the introduction of the Line Array to the pro audio world in the 1990s," Moreno adds.

The X1 Project

Holoplot as a company emerged from the desire to radically change the way we can manipulate and utilize sound. It acted upon the realization of major shortcomings in the pro audio industry—not much had changed in the way sound systems were deployed since the introduction of the line array, and the Holoplot team was determined to create a new, unsurpassed loudspeaker solution that would start a new chapter on what's possible in sound reproduction and control.

The creation of X1 was accelerated by the need to create something custom and never done before, to cater to the specific requirements of one of the most ambitious AV venue projects in this decade, set to redefine audience experience standards inside the live entertainment industry. This was the catalyst for Holoplot to evolve its technology by an order of magnitude to truly deliver on concert-level performance requirements.

Essentially, Holoplot came to the realization that a new product had to be developed, and needed to be bigger and much louder. To achieve this, the company extended its engineering team with Michael Hlatky as the Development Lead on the engineering side, Philippe Robineau as Acoustic Lead, and Evert Start as DSP Lead.

As Michael Hlatky, now Head of Engineering at Holoplot details, "Philippe `Bob' Robineau approached us in 2018 and very modestly suggested he might know a thing or two about designing loudspeakers, and that he thought he could be of help. Having started his career as the Chief Engineer at Nexo, to later leading R&D at Turbosound, Focal, Tannoy, and Adam Audio, we couldn't have wished for anyone better on the electro-acoustic design of X1. Philippe was instrumental in the creation of the multi-layered Matrix Array configuration, enabling control over sound propagation across a large frequency range, while providing the SPL and quality desired for high performance sound reproduction. X1 is the culmination of Philippe's experience in designing loudspeakers for the pro audio industry, spanning a career of more than 50-years."

"Philippe suggested extending the team with Evert Start, who himself brought invaluable years of pro audio experience when he joined Holoplot in 2019. He'd been working on Wave Field Synthesis since university and later, as a research engineer at Duran Audio (acquired by Harman). He introduced the world's first beam-shaping algorithm for the Intellivox loudspeaker arrays, which marked the starting point of the successful adoption of beamsteering column loudspeakers inside the pro audio industry."

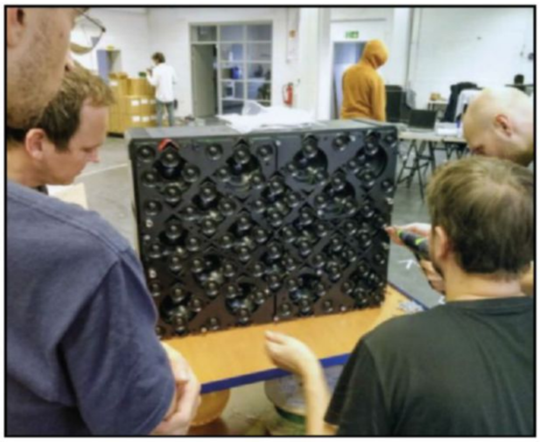
August 2018: Phillipe Robineau examines the first alpha prototype of an X1 MD80-S. It was made of wood and driven by external off-the-shelf electronics.

Creating X1 certainly hasn't been a simple process, or a natural one. The issues Holoplot faced in bringing this technology to life have been many. For example, X1 uses an unheard of number of drivers, each one individually driven and signal processed via 8700 digitally controlled filters computed in real time.

The amount of amplifier channels required to achieve this inevitably leads to a huge amount of heat to manage, and a seemingly insurmountable amount of DSP needs to be computed in real-time. To put this into perspective, a relatively small 3×3 X1 Matrix Array usually has more amplifier channels than the largest installation of conventional loudspeaker systems.

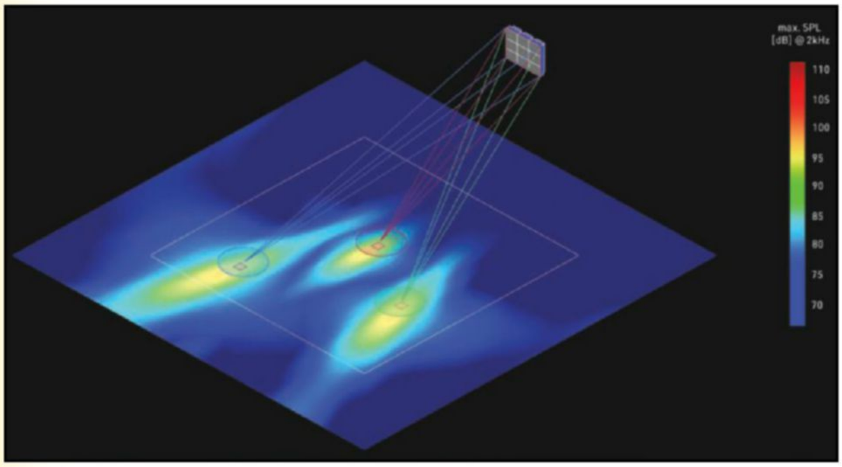
Michael Hlatky details how the project evolved: "I joined Holoplot in May 2018 to manage the X1 development. At that point the general concept of a three-layered Matrix Array already existed, and,

The Holoplot booth at Integrated Systems Europe 2022 in Barcelona.



July 2019: Assembly of the first integrated X1 prototype at the new Holoplot factory site in Berlin.

October 2019: First large-scale testing of a fully integrated X1 array comprising 20 MD96 modules. Holoplot spent four weeks inside the Marlene Dietrich Halle in Potsdam, collecting measurements to validate the hardware and software.



With Holoplot's 3D Audio Beamforming technology, each array is able to generate one or multiple sound beams that can be steered vertically and horizontally in space as well as shaped to the geometries of the targeted audience areas. Thus, they project sound exclusively to where it is desired, while avoiding unwanted reflections and audible echoes.

based on that, an off-the-shelf driver and amplifier selection for a very first acoustic prototype had been made.

"Our initial development team was small, consisting only of one electrical engineer, one mechanical engineer, two software developers, and one DSP developer. Our mission was to build an acoustic prototype of X1 in the space of only four months so we could validate the viability of a design proposal for a certain mega project we were approached by. We delivered on their brief thanks to X1's unique and unparalleled audio capabilities and are immensely proud to have been chosen as the exclusive audio supplier for the project.

"After presenting that very first acoustic prototype we gained the confidence of the project's executive and continued the development of X1, to build the highest performance concert-grade loudspeaker ever created.

"One of the main challenges in designing X1 was taking very high-level project requirements and breaking them down into a product design and system concept that not only sounds and works great, but is also mass manufacturable within a certain cost envelope. Achieving that within the timeframe set by the project's construction timeline meant we were limited in certain aspects of what we could do, which was insanely challenging. To be able to do this, we grew the development team from initially five people to more than 40 in just a year's time, a mammoth challenge."

Apart from creating an entirely new category of sound system, rapidly growing a capable team of experts to execute the goal represents another great achievement for Holoplot, and shows how impactful and convincing its mission and technical concept was from very early on.

Taking It Further

With the development of X1, Holoplot broke the misconception that directed sound products could not be "loud." Holoplot can go big, go loud, while providing uncanny control over sound propagation, and still offer exceptional sound quality.

The main principle of any array technology is wave interference. Sound waves emitted by multiple sources interact with each other. Constructive interference occurs when the individual sound waves reinforce each other—building a wave of greater amplitude. Destructive interference occurs when the waves oppose each other. For waves that are neither fully in-phase nor out-of-phase, partial reinforcement or cancellation occurs.

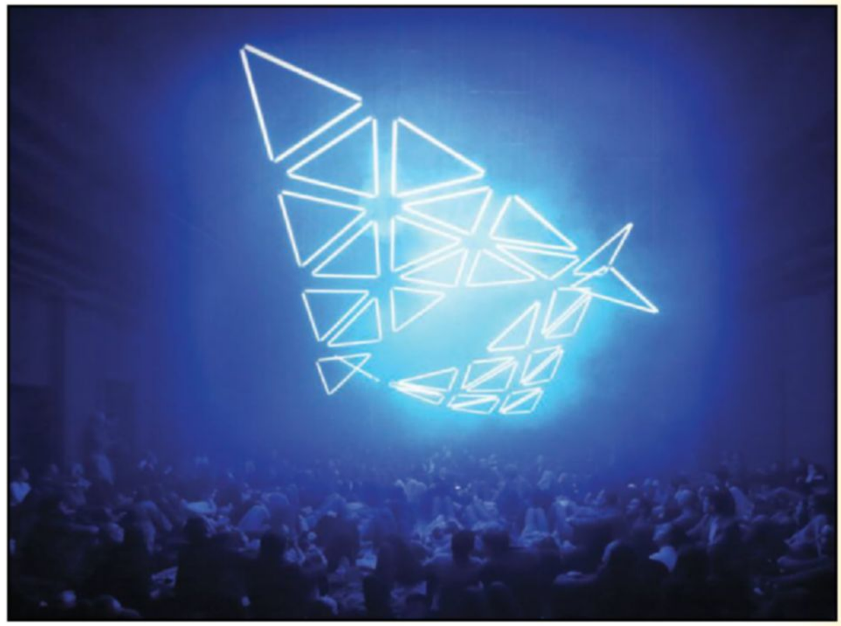
"By combining multiple loudspeaker drivers into an array, sound can be reinforced in a specific direction, while being attenuated in other directions. Similar to frequency filters boosting or attenuating certain parts of a signal spectrally, beamforming allows us to 'filter' sound spatially. The conventional way of directing sound is by mechanically aiming and curving the loudspeaker array. Obviously, this doesn't provide much flexibility. If the sound coverage of the audience needs to be changed, the mechanical array configuration needs to be modified," details Evert Start, Fellow Research Engineer at Holoplot.

In addition to this, in many venues the architectural integration of curved and tilted arrays is quite challenging. Using 3D Audio Beamforming, these problems can be overcome. "By processing the signal to each driver individually, the radiation pattern of an array can be 'molded' into any desired shape, providing more flexibility regarding system integration—which has clear advantages for both indoor and outdoor applications."

"The biggest challenges we face when developing a beamsteered product are the array size and driver spacing. At low frequencies we need a large array to obtain sufficient directivity control. At high frequencies a small array will do, but a very dense driver distribution is required in order to avoid spatial aliasing. Meeting all of these physical requirements is practically very challenging," Evert Start adds.

"In the past, several attempts have been made to fulfill these criteria. Depending on the chosen approach, either a mechanically curved line array was used consisting of sparsely spaced, highly directional wave guides, or a column loudspeaker consisting of densely spaced, wide angle drivers. The former approach is quite inflexible due to the mechanically set array curvature and doesn't provide the option of multiple beams aimed in different directions, possibly playing different audio content. The latter can provide this flexibility but is limited with respect to the maximum SPL due to low array efficiency at high frequencies."

"Building on from the principles of Wave Field Synthesis and optimized Beamforming, a new and unique technology arose, which can be best described as "Inverse Wavefront Shaping." Applying very efficient, ultra-high-channel, numerical array optimization, any sound field or beam can be created, originating from a virtual point, line or optimally curved, planar source. This ensures optimum coverage, high array efficiency, and spectral consistency across large audience areas of any shape and size, or, conversely, allows tight focusing of audio content onto small groups of people, even down to individual audience members. In this way, a single Matrix Array can generate

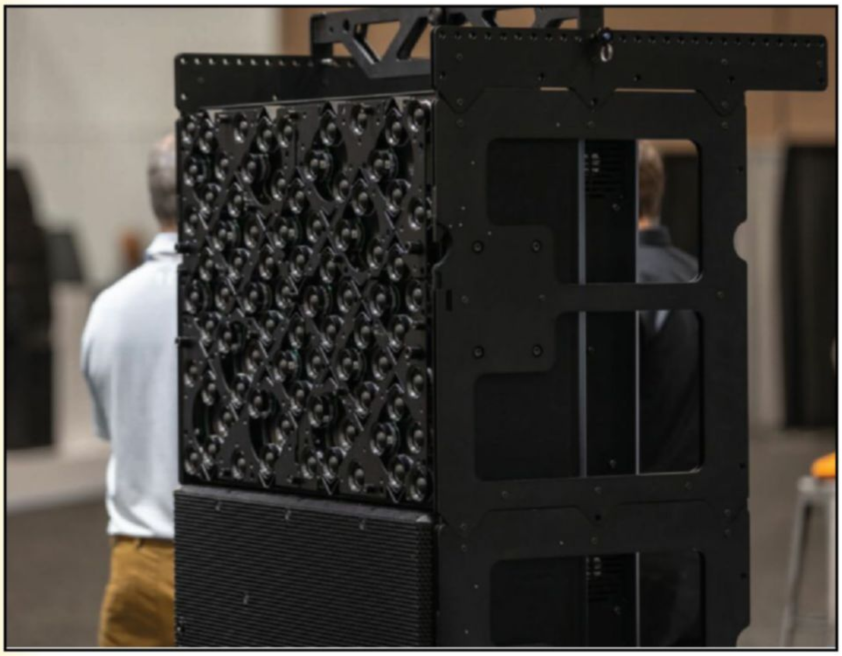


The Dark Matter installation in Berlin, Germany, is a set of interactive and immersive installations. The image shows the GRID kinetic installation, defined as "a tangible embodiment of a computer generated dynamic surface." A massive spatial sound system from Holoplot spans the entire room. Every sound and musical element in Robert Henke's composition is reproduced in direct relationship to the animation of the sculpture.

Demonstration of the Holoplot X1 Matrix Array in Barcelona, during ISE 2022

About the Author

Michael Kastner is the Head of Product at Holoplot. Since joining in 2019, Michael has been instrumental in bringing the X1 Matrix Array to life and into the market. Before this, he held product roles at Bragi, the creator of the world's first truly wireless and smart headphones. He later led the commercialization and development of Bragi's next-generation hearable platform, laying the groundwork for its software-driven B2B business.



Holoplot introduced a complete rigging solution for the X1 Matrix Array system at InfoComm 2022 in Las Vegas.

multiple sound fields simultaneously, each with its own content."

X1 is software-driven hardware. Both elements are intrinsically linked and interdependent. The hardware requirements are high and Holoplot believes that only few others in the industry are producing products with similar standards in performance and quality.

It could probably be argued that Holoplot's software is the major innovation driver now, and our efforts are currently focused on creating better sound through better signal processing algorithms, and better workflows. To that end, we are constantly expanding our software engineering team under Head of Software Engineering Bram de Jong, with a focus on world-class talent in signal processing, distributed systems, and modern C++.

High-performance entertainment is where Holoplot operates right now, and the best examples of that can currently be experienced at the Dark Matter light and sound exhibition in Berlin, or at Illuminarium Atlanta and Las Vegas—immersive entertainment experiences that guide visitors via a multi-sensory journey through the heart of an African Safari, space exploration or deep dive into seemingly out of this world kinetic sound and light sculptures.

Holoplot is now world leading in terms of spatial and immersive capabilities, such as enabling true lifelike reproduction of content via virtual sources and three dimensional placement of audio objects, but its technology is certainly not limited to the "immersive" space.

X1's 3D Audio Beamforming and zoning capabilities not only make it a creative enabler but also a high quality audio product, able to solve complex acoustic problems.

The X1 Matrix Array can create precise sound fields or zones that contain the sound where it is needed without exciting the room unnecessarily. The result is clear, uncompromised audio for speech, music or other content that renders the audio system acoustically invisible.

Holoplot wants all sectors of the sound world to benefit from the technological leaps it made and is working on a harmonized, integrated, and seamless workflow for system designers, content creators, and integrators that will make it easier to pick Holoplot as their preferred choice for a wide range of use cases. To that end, X1 is just the start.

The software toolkit, together with the X1 rigging system, which Holoplot just demonstrated for the first time at InfoComm 2022 in Las Vegas, NV, will serve as accelerators for the company's go-to-market, further empowering users to realize their creative visions—on any given scale, large or small.

Motion Feedback Desktop Subwoofers for Nearfield Studio Monitors

By
Zami Schwartzman

Photo 1: My ZRS-1 desktop subwoofers with Neumann KH 120 monitors on top.

This article describes the design of a pair of desktop subwoofers that can extend the response of compact studio monitors to the most desired 30Hz to 60Hz range. The author perfected his own accelerometer-based motion feedback (MFB) designed, fully assembled plate amplifiers with MFB control, crossover, and voice coil thermal protection that can be matched with any low-frequency driver.

When I started my own engineering business here in Israel, my initial plan was to serve the electronics industry with my thermal design and RF skills that I gained during my 26 years working on transmitter designs for Motorola. That is why I called my company "Thermoguide." During the last approximately 16 years, I have been extremely busy working on engineering projects for small high-tech firms and one of my major customers was Waves Audio, also based here in Israel. I did lots of design and analysis work for them and I still do.

I first started my audiophile journey 25 years ago. I had a friend who was buying fancy high-end gear and was never satisfied with what he got. I liked high-end music, but I decided that building my own gear based on DIY electrostatic panels was the way to go. I built five different models, until I started to search for a subwoofer that could match my electrostatic speakers.

The problem was that I needed a 400Hz crossover to match my 100cm × 50cm electrostatic panels. Commercial subs did not go that high so I tried several types of subwoofer concepts, including a couple of floor-to-ceiling transmission lines but they didn't blended well with my panels. Eventually I decided that motion feedback (MFB) was the right way to go.

I designed a unique capacitive position sensor to feed my feedback loop and in those years—when the Internet was still young—I did actually think that MFB was another of my inventions.

I also had to solve some challenging issues of loop stability and noise but eventually it all worked fine, and my subwoofer project blended very well with the panels. I was very pleased with the results and I still have this setup working fine. I kept my MFB and sensor research going through all these years with a few types of position sensor concepts that were all too complicated or too noisy.

Home Studio Opportunity

In 2018, the idea for desktop subwoofers crossed my mind. During my work with Waves, I met quite a few sound engineers in small and home studios. They all loved their small monitors—claiming they were "extensions of their ears" and that they would never consider using other monitors. They did their mixing jobs without listening to really accurate base notes through their speakers. They all tried subwoofers at some point and rejected them because they did not blend well with their monitors and excited nasty room modes.

So I thought that if I could make subwoofers small enough to be placed under the monitors, that would be a good start. And if I could design the subs wideband enough, then I could get a perfect acoustic wave front blend at the listener's ear height. By using a crossover higher than 100Hz, I could get rid of bass-midrange intermodulation in the monitors and drive them to higher SPL. And by using MFB, I could get wideband flat response down to 25Hz with small sealed enclosures. I could also get more benefits, such as fast transient decay, mitigation of driver parameter aging, and thermal compression. And with nearfield listening to the subs, the direct wave would dominate, mitigating room reflection effects.

So I headed out on a more than two-year journey of designing and building several MFB subwoofer prototypes. This ended with my current version of desktop subwoofers, the ZRS-1 motion feedback desktop subwoofers shown in **Photo 1**, which I am now showing on my website and that I'll explain.

Listening tests with professional sound engineers left them speechless, claiming they had not had such a listening experience. They also told me that home studios could be a huge market for such a product.

One of the most regarded sound engineers here in Israel (www.yuvalsound.com) had a session with these subs and the Neumann KH 120 monitors that he uses in his studio. He was so impressed by the sound quality enhancement that he keeps this setup for demonstrations in his studio, claiming that this concept is a paradigm shift. He helps me with his listening skills, contacts, and knowledge free of charge.

All that made me think that the desktop sub idea is worth commercializing. I do not have the financial means to invest in a manufacturing operation and it doesn't make sense to me, with all the supply chain issues these days. My goal is to get into a technology licensing agreement with speaker manufacturers. Although I have no patent protection, I have quite a bit of knowledge, hands-on experience, and knowledge of the hardware needed for successful MFB implementation.

Other than the desktop subs I am also focusing on subs for automotive sound systems and for soundbars. There are not many audio engineers that know their way with MFB. It is an almost forgotten technology that I brought back based on current driver materials, MEMS accelerometers, and Class-D amplifiers that are the key MFB enablers of my design.

All that technology was not available in the 1970s and 1980s when MFB was a hot research topic. Any manufacturers would have to invest years with some talented engineers working to get to where my prototypes are today.

Bottom Line

I have no secrets or protected IP other than my know-how and the specific details of my circuit design. I can bring any speaker designer up to speed with any MFB subwoofer application using any driver and box size based on my power and control circuits. It is all hardware, no software or DSP—just old fashioned electronic circuits. But it works! Adding DSP features with room correction, smart crossover and Bluetooth (BT) connectivity, the design could also fit consumer products.

The Concept

Small studio monitor speakers are widely used at home and in professional studios. Most models are based on a two-way design with a 3" to 5" midbass driver in a ported enclosure and a tweeter. The midbass driver design parameters are selected to follow Thiele-Small or other design formulas and for a given enclosure size, excursion level is limited, restricting low-frequency production. When the maximum excursion level is exceeded, driver nonlinearity causes midrange modulation by the bass notes, compromising stereo image and clarity.

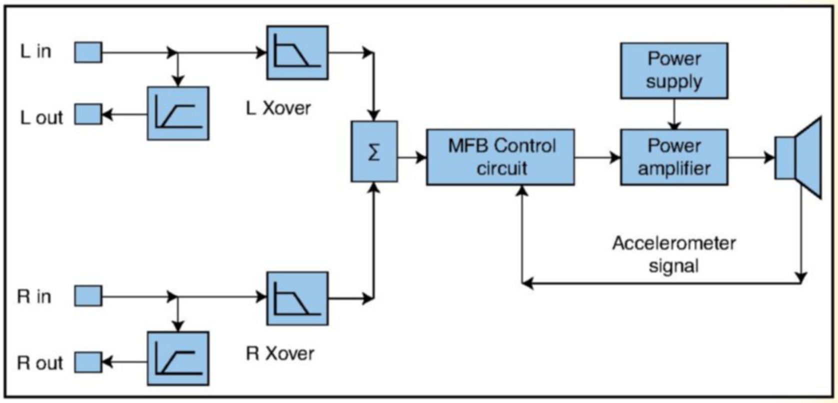


Figure 1: This is the block diagram for the ZRS-1 desktop subwoofers.

Therefore, many small nearfield monitors roll off sharply and filter out the frequency range between 30Hz and 60Hz.

Even with careful design, a ported enclosure is resonant, yielding sustained, over hanged low frequency notes (resulting in boomy or boxy sound). Adding a floor-standing subwoofer to complement compact studio monitors with a low-end frequency range causes room mode excitation. Sometimes, even with careful placement, subwoofers do not blend well with studio monitors.

Small desktop subwoofers that can cover the 30Hz to 400Hz range can be used with a crossover at 100Hz or even up to 200Hz, thus relieving the monitors to produce only the midrange and high-frequency notes, basically forming a three-way configuration.

Perfect acoustic wave front alignment can be achieved at a listener's ear height when the monitor speakers are placed on top of desktop subwoofers. And with nearfield subwoofers aimed directly at the listener, room reflection effects can be significantly tolerated.

A 20cm × 20cm × 20cm enclosure with a 6" long throw woofer seemed to be a practical design goal for a desktop subwoofer that can fit many types of studio monitors on top of them. Designing such a small subwoofer for powerful bass production is a significant challenge that can be met implementing motion control to mitigate the inherent resonances of such a small enclosure.

Overview of Commercially Available Subwoofers

Sealed or ported enclosures are the classic design concepts commonly used for subwoofers. Driver parameters have to be carefully selected for a given enclosure size following Thiele-Small (T-S) parameters or some Computer Aided Design (CAD) rules. The frequency response of a ported enclosure is resonant (a port is a Helmholtz resonator), yielding slow decay, sustained, over hanged bass notes (boomy, boxy bass).

Sealed enclosures are known to have better bass control than ported enclosures but larger enclosures and carefully selected drivers are required to get the bottom octave covered with acceptable transient response.

Large enclosures also require carefully designed bracing to mitigate panel vibration and driver parameters have to be well controlled in production and designed for long-term stability.

Thermal compression is another well-known issue with subwoofers. The voice coil temperature increases during regular operation and when it reaches 125°C, the copper winding resistance can increase to twice as much, reducing the power drawn from the amplifier by up to 6dB, thus compressing SPL and degrading transient response. Many commercial subwoofers get damaged with voice coil structural failures due to adhesive overheating, therefore, expensive over-rated drivers are often used. All these issues can be mitigated by using MFB control.

MFB for Bass Reproduction

Speakers using MFB have some sort of a sensor monitoring the voice coil's movement. The sensor's signal is fed back to the amplifier's input through a control circuit and compared with the input signal. The error signal forces the voice movement to produce sound pressure that follows the input signal.

MFB technology for low-frequency speakers is not something new. For many years only a few manufacturers implemented this technology because of its design complexity and driver and amplifier limitations. But these are no longer considered an issue with current electronics and driver materials.

With MFB, driver specifications are no longer critical to satisfy T-S or other design rules because the sensor and control circuit take care of frequency response and resonance suppression.

Long throw voice coils with reduced force factor can be used for MFB with increased linearity and higher SPL without compromising sound quality. Frequency response can be made flat from 20Hz and up to the driver's break-up mode, thus the crossover to the monitors can be set at 100Hz and up to 200Hz.

Also, relatively small enclosures can be used with MFB without compromising SPL and sound quality. With small enclosures, panel vibration is not an issue and bracing is not required. Thermal

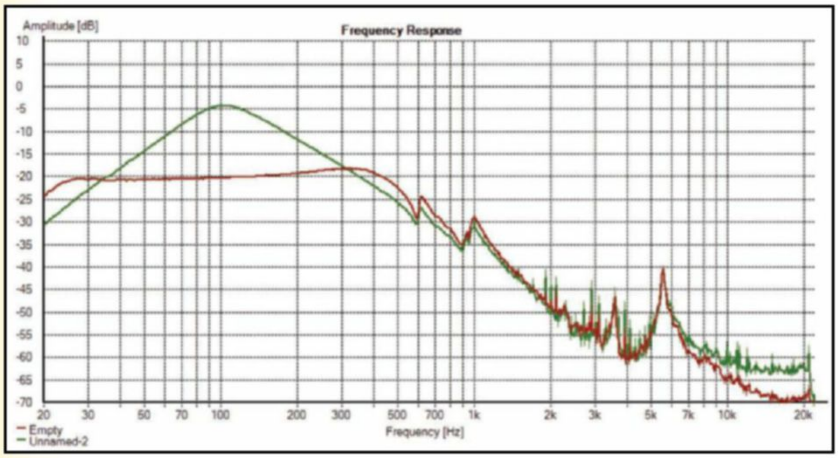


Figure 2: Accelerometer output frequency response with and without MFB, measured on a 6" driver in 20cm × 20cm × 20cm sealed enclosure.

compression and harmonic distortion are reduced significantly with MFB due to the global negative feedback.

During the last 40 years, many methods were investigated for real-time sensing of voice coil dynamics. Phillips published a research paper in 1968 and manufactured several MFB bookshelf speaker models. Infinity, Velodyne, Sony, Yamaha, Grimm Audio, and other speaker manufacturers also used MFB successfully.

Optical, capacitive, inductive, Hall effect, sensor coil, accelerometer, ultrasonics and smart amplifiers were extensively experimented for voice coil motion control methods with limited success.

Many MFB-related patents were issued by private designers and by leading speaker manufacturers. Most of the MFB-related patents were abandoned, terminated, or expired after 20 years from their patent grant date and now can be used freely by the industry.

Current MEMS accelerometers technology, robust driver materials and availability of efficient, high power Class-D amplifiers, have changed the perception about MFB and enabled the design of affordable small subwoofers.

Proof-of-Concept Prototype Design

To implement my ideas and demonstrate the feasibility of a small nearfield desktop subwoofer, I chose to work with an accelerometer-based MFB. I designed an MFB control circuit, fourth-order Linkwitz-Riley crossover filters, switching

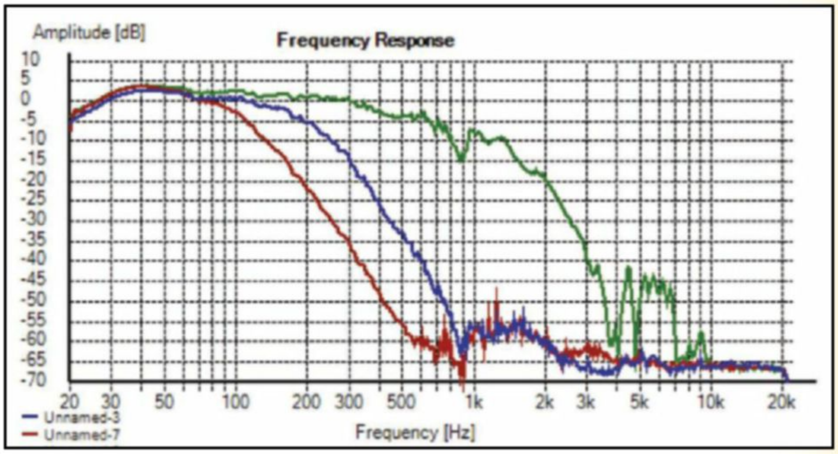


Figure 3: I measured this 6″ driver in a 20cm × 20cm × 20cm sealed enclosure measured with a very close microphone on the subwoofer's acoustic axis. Crossover filter: Bypassed (green), 100Hz (red) and 200Hz (blue).

SPEAKER FOCUS

Photo 2: These are prototypes of my ZRS-1 motion feedback desktop subwoofers.

power supply and a powerful Class-D amplifier, which I integrated into a 20cm × 20cm × 20cm MDF enclosure. I also modified a 6″ long throw driver with the accelerometer bonded on the driver's voice coil former.

Based on driver measurements and control theory, I decided to set MFB loop parameters for flat frequency response between 20Hz to 400Hz and I implemented a real-time voice coil temperature protection circuit, to keep the voice coil temperature below 125°C during overstressed operation. I also designed an overload monitoring circuit to sense the amplifier's output voltage and switch on a red light when reaching close to the power supply rails.

A block diagram (**Figure 1**) shows the basic features of this prototype design. The subwoofer was designed to be used in a full-stereo configuration, but it can also be configured as a single subwoofer feeding two monitors with left and right low-frequencies summed.

ZRS-1 Concept Test Results

Six prototypes were built. All these prototypes yielded flat frequency response from 30Hz to 400Hz without any adjustments. Listening tests performed in untreated rooms showed that the subwoofer prototypes blended well with several brands of nearfield studio monitors, revealing the best of their midrange-high frequency performance with high SPL, superb clarity, and stereo image. Well-controlled powerful bass notes were experienced down to 30Hz.

Speaker acoustics theory teaches us that voice coil acceleration is closely proportional to the acoustic output of a woofer in a sealed enclosure (when in piston mode at low frequencies below breakup). With that in mind and having an accelerometer already mounted on the voice coil, measuring the accelerometer output was the easiest alternative for minimizing room reflection measurement errors.

The measurements (**Figure 2**) show the prototype's accelerometer output with and without MFB. The first plot (green) shows the accelerometer signal measured with the feedback loop disabled. The second plot (red) shows the accelerometer signal with the MFB loop active. It is clear that without MFB a driver has a sharp resonant response. With MFB the response is flattened to 25Hz to 400Hz within ±1dB.

It is also known that at these low frequencies, very close microphone measurements on the driver's acoustic axis are proportional to 1 meter distance measurements in an anechoic chamber or in free field. **Figure 3** shows the 6″ driver measured on the subwoofer's acoustic axis with crossover filter bypassed (green), configured for 100Hz (red), and finally 200Hz (blue).

Conclusion

I demonstrated the feasibility of my ZRS-1 MFB small desktop subwoofers (**Photo 2**) that can be used to complement small studio monitors. Listening sessions performed with nearfield monitors placed on top of the desktop subwoofers demonstrated the mitigation of room reflection effects, enhanced SPL with superb clarity and stereo image. This MFB approach allows implementing subwoofers in compact enclosures, including for soundbars, car audio systems, and portable music devices.

I am open to cooperation through licensing, or any kind of business engagement with speaker manufacturers.

About the Author

Zami Schwartzman is an electrical engineer, who graduated in 1974 from the Technion in Israel. He worked for Motorola communications for 26 years, doing engineering and management of radio communication products development. He left Motorola 16 years ago and started his own engineering business—Thermoguide, Ltd. Most of his business focuses on analog electronics, thermal design, and professional audio engineering. During these years he was deeply involved as a subcontractor with Waves Audio, Ltd., a major software tools provider for the music industry. The desktop subwoofers described in this article were developed and engineered during COVID-19 as a complementary product that can be added to any nearfield studio monitor. Zami can be reached through his website, www.servobass.com or via email at zamisch1@gmail.com.

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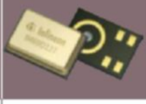
	IM69D127 (NEW) 	IM72D128 (New) 	IM73A135 (New) 	IM73D122 (Coming soon)	IM69D128S (Coming soon)	IM70D122 (Coming soon)	IM68A130 (Coming soon)	IM70A135 (Coming soon)
SNR	69 dB (A)	72 dB(A)	73 dB(A)	73 dB(A)	69 dB(A)	70 dB(A)	68 dB (A)	70 dB(A)
AOP	127 dBSPL	128 dBSPL	135 dBSPL	122 dBSPL	128 dBSPL	122 dBSPL	130 dBSPL	135 dBSPL
Sensitivity	-34 dBFS	-36 dBFS	-38 dBV	-26 dBFS	-37 dBFS	-26 dBFS	-38 dBV	-38 dBV
IDD	980 µA	980 µA	170 µA	980 µA	420 µA	980 µA	110 µA	170 µA

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Replacing IEC 60268-5 (Part 1)

What Comes Next and Why

By
Geoff Hill

This article discusses the work taking place on loudspeaker measurement standards, and the progress the International Electrotechnical Commission (IEC) has made with its standards efforts.

Loudspeaker standards are and have been central to my working life, though for many years I—and probably many people—did not realize why they were and are so important. To explain, I need to go back many years to when I first got into loudspeakers—in my case initially through electronics. I first came across loudspeaker standards in my work at Impulse, Bowers & Wilkins, and Goodmans. The key requirements almost inevitably came from customers, and this was especially so when working in programming the test equipment used by our automotive customers.

As a general rule when you are making one-offs, precision and repeatability is not or should I say was not as highly regarded then as it is now. And that is the link between standards and calibration. In order to calibrate something, you have to have adequately defined Standards, so when you measure it, you can refine the measurements defining measurement uncertainties.

Calibration Systems—Links to Metrology

Arguably some of the first defining characteristics of civilization are those of (1) Record keeping and (2) Measurement. It's difficult to imagine one without the other so in all probability they both emerged together.

When you have a measurement, you need at some point to deal with uncertainty. As a simple example, it is easy to count 10 coins, but what about counting grains of rice? You can count 10 individual grains but what about the number of grains of rice in a bag? A barn or a silo of grain? Taking another physical example, a length of a beam? The distance between aircraft in the sky? Inherently there will be some uncertainty of these (if only because they are moving) but how much and how can we define it so that we can have confidence in our measurements?

Even within "simple" measurements uncertainty can quickly overcome/exceed the measurement tolerance for many reasons, often these come down to the measurement environment. The science of this is called Metrology and was defined by the International Bureau of Weights and Measures (BIPM, 2004) as "the science of measurement, embracing both experimental and theoretical determinations at any level of uncertainty in any field of science and technology." I believe the key words here are "Determination of Uncertainty." We don't know and can't know a measurement value absolutely only within a tolerance of that value.

And how do we make reliable, consistent, and accurate measurements? The obvious thing is first to determine why a measurement may suffer from variation(s) and then to strive to eliminate as many of these cause(s) as possible. We then need to consider how accurate we need to be. This largely depends upon the individual usage of a measurement. With loudspeakers, it probably does not matter much if the sensitivity of a speaker is off by 0.1dB, however if we were calculating a trajectory for a spacecraft, we would need a much higher degree of certainty in our measurements.

So how do we guarantee any measurement? The first thing to say is that there is no such thing as an absolutely accurate measurement—only one that is known to a degree of uncertainty. The primary standards are known to a higher degree of accuracy and have a smaller range of uncertainty. A "Hierarchy of Standards" is defined as:

- Primary standards
- Secondary reference standards
- Working standards
- Laboratory standards

Primary standards are held in only a very few selected locations and organizations throughout the world—these are the internationally agreed standards and are held by the International Bureau of Weights and Measures, which is an intergovernmental organization through which its 59 member-states act together on measurement standards. An example was the standard meter, an example of which is shown in **Photo 1**. This is now defined as 1/299 792 458 of the speed of light...

A continuing trend in metrology is to eliminate as many as possible of the artifact standards and instead define practical units of measure in terms of fundamental physical constants, as demonstrated by standardized techniques. One advantage of elimination of an artifact standard is that inter-comparison of artifacts is no longer required. Another advantage is that the loss or damage of an artifact standard will not disrupt the system of measures. Secondary reference standards are held by individual countries primary calibration organizations (e.g., NPL in the UK and NIST in the US).

Finally, working standards are held by calibration labs usually commercial or academic institutions that reference the national standards organization to calibrate their standards. Individual instrument companies thus refer for their calibration upon these working calibration standards. Thus in theory and practice an unbroken calibration chain can be realized, the point to note that at every stage there is a measurement uncertainty and one of the prime aims of metrology is to determine this, and of course we want to minimize any such uncertainty at every stage.

Sound Standards

Sound pressure level (SPL) is the pressure level of a sound, measured in decibels (dB). It is equal to 20x the Log10 of the ratio of the Root Mean Square (RMS) of sound pressure to the reference of sound pressure (the reference sound pressure in air is 2 x 10-5N/m^2, or 0.00002Pa). The primary requirement to calculate SPL relies upon N=Newtons a measure of Force, m^2,or m^2 = distance and area. This is referenced to the nominal threshold of human hearing at 1kHz taken to be 0dB SPL or 20μPa.

In high volume applications—be that automotive, mobile, telecom, or consumer products—it is essential that designs are safe and compatible between competing products, and also that the same item is being made in the same way every time. That is where standards and in particular international standards come in.

Standards and Standards Organizations, enable people from individual companies to contribute freely to the common good—so a standard, represents a balanced consensual agreement by experts in that subject and the best implementation at a given time.

A standard clearly outlines the overall results that can be expected by following a general outline of a methodology, so they, like patents—which by granting a time-limited monopoly in return for the open disclosure of an idea can gain protection—refer to earlier inventions and techniques.

A natural result of this is that standards need to evolve and they enable any discipline to continually update itself to current best practice. As an example, individual authors publish books or issue papers about Loudspeakers through journals in our field of interest, by the American Acoustical Society (ASA), Audio Engineering Society (AES), and the Institute of Acoustics. However these books and papers have not and cannot have been fully peer reviewed, while the process of publication as a standard requires a full peer review together with a public call for comment. This ensures that a standard can be relied upon to

Photo 1: At 36 rue de Vaugirard, in Paris, a *mètre étalon* (or standard meter) from the 18th century is embedded in the exterior of a building.

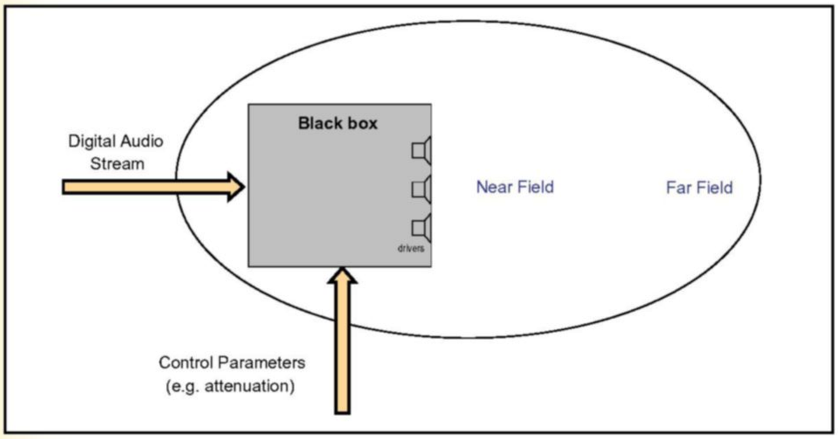


Figure 1: Acoustical output based measurements are shown as defined in IEC 60268-21.

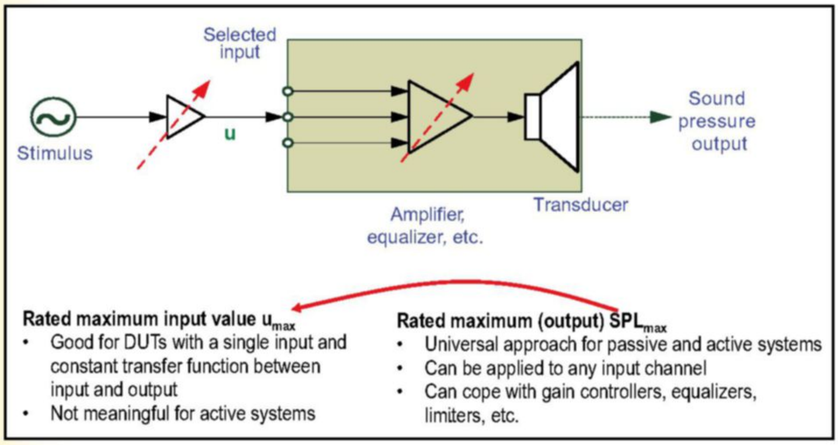


Figure 2: This graph shows the maximum input and output signal. The rated maximum input value u_{max} is good for DUTs with a single input and a constant transfer function between input and output. And, the rated Maximum (output) SPL_{max} is the universal approach for passive and active systems, can be applied to any input channel, and can cope with gain controllers, equalizers, limiters and more.

About the Author

Geoff Hill, inventor of the Tetrahedral Test Chamber (TTC) and CTO at Hill Acoustics, has been working in the loudspeaker and audio industry for more than 40 years. He is Chair of the AES SC-04-03 Working Group on loudspeaker modeling and measurement and a member on the IEC Working Group TC 100/TA 20/PT 60268-22. He is also the author of *Loudspeaker Modelling: A Practical Introduction*. Geoff welcomes your comments and can be contacted via email geoff@hillacoustics.com.

Recources

W. Klippel, "Electrical and Mechanical Measurements of Loudspeakers and Sound System Equipment: Tutorial to a New IEC Standard," 2016, www.klippel.de.

"Meter," National Institute of Standards and Technology, updated June 2021, www.nist.gov/si-redefinition/meter

"What is Metrology," NCSL International, https://ncsli.org/page/WIM

achieve what it sets out to do and importantly that the underlying science is correct.

I first came to Loudspeaker Standards around 1980, via IEC60268-5 (which was first issued in 1972). By 2003, the Total Q-factor (Qt) Thiele-Small parameters and the compliance of a loudspeaker drive unit had been defined along with the basic measurement setup of an anechoic chamber, the IEC baffle, and various test boxes, together with harmonic distortion, noise signals, and so forth.

The Abstract of the first edition summarizes it as: "Applies to sound system loudspeakers, treated as entirely passive elements. Gives the characteristics to be specified and the relevant methods of measurement for loudspeakers using sinusoidal or specified noise signals."

To better understand it, let us briefly look at the contents of IEC60268-5:2003, which is the last major incarnation.

We had just four signals defined: Sine, Broadband and Narrow band noise, and Impulsive. The measurement environment was characterized as free-field and half space, diffuse, and simulated free-field, and the document specifies standard baffle or enclosures, impedance measurement, total Q-factor (Qt), Vas, rated voltage (Power), frequency range, and sensitivity, radiation angle and directivity index, amplitude nonlinearity, THD, and stray magnetic fields.

However, the loudspeaker theory and practice in 2021 is much more advanced compared to the 2000—let alone the 1980s! And a major update was arguably overdue to make those standards applicable to all kinds of modern audio devices whether active or passive.

There's also the need to cope with any input signal, analog, digital, wireless, or something new to include new measurement techniques (e.g., Rub & Buzz test); and to bring together manufacturing (QC) and system development (R&D).

Updates were also required to provide comprehensive information in a shorter measurement time, simplify interpretation (e.g., root cause analysis), and to increase flexibility to consider particularities of the applications (e.g., home, automotive, personal, and professional).

These new considerations are approached in the updated standards:

- IEC 60268-21: Output Based Acoustical Measurements
- IEC 60268-22: Electrical and Mechanical Measurements on Transducers
- IEC 63034: Micro-speakers

These multiple standards are necessary as increased use of loudspeakers in many areas show a single standard is no longer enough.

Acoustical (Output Based) Measurements

The scope of the IEC 60268-21:2018 International Standard applies to passive and active sound systems, such as loudspeakers, headphones, TV sets, multimedia devices, personal portable audio devices, automotive sound systems, and professional equipment. It applies to those loudspeakers that are best described as "Black Boxes" in that they accept an input—analog, digital, radio, or something else—and then produce an acoustic output. It is this acoustic output which is the subject of the standard.

The evaluation of the Output is of course dependent upon the Black Box (**Figure 1**).

Where A Black Box is used as description of a system whose internal workings are (1) Unknown or (2) the internal status at any time is unknown, all that is known are the inputs and the overall output. As in previous standards, measurements can be made in the near field or the far field.

The outputs of concern are generally those that a user will directly notice: amplitude (SPL) versus frequency (Hz) and distortion (% or dB), together with maximum level.

Nonlinear Distortion Measurements

1. Harmonic Distortion (single-tone stimulus) *Old*
   - Total harmonic distortion
   - Nth-order harmonic distortion component
   - Maximum SPL for defined THD limit *NEW*
   - Equivalent harmonic input distortion
   - Higher-order Distortion
2. Intermodulation Distortion (two-tone stimulus)
   - 2nd and 3rd-order intermodulation component
   - Amplitude modulation distortion
3. Multi-tone Distortion (multi-tone stimulus)
4. Impulsive distortion (chirp stimulus)
   - Impulsive distortion level
   - Maximum impulsive distortion ratio
   - Mean impulsive distortion level
   - Crest factor of impulsive distortion

Figure 3: The nonlinear distortion measurement standard has added several more layers to the definition of harmonic distortion and intermodulation distortion as well as adding multi-tone distortion and impulse distortion to the measurement options.

The key is that specifications and measurements made under this standard are relevant user output-based ones. The users have no knowledge of the internal workings of the system. The specifications therefore are purely input to output based, or ones that rely only upon measurable acoustic outputs such as:

- SPL vs. Frequency
- THD vs. Frequency
- Output level vs. Distortion

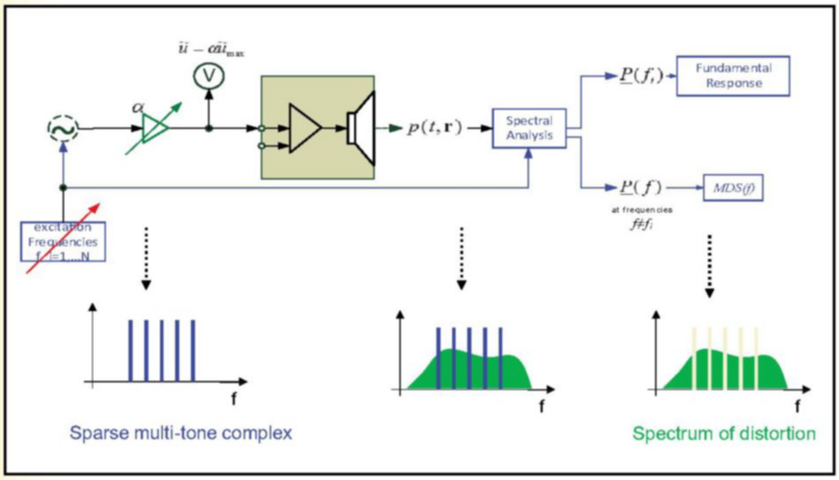


Figure 4: The problem with the measurement of multitone distortion is that the result depends on the excitation lines selected, requiring a standard for multitone stimulus.

So, IEC 60268-21:2018 is primarily focused on output-based measurements. It updates the measurement techniques to include stimuli (chirp, multi-tone complex, and burst), includes comprehensive physical evaluation of the acoustical output, and assesses large signal performance (considering heating and nonlinearities). Umax or SPLmax are rated by the manufacturer to calibrate the Real Mean Square (RMS) value of the stimuli.

The updated standard also extends acoustical measurements of loudspeaker beyond the IEC baffle to include relative and end-of-line test chambers based upon the tetrahedral test method, and considers a complete assessment of the 3D sound field radiated by the loudspeaker in an anechoic environment (near and far field).

It also describes physical measurement of higher-order harmonics and impulsive distortion in the time domain to assess Rub & Buzz and other loudspeaker defects. It effectively forms a bridge between manufacturing (QC) and research and development (R&D) measurement procedures.

Maximum Input and Output Signal

These measurements are required to allow effective measurements of both conventional passive and more commonly of active loudspeaker systems (**Figure 2**). However, to make such measurements we need to firstly define and calibrate our input signals. The updated standard considers new sinusoidal chirp, steady-state two-tone signal, sparse multi-tone complex, and Hann-burst as input signals for that effect.

Such signals can be applied through the appropriate methodology, be that analog, digital, or any other suitable means compatible with the device under test, which give us the possibilities of applying new measurements such as adjustment of input levels, compression, and delay, impulsive and multitone distortion (**Figures 3–5**).

Next Month

In Part 2 next month, we discuss how these new standards apply to measurements of sound in 3D space and in production environments, measurement of SPL using voltage sensitivity, and the measurements of microspeakers.

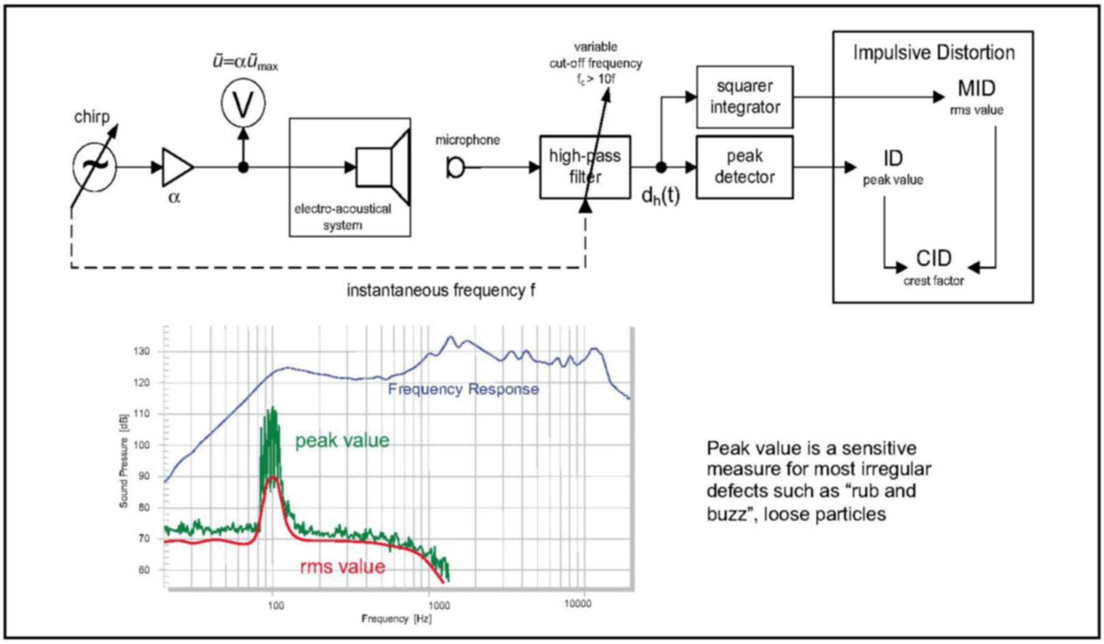


Figure 5: With the measurement of impulse distortion, a peak value is a sensitive measure for most irregular defects such as Rub & Buzz and loose particles.

Show Report

AES 2022 Automotive Audio International Conference

Immersive and Energized

By
Roger Shively
Photo credits © Spencer Shively

Over the course of three days and two nights, June 8–10, the Audio Engineering Society's 2022 International Automotive Audio Conference in Dearborn, MI, served as a great event for everyone in the automotive audio community to reconnect.

Photo 1: After the past years of isolation, the 5th AES International Conference on Automotive Audio returned to Dearborn, MI, where it was first held in 2008, offering 26 technical papers, along with workshops ad tutorials throughout.

After years of isolation and working remotely, the automotive audio community had a chance to catch up with old colleagues and discuss new ideas. As bright as an early summer day in southern Michigan, there was a great joy and a feeling of satisfaction to listen to music in a car again with someone who had created a unique system integration or the latest immersive algorithm, and then to talk about what would come next, or to experience unique testing and simulation methods.

At the end of the conference, attendees were reluctant to leave and some demos continued past the point of the exhibit hall being cleared out and readied for the next event. Everyone was already looking forward to the next conference in Europe, as if it couldn't come soon enough.

These were the sentiments of the record-setting 200-plus attendees of what is always a very popular Audio Engineering Society (AES) conference (**Photo 1**). It is a conference where OEMs, Tier Ones, and independent suppliers and contractors can gather at one event to discover possibilities and solutions in a very specialized audio community. It engaged the attendees with 26 excellent technical papers in the program, and it also instructed with workshops and tutorials throughout. In parallel with the technical program, the AES's Audio Product Education Institute (APEI) initiative provided a hands-on track, covering Voice DSP, Machine Learning, CAE Simulations, and Test and Measurement. And, for the first time, there was a free "Careers in Automotive Audio" event for any student who could make their way to the Dearborn Inn (**Photo 2**).

Photo 2: The future is in good hands. College students were invited to meet industry professionals to talk about careers in automotive audio. Face-to-face questions and answers, led by APEI leader Scott Leslie. Students were given free entrance to the exhibit floor, where the conversations continued.

Concurrent with the technical programs and workshops, the 7,600 square foot ballroom was home to the exhibit's hall for 16 sponsors with nine vehicles (**Photo 3**). There was also a showcase of the AES's Technical Committee for Automotive Audio's working group's results on "Measurement Techniques in Vehicles," which will be released as a white paper this fall (**Photo 4**).

It is no surprise that this community is so enthusiastic and the conference so popular. As one attendee was overheard to say, "The sound system in a person's car is generally the best sound system they have."

Coming together again in the largest group yet, the passion that drives this automotive audio community was on display and buzzing with excitement. From the industry veterans, newcomers, and students alike, no one could have been happier to be there and found it harder to let the time be over.

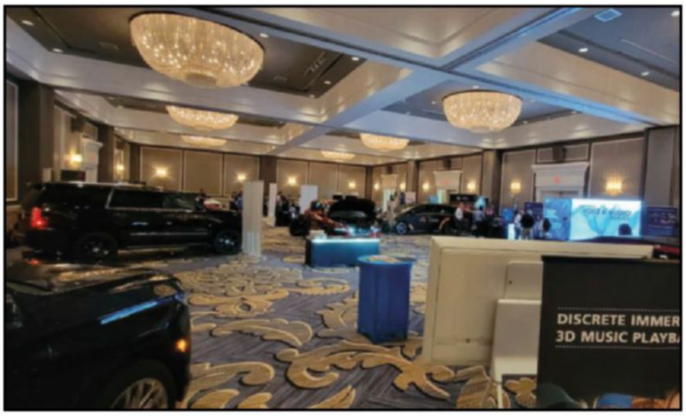


Photo 3: The 7,600ft^2 ballroom was home to the exhibit's hall for 16 sponsors with nine demo vehicles. Networking with colleagues, customers, and future partners, the excitement overflowed during coffee and lunch breaks, then into the night during dinner on Day Two.

Photo 4: The in-vehicle measurement working group for AES's Technical Committee on Automotive Audio, was demonstrating the techniques that will be published as a white paper this fall. Demonstrations were provided for the measurement of frequency response, maximum SPL, and impulsive distortion (aka buzz, squeak, and rattle) best practices. All supported by Harman, Volvo, Nissan, Bose, Stellantis; GRAS Sound & Vibration, Klippel, Listen Inc., JJR Acoustics, and Sound Equity Investments.

Just the Highlights

The conference opened with just the right keynote from Dr. Mathias Johansson, co-founder and Chief Product Officer of Dirac Research (**Photo 5**). Many automotive industry leaders talk of an on-going paradigm shift that resembles the disruptive change that the mobile phone industry underwent a decade ago. In his talk, Johansson discussed how this industry transformation has already increased the demand for high audio quality of in-car entertainment systems. He discussed the primary drivers, and what the demand and opportunities for higher audio quality means in practice.

The first day focused on test and measurement techniques for enhanced perceptual Rub & Buzz from Listen, Inc., automatic head position detection and system adjustment from Panasonic (**Photo 6**), as well as in-vehicle control of nonlinear speaker behavior, from Klippel. The day also included presentations on automated control of reverberation level using a perceptional model, and object-based audio as a platform technology in vehicles, both from Fraunhofer (**Photo 7**), as well as a new stereo up-mix design for shaping sound experiences in automotive from Dirac Research (**Photo 8**). A comprehensive tutorial on position dependent amplitude response in automotive loudspeakers, with measurements to isolate or eliminate the causes of these audible anomalies, was presented at the end of the day to a full house, by Mark Ziemba from Panasonic (**Photo 9**).

The second day was led off by Andreas Ehret (**Photo 10**), Director of Automotive at Dolby, whose keynote set the tone for the day on a trend in audio that is beginning to mature in Automotive: Immersive Audio—the topic that captivated the conference throughout. Content creation, playback technology, and consumer preferences all point toward the growing momentum of immersive audio in the consumer entertainment industry. The biggest question was: Is the automotive audio industry ready to embrace this revolution? The audio industry will face new opportunities and challenges for how to make immersive in-car entertainment technically possible across a growing number of

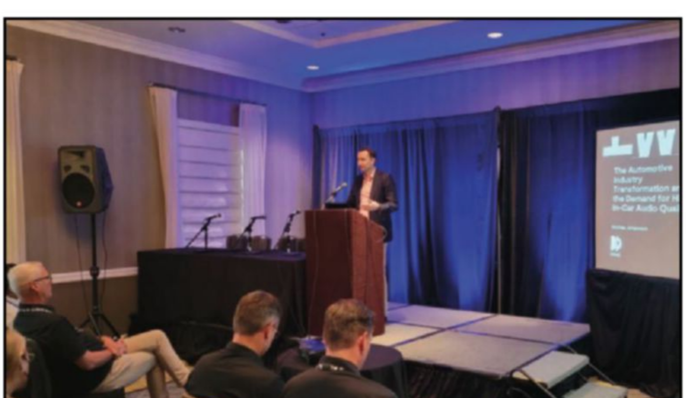


Photo 5: Co-founder and Chief Product Officer Dr. Mathias Johansson's Day One keynote, "The Automotive Industry Transformation and the Demand for High In-Car Audio Quality," focused on the on-going paradigm shift in automotive audio.

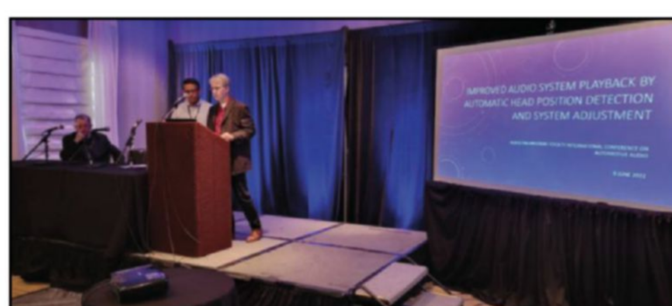


Photo 6: Jonathan Lane, from Panasonic, presented a talk on "Improved Audio System Playback by Automatic Head Position Detection and System Adjustment." Trying to achieve a single average setting for the range of listener positions has shown to result in degraded audio playback for those whose seating position puts their ears on the fringes of the normally measured. Deviation over the space is especially notable in the case of nearfield speakers (e.g. headrest and overhead speakers).

Photo 7: The Fraunhofer team was in force on the exhibit floor with demonstrations and with technical papers, presenting on an "Automated Control of Reverberation Level Using a Perceptional Model" and "Object-Based Audio as Platform Technology in Vehicles."

Photo 8: The team from Dirac Research gave more demos than time allowed, being the first in and the last out of the exhibit hall every day. They also provided a hands-on workshop in the APEI sessions on Intelligent Audio Tools, and moderated a panel on "Making Immersive Audio in Cars a Reality." This was an exhausted, happy crew.

use cases. There exists tremendous future potential for enabling these types of experiences, and by analyzing this topic, Dolby is helping to unlock what this revolution will mean for the industry and consumers.

That very topic, continued into a panel discussion that focused on "Making Immersive Audio in Cars A Reality" (**Photo 11**), with content providers from mediaHYPERIUM, an OEM (Audi), a Tier One (Panasonic), and system providers (Dirac, Dolby, and Fraunhofer). The topic had others talking about it from the outside looking in, realizing that it's no longer multichannel in a car, it's multi-positional. Thoughts came up of where the audio processing would be taking place: in an amp, a cockpit domain controller, a module added to a bi-directional network, or in the cloud, looking at vehicle metadata and delivering the content in the optimal format. Also considered were the challenges for the content providers, the content licensees, and the hardware and firmware suppliers.

This was followed by a range of topics, with talks on evaluation of sound quality from New York University and from HEAD acoustics, and on achieving maximum audio processing throughput on the latest automotive chipsets from DSP Concepts. Also presented was a study on speech intelligibility performance of automotive voice microphones from Harman, an acoustic front-end to speech recognition in a vehicle from Blackberry QNX, and on acoustic event detection, seeing with sound: detection and localization of moving road participants with AI-based audio processing from Reality AI and Infineon.

The third day focused on airborne and vehicle noise and the audio system co-development. Presentations were given on AI in

Photo 9: Mark Ziemba, from Panasonic, held everyone's attention with his tutorial on "Position Dependent Amplitude Response in Automotive Loudspeakers," demonstrating audible impact of this distortion on the accuracy of music reproduction in automotive audio systems. This performance was monitored through the sound reinforcement system with video so every conference attendee could see the loudspeaker under test and experience the sound of the distortion with music examples.

Photo 10: Day Two Keynote Speaker Andreas Ehret, Automotive Director at Dolby, watches while students Kyle Laemmle and Dannalee Mata demonstrate Apple Airpod Max Dolby Atmos to Conference Chair Roger Shively.

Photo 11: The panel on Immersive Audio with panelists Wolfram Jähn (Audi), Herbert Waltl (mediaHYPERIUM), Stefan Meltzer (Fraunhofer), Andreas Ehret (Dolby), Mark Ziemba (Panasonic), and Moderator Rüdiger Fleischer (Dirac). Automotive could be the most ideal environment to enjoy all aspects of immersive music on a day-to-day basis. And, all players across the ecosystem must collaborate to implement high-quality immersive music in cars to create the experience artists originally intended. The panelists dove into greater detail about bringing immersive audio into cars.

Photo 12: "Pop and Burble Triggered Sounds Development Process" from General Motors and Mueller BBM. Automotive customers have become accustomed to triggered sounds in vehicles for alerts and communications. The field of triggered sounds is expanding to include personalization and customer-exciting triggered sounds. These triggered sounds are broadcasts from the infotainment system and are both triggered and controlled via the vehicle CAN messages. This level of complex triggering and control allows OEMs to craft excitement generating triggered sounds while providing audible feedback which corresponds to vehicle operation.

automotive audio, approaching dynamic driving sound design from Impulse Audio Lab, active sound design where noise, vibration and harshness (NVH) meets audio from Siemens and Neosonic, active noise cancellation (ANC) simulation considering the coupling effect of secondary sources from IAV, virtual pass-by noise sound synthesis from transfer path analysis data from Siemens, and on pop and burble triggered sounds (one example of a triggered sound that can generate excitement and provide a perception of powerfulness for the vehicle) from General Motors and Mueller-BBM (**Photo 12**). And, finally, there was an excellent full panel discussion on Noise and Sound with Siemen, Mentor, Harman, and Silentium covering the interaction of ANC, NVH, Acoustic Vehicle Alerting Systems (AVAS) and the sound system in cars, and how they can be simulated and integrated together.

It could be said that change is slow to come, and then it comes all at once. The shift from two-channel and multichannel audio with simple sources to an immersive, multi-positioned audio, which is integrated with audio systems that manage and process noise and alerts, in parallel with the shift to personal sound zones, is ongoing and maturing rapidly.

The Demonstrations and the Displays

Walking through the exhibits and demonstrations, there was a wide range of types and opportunities to experience and learn (**Photo 13**). Audio Precision was on hand demonstrating best practices for getting trustworthy audio measurements in a vehicle as well as part of computer model verification in an APEI workshop (**Photo 14**). HEAD acoustics demonstrated the use of head-and-torso simulators (HATS) in recording how cars sound and how headphones and in-ears sound.

Using their MDAQS, a standardized ranking for timbre, immersiveness, and distortion of cars and earphones was possible through testing in a matter of minutes. HEAD acoustics also demonstrated the BHS II, a mobile headset for aurally accurate

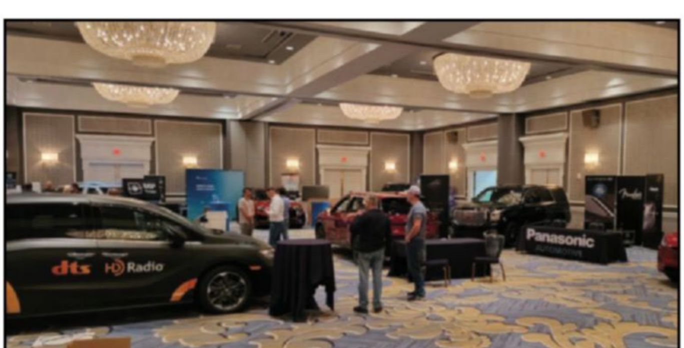


Photo 13: The showroom floor was filled with the sponsors: Analog Devices, Listen Inc., PCB Piezotronics, COMSOL, Xperi DTS, Impulse Audio, HEAD acoustics, Sennheiser, Tymphany, Dirac Research, Dolby, Panasonic, Harman, Fraunhofer, and DSP Concepts.

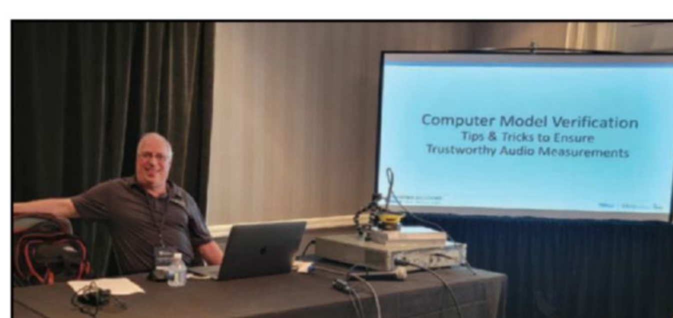


Photo 14: Verifying the accuracy of a computer model is highly contingent on getting accurate measurement results. Dan Foley of Audio Precision demonstrated tips and tricks for ensuring trustworthy audio measurements. In many instances, measurement results are often blindly accepted when in actuality the measurements are highly suspect.

Photo 15: The Acura MDX Type S utilized overhead speaker locations to create an immersive experience. The Acura had the production ELS Studio 3D Signature Edition. In addition to the rear subwoofer enclosure, Panasonic includes a pair of woofers in the center console, evenly distributing the bass, as well as providing a three-way architecture for the center speaker. The overhead speakers were well tuned and helped create a 3D ambience.

Photo 16: Similar to the Acura, Panasonic implemented the next version of its 3D immersion with overhead speakers, which were well-tuned and aligned to give the driver and passenger sense of a personal immersive experience.

sound recordings. A set of headphones with little microphones on the outside of each ear cup is used. The microphones measure 360 degrees of detail around the head and the auralization of the measurement seemed very realistic. Across the hall, Panasonic was demonstrating in an Acura MDX Type S and a GMC Denali. Both utilized overhead speaker locations to create an immersive experience (**Photo 15** and **Photo 16**).

Next door to Panasonic, Dolby brought a Tesla Model X with Dolby Atmos automotive. The Model X had additional height speakers to allow for a true Atmos experience. And, Atmos encoded material was used, which did a very good job of exemplifying the immersive experience. Movie samples even outperformed the audio tracks in a fully-realized Atmos environment. On down the line, Dirac brought a Volvo XC60 T8 Recharge with the Bowers and Wilkins sound system. They added two additional height speakers where the grab handles were in the backseats. The speakers ended up giving a very good three-dimensional experience, while demonstrating their new-mixing technology, Virtuo. Using the common source of Spotify streaming material that wasn't specifically mixed or altered, they flipped the Virtuo switch. The system felt like a spectral and spatial veil had been lifted. The fact that this technology could just be put into a car with little to no other special features makes it incredibly impressive and applicable across the industry.

Dirac's neighbor at the show was Sennheiser, which was also using a Volvo XC60 Polestar with the Bowers and Wilkins sound system, in which they added Continental shakers in the headrests and a subwoofer shaker in the rear. The demonstration was of AMBEO immersive audio technology (**Photo 17**). The near-field headrest audio worked well for getting the backing vocals subtly placed right behind the listener and also creating listening zones for each passenger. Across the hall was Harman Automotive, where they brought a production Cadillac Escalade with the 36-speaker AKG Studio Reference sound system, which

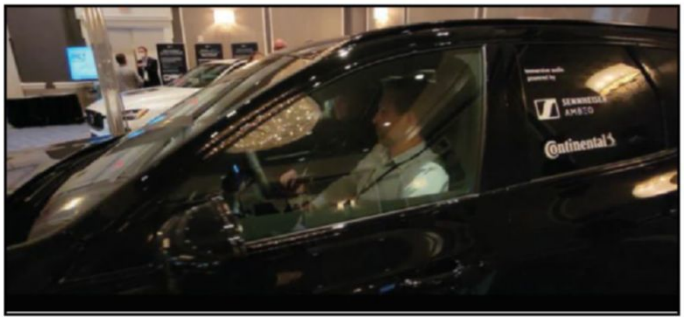


Photo 17: Sennheiser demoed AMBEO immersive audio in a Volvo XC60 Polestar with the Bowers and Wilkins sound system, to which they added Continental shakers in the headrests and a subwoofer shaker in the rear. The demonstration was of AMBEO immersive audio technology.

Photo 18: Tymphany's Automotive Audio Development Environment (AADE). Powered by DSP Concepts' Audio Weaver, the AADE is a unique SUV-sized audio development tool used for creating new acoustic solutions, layouts, and proof of concepts.

offered customizable passenger seat volume control and additional headrest mounted speakers for immersive, 360-degree sound throughout the cabin. AKG's Acoustic Lens technology in the front of the Escalade's IP speakers directed the sound smoothly toward the listener, while reducing the amount of sound reflected off of the windshield.

Fraunhofer demonstrated object-based audio as an immersive format and the use of automatic control of reverberation levels to improve the naturalness of the immersive sound formats. Next to them was Tymphany, partnering with DSP Concepts. Tymphany's Automotive Audio Development Environment (AADE) vehicle buck was on display. Powered by DSP Concepts' Audio Weaver, the AADE is a unique SUV-sized audio development tool used for creating new acoustic solutions, layouts, and proofs of concept. They demonstrated implementation examples, detailing voice recognition from multiple sources, telephony, sound zones, microphone performance, array studies, and object-based audio (OBA) (**Photo 18**).

That immersive audio technology has matured to its current levels and can be put into a car with little to no other special features makes it incredibly impressive and applicable in any vehicle. We can see that the benefits of immersive architectures, object-based audio, auralization and simulations are fully realized now in the automotive audio space, and that unique methods for controlling the soundfield and measuring will, and need to, evolve.

About the Author

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

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

Audio Objects as Platform Technology in Vehicles

By
Christoph Sladeczek
(Fraunhofer IDMT)

The availability of immersive entertainment and personalized sound staging will be essential in automotive audio. This article describes a new workflow comprising the rendering of audio objects for the production and tuning process, as well as the implementation on resource-limited hardware. As audio will play a significant role in upcoming interior concepts, object-based audio provides a novel interface to create immersive and interactive audio content significantly reducing the complexity of the audio production workflow for car manufacturers.

The trend for immersive sound reproduction in cars is obvious, as car manufacturers introduced height speakers and advanced multichannel loudspeaker setups in recent years. Augmented reality brings with it new kinds of applications and therefore new demands on in-car sound systems that are not only used for entertainment purposes but increasingly as a general communication channel between car and occupants. The use of immersive reproduction technologies will have a huge impact on the interface design of the interior of the future. However, focusing on audio to create new interior experiences comes with major challenges, as various usage scenarios are developed in different specialist departments of automotive manufacturers. In this context, the effort required to create unique audio experiences will take on a significant role, requiring a new unified interface for spatial presentation to limit production efforts. Another key element is interactivity as the concept of connected car requires intense interaction between car devices, traffic participants, infrastructure, and between vehicle and occupants.

Interior of the Future

Vehicle interiors will change more than they have in decades. As more and more of the driver's tasks are being taken over by intelligent assistance systems the focus shifts away from the road and towards the interior experience. With the new demands on comfort and safety, audio will play a significant role soon. We will discuss the important use cases for audio, which lead to new requirements on the general audio production platform.

Entertainment Audio

The development of in-car sound systems increasingly focuses on creating immersive listening experiences for occupants. However, there is a wide variety of different audio formats on the market, as well as audio content that is available in 2D or 3D. Future in-car sound systems should therefore be able to reproduce all relevant audio formats and all audio content, whether in 2D or 3D, to provide the driver with a consistently high-quality 3D listening experience.

Assisted Audio

Today's driver assistance systems often limit their feedback to visual warnings and information, but do not fully exploit the potential of the acoustic channel, even though the ear is on 24-hour reception, is more powerful than the eye and perceives information from all directions. Future driver assistance systems will therefore require an interactive, 360-degree reproduction of acoustic events that are spatially unambiguously placed in and around the vehicle, thus enabling a direct and more intuitive perception of the vehicle's environment.

Social Audio

Nowadays, telephone calls via the vehicle's hands-free system are part of everyday life. However, they often impose a high mental load on the driver, especially when communicating with several participants, as word contributions are difficult to distinguish. Particularly in complex traffic situations, this is an additional challenge for the driver. A spatial correct reproduction of multiple phone call participants allows users to trigger the so-called "cocktail party effect." This significantly reduces the driver's listening effort, freeing up cognitive resources to focus on essential driving functions.

Smart Interior Audio

Smart surfaces are increasingly conquering car interiors. Control elements only appear when the driver needs them and then disappear again. Smart surfaces must therefore be intuitive to use and provide the driver with clear feedback as to whether the function has been used successfully. As the acoustic feedback has an important function for clear and comprehensible perception, future in-car systems must therefore be able to provide spatially correct acoustic feedback for smart surfaces.

Comfort Audio

Car manufacturers increasingly focus on creating emotional driving experiences. The decisive factor here is that the driving experience can be individually adapted to the driver's wishes. Sound is of great importance for individual well-being, and in combination with other functions in the vehicle, it can provide an emotional driving experience. One feature of future in-car sound systems should therefore be to link immersive sound events to various functions in the vehicle.

Channel-Based In-Car Sound Systems

The audio concept used in today's cars follows the channel-based approach. This implies that the spatial information of the audio scene is mixed and saved in loudspeaker signals (channels), valid for a dedicated loudspeaker setup. One of the main disadvantages is that the same loudspeaker arrangement is always required for playback as well as for production, which is impossible to realize in today's vehicles due to the limited loudspeaker mounting options. Since this mix is not compatible when the playback setup is different, the spatial mix must be recreated or adapted separately for each speaker setup of a vehicle. Apart from the fact that this is very time consuming, it is far from resulting in the perfect mix as intended by the audio engineer. As different car lines have different speaker setups, and the number of in-car speaker channels increases, the problem will become more acute with new use cases requiring immersive audio. The use of object-based audio as a platform technology can overcome these limitations.

Object-Based Audio

Object-based audio (OBA) is a concept for storage, transmission, and reproduction of audio content. An audio object can be seen as a virtual sound source that can interactively be placed in space. The arrangement of all audio objects and their metadata is called an audio scene. Instead of mixing dedicated loudspeaker signals separately, object-based audio scenes are independent of the loudspeaker setup and therefore content only needs to be produced once and can be rendered on any setup. An object is defined by an audio signal and corresponding metadata. The metadata stream contains the properties of all audio objects, such as position, type, or gain. The process of calculating reproduction signals is termed rendering. This is depicted in **Figure 1**.

The audio renderer is a piece of software that uses the loudspeaker coordinates of the current speaker setup to calculate the loudspeaker signals from the individual input audio signals in real time. This calculation is based on mathematical, physical, or perceptual models and controlled by the metadata of the spatial audio scene. The audio input stream can be provided by any multichannel audio player. In contrast to channel-based audio production, an object-based production does not result in ready-mixed loudspeaker signals. Instead, an audio signal

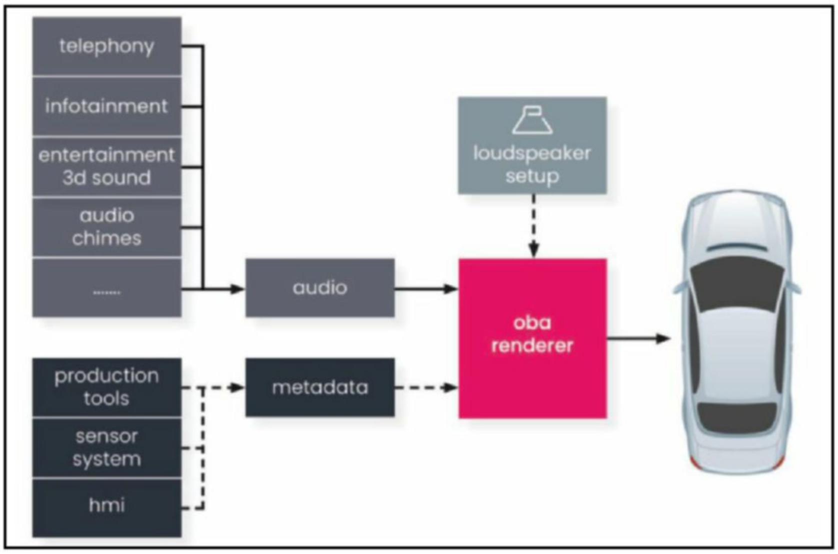


Figure 1: The object-based renderer calculates the loudspeaker signals for a prevailing loudspeaker setup from incoming audio signals and metadata in real time.

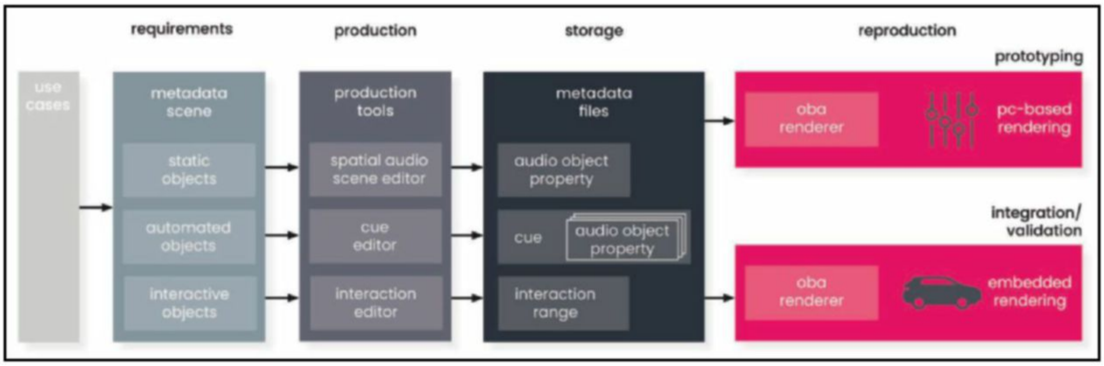


Figure 2: Workflow to use the object-based audio approach as a platform technology in vehicles. Specific use cases define the requirements for the metadata scene, which is created using production tools. The results are metadata files that can be used to prototype sound scenarios and to validate them in the car environment.

and a set of corresponding metadata for each audio object are stored. The integration of object-based audio into today's car loudspeaker system is easy, as no special loudspeakers setups are needed. Minimal setups can consist of only a few speakers, distributed in 2D up to multichannel setups including height speakers. The commissioning of an object-based vehicle playback system is not significantly different from the previous way of working. Since the object-based renderer takes over the spatial mapping of the audio scene, there is no need for time-consuming tuning processes in the form of manual adjustment of delay and gain values. Nevertheless, separate tools are available for tuning, which intuitively allow adjusting the audio objects or an extended rendering configuration.

Production Workflow

To use the object-based audio approach as a platform technology in vehicles, the workflow depicted in **Figure 2** needs to be supported. For this, Fraunhofer IDMT provides a full production and reproduction software suite.

To arrange audio objects in the form of audio scenes, the spatial audio scene editor can be used, which is shown in **Figure 3**. The user interface is divided into two interaction areas. While on the left side a large part of the properties of the audio objects can be set, the right area is used for positioning the audio objects. The audio objects are represented by colored circles.

Depending on the use case static objects, automated objects, and interactive objects represent the range of basic ways to control virtual audio sources. If the audio objects are to be static—their properties do not change over time—this can be realized in a simple way by placing them in the desired location and setting the attributes accordingly. An example in the field of Immersive Audio use cases is the reproduction of channel-based audio data on arbitrary loudspeaker setups. This can be implemented by using static audio objects as virtual speakers at standardized

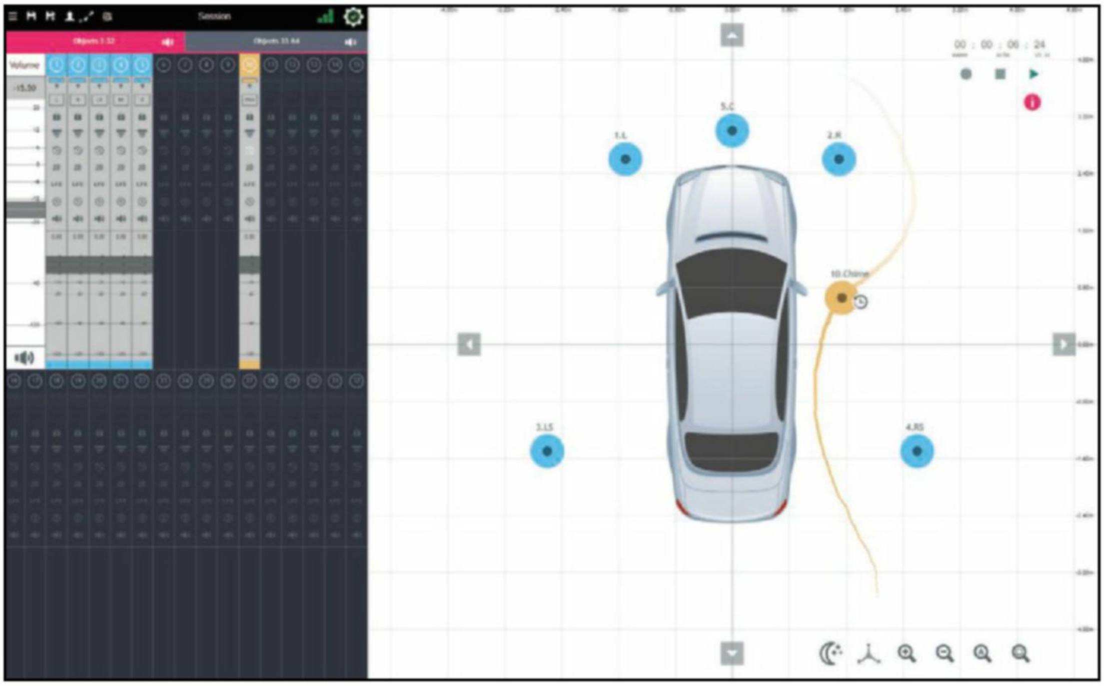


Figure 3: Spatial audio scene editor to produce object-based audio scenes. The blue objects form a static audio scene representing a virtual surround sound loudspeaker setup. The yellow line shows the hand-drawn creation of an object movement.

Figure 4: Spatial audio scene editor is used to produce cues based on linear and circular templates.

positions. In Figure 3, this is illustrated by the five blue audio objects whose placement represents a virtual speaker setup. Other use cases require pre-produced positions and property changes of the audio objects over a given time. This can be realized using cues. A cue is a value sequence that is started by means of a trigger (e.g., event or pressing a button) and can consist of stored metadata and actions over a defined period. An example is a welcome chime that is moving around the occupants as shown in Figure 3 with the yellow audio object. In this case, the metadata of the audio objects are captured in a time-dependent manner by a freehand creation of the properties.

In contrast to the freehand source editing, **Figure 4** shows a mode for defining object properties

Fraunhofer IDMT

sense one

based on linear or circular sequences of values. This approach is useful when audio objects are to be presented in a template-like manner. The parametric definition of predefined motion paths and value curves enables the producer to create complex audio scenes in a very simple and fast way and gives the possibility to reuse or modify the results. The circle and the three lines in Figure 4 show the predefined movement patterns of four audio objects.

In case the audio objects need to be controlled interactively, the interaction editor can be used to ensure that all interactive properties are restricted to sensible ranges. So, movements may be limited to some volume outside the vehicle, and gains may not exceed predefined limits. An example is a sound to draw attention to the specific position of vulnerable road users. In such a scenario, the sensor system for environmental monitoring is detecting the position of a weak road user. This data can be used to map an audio signal to the position of the event in the real world. The sensor module thus can trigger the audio playback and at the same time the virtual source moves to the required position. A translation unit can be used to apply perceptually correcting operations like translation and rotation to the incoming data, or to simply transform the data into the metadata format, that is understood by the rendering unit. This is depicted in **Figure 5**.

Figure 5: The position of the cyclist is detected by the vehicle sensor system and mapped to a corresponding virtual sound source.

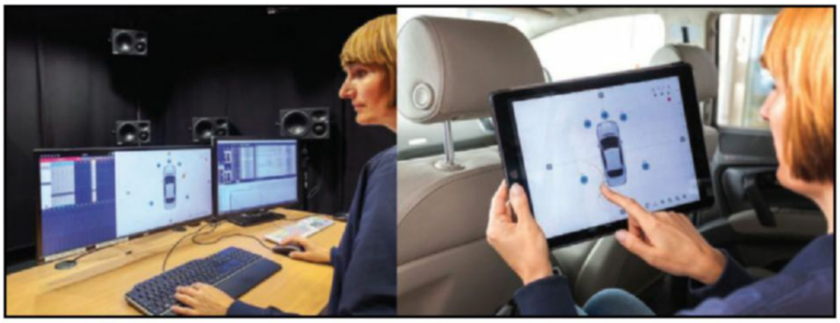


Figure 6: Prototyping of object-based immersive audio content in the studio and its acoustic validation on embedded devices in the vehicle.

About the Author

Christoph Sladeczek heads the Virtual Acoustics research group at Fraunhofer IDMT. His research interests include the investigation and development of innovative sound reproduction methods as well as real-time audio signal processing algorithms. He is author and co-author of several scientific papers and textbook chapters. Christoph is an active member of the German Society for Acoustics (DEGA) as well as a member of the Academic Board DACH of the SAE Institute. His activities include reviewer activities for the Audio Engineering Society (AES) as well as public project proposals.

Prototyping, Integration, and Validation

All metadata files containing the spatial audio scenes are saved in the metadata storage, which acts as an interface between production and reproduction. On reproduction site the metadata storage is responsible for providing the created control data and definitions to the object-based renderer when needed. The renderer receives the interaction parameters and—depending on the type and content of the parameters—requests and executes the corresponding metadata actions. A distinction can be made between PC-based prototyping and the actual integration into an embedded platform. PC-based rendering can be used in a studio for the design of the audio content as well as the production of its spatial mapping. Here the renderer is designed to be used with any digital audio workstation, so that the sound designer can use all favorite tools. While PC-based rendering is used for production in a studio or testing of the created content, the integration step must be performed on the vehicle hardware for validation under real conditions. The data basis is identical for both use cases and the result is therefore suitable for comparison, when using metadata scaling concepts. As can be seen in **Figure 6**, the transfer and comparison between studio production and in-vehicle playback is possible in this innovative way, because with the existing modules the workflow is greatly simplified, and the time required is reduced enormously.

Embedded Rendering

For the rendering on embedded automotive devices, several implementations are available. This also includes software modules for DSP Concepts' AudioWeaver. The core of the object-based rendering system is the OBA Coefficient Calculator. It contains the model to calculate rendering coefficients using the metadata stream. As the rendering coefficients contain the information on how the audio input of corresponding audio objects will be transformed into output (loudspeaker) channels, they will be applied by using signal processing matrices with the size of N × M where N is the number of audio objects and M the number of output channels. While the matrix operation must be carried out in real time, the coefficient calculation only needs to be active, whenever a source object changes one or more of its properties.

To provide maximum flexibility while conserving available computing resources, the object-based rendering can be divided into several separate units. This allows different departments to be assigned their own rendering units and thus act independently of each other. The parallelization of the processes thus increases the speed of implementation. Each unit is configured using the same data and the results of all units are mixed into the final loudspeaker signals. **Figure 7** shows an example of such a system.

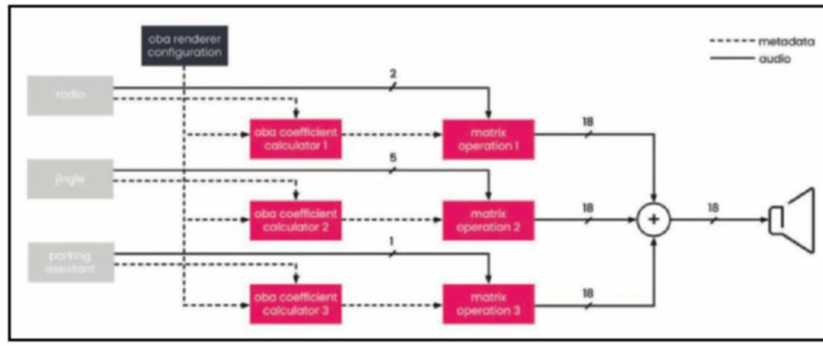


Figure 7: Parallel execution of several rendering units support independent development of separate use cases in the development process.

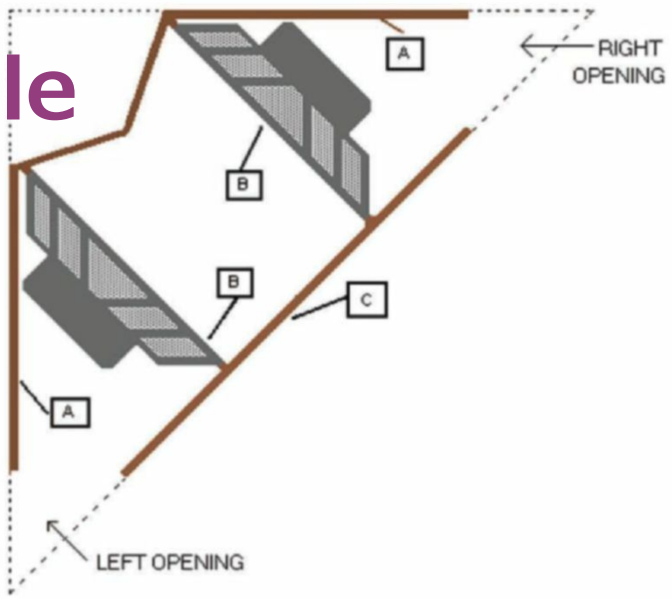
Availability

Using the object-based audio approach in vehicles will revolutionize the way audio is controlled for in-car sound systems. The technology is available as a software bundle supporting the full development process. This includes real-time implementations for development and series production, graphical authoring tools to create object-based scenes as well as tuning tools for the sound setup. The technology was developed by Fraunhofer IDMT. To support customers' value chains in the best possible way, the technology will be made available soon through the company Sense One. Clicking on the QR codes for Fraunhofer IDMT and sense one will provide you with more information.

The Double-Dipole Subwoofer

Take Two

By
George Ntanavaras



Looking back at some speaker projects published in *Speaker Builder* and the first years of *audioXpress*, we find concepts and designs that are still being built today and frequently referenced. That is the case with the Double-Dipole Subwoofer, which deserved to be revised and republished.

Foreword

I was pleasantly surprised when João Martins, editor-in-chief of *audioXpress*, e-mailed me saying that he would like to recover one of my past articles and he was asking for my thoughts.

This happened to be the first article that I submitted to *audioXpress*, almost 20 years ago. I still remember the agony I had before and then the great pleasure I felt when Ed Dell agreed to publish the article!

Around 2000, I discovered the site of Siegfried Linkwitz and I was excited. I read everything on the site a lot of times and the "Single Dipole subwoofer" shown in **Photo A** (without the top cover) was my first serious try on dipole loudspeakers. I was partially satisfied with the result probably, as I discovered later, because I listened to the subwoofer on its theoretically null axis. I decided to proceed with the "Double-Dipole Subwoofer" in an attemp to fix this. I remember that I was not very sure that this idea would be good enough for bass reproduction and I dared to send an email to Mr. Linkwitz, describing the project and asking for his opinion and advice. Mr. Linkwitz was very kind and sent me a response, urging me to build the subwoofer, describing it as "a novel variation of W-frame." I was very happy with the response and I built the subwoofer.

Photo A: My first attempt with a single-dipole subwoofer, inspired by Siegfried Linkwitz.

And indeed, they were very nice subwoofers with very "transparent" bass reproduction. They played very well and loud enough with almost any kind of music. I used them for a lot of years, and I replaced them only when I built my first full-range open-baffle stereo loudspeakers that also had dipole woofers based on four 10" Peerless drivers.

Re-reading the article after all these years, my only wish was that we had the miniDSP device back then. What a much easier job I would had! The equalization would have been finished in a couple of hours and not in a couple of weeks.

For the present article, I made some corrections that I think better clarify several things, I replaced some of the figures with better quality new ones and I included the original colored figures and photos. I hope you enjoy the article.

***

Open-baffle loudspeakers offer a lot of benefits to the reproduction of the audio spectrum. Complete freedom from internal box reflections and dipole radiation are the most important of them. The result is an openness and neutrality to the reproduction of the music that no other loudspeaker can offer.

My previous loudspeakers were full-range open-baffle speakers; each one of them used a 12″ woofer for the bass reproduction and a large custom-made electrostatic panel for the frequencies above 600Hz. I listened to these loudspeakers for several years and I was very pleased with its sound.

But these speakers were big and with the arrival of the home cinema and its multichannel sound, it was not feasible to put five of them in my listening room.

My first decision was to maintain two systems in my room, one with the two open-baffle loudspeakers for the music reproduction and another with five small loudspeakers and a closed-box subwoofer for the home cinema. So, I bought five small loudspeakers and I constructed a closed-box subwoofer for the home cinema using two 12″ woofers similar to the ones used in the open-baffle speakers. After living with these two separate systems for some time, followed by the arrival of the multichannel SACD and DVD-Audio, it was obvious that the only system that would remain was the multichannel one. So I decided (one more time) to exchange the loudspeakers and the subwoofer for new ones.

All that time, I was planning to build an open-baffle subwoofer, since with the closed-box subwoofer that I used for the home cinema system, I missed the clarity and the neutrality of the low range that I had when I used the open-baffle loudspeakers.

About that time I found Siegfried Linkwitz' website (www.linkwitzlab.com) where Mr. Linkwitz presents a very good analysis about open-baffle speakers and the construction of an open-baffle subwoofer with its equalization. But the shape of this subwoofer was not suitable for installation in the corner of my listening room.

So I designed and built this open-baffle subwoofer that could be installed in the corner of my listening room (**Photo 1**). Maybe it can be installed in your room also.

The Requirements

My projection TV was placed in the corner of my listening room. Behind the TV box there was a lot of unused space, so I thought it was the ideal place for the subwoofer, first because the corner could boost the low frequencies and second, because it could hide the huge volume of the subwoofer.

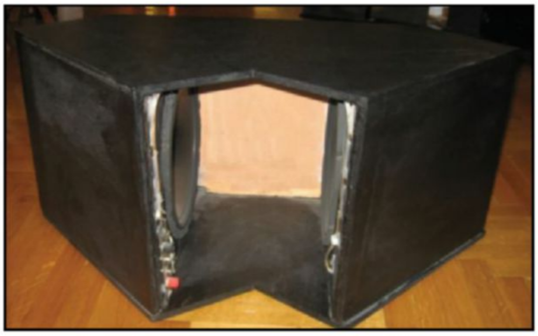


Photo 1: This is the completed open-baffle subwoofer that I designed, built, and installed in the corner of my listening room.

I had four Infinity KCS-120IB 12″ woofers of excellent quality and suitable for open-baffle systems left over from previous projects.

According to the manufacturer, their features included: IMG injection-molded graphite cone, high temperature Kapton/Nomex voice coil formers, Kapton laminated copper ribbon voice coil lead wire, and extremely high power reception.

The Thiele-Small parameters of the woofers as I measured are shown in **Table 1**.

The Open-Baffle Theory

One of the most important factors for the design of the open-baffle subwoofer was the distance that separates the positive and the negative sides of the woofer's radiation.

There are several different shapes that can be used as open-baffle subwoofers. The purpose of all of them is to increase the distance between the two opposites sides of woofer's radiation in such a way that the total size of the subwoofer will remain as compact as possible.

One of the simplest shapes used to perform this task is the H-frame as shown in **Figure 1**. The woofer is placed on a baffle that has extended its sides both in the front and in the back. The result is that the distance between the positive and negative sides of the woofer radiation becomes larger while the total size of the subwoofer is not much bigger.

A typical near-field response (with the microphone placed in front of the one opening as indicated in Figure 1) is shown in **Figure 2**. The peak in the response is caused by the λ/4 resonance of the

Parameter	Woofer #1	Woofer #2	Woofer #3	Woofer #4
Qes	0.735	0.712	0.808	0.72
Qms	11.23	10.14	11.72	10.23
Qts	0.69	0.66	0.75	0.67
Fs	34Hz	31.5Hz	32.7Hz	30.7Hz
Rdc	3.79Ω	3.85Ω	3.94Ω	4.03Ω
Sd	0.053m²	0.053 m²	0.053m²	0.053m²
Xpeak (According to manufacturer)	6.75mm	6.75mm	6.75mm	6.75mm

Table 1: The Thiele-Small parameters of the woofers as I measured.

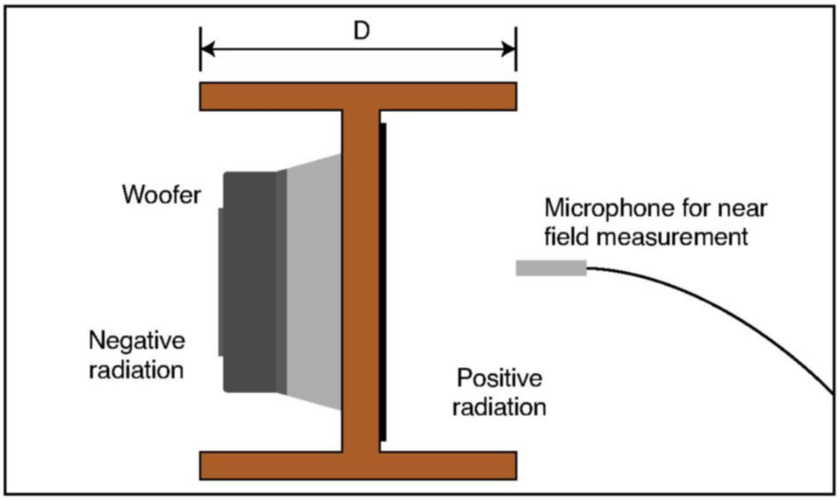
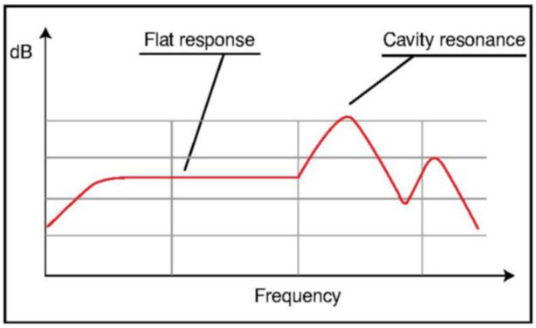


Figure 1: One of the simplest shapes used to increase the distance between the two opposites sides of woofer's radiation is the H-frame.

Figure 2: Here is a typical near-field response (with the microphone placed in front of the one opening as indicated in Figure 1).



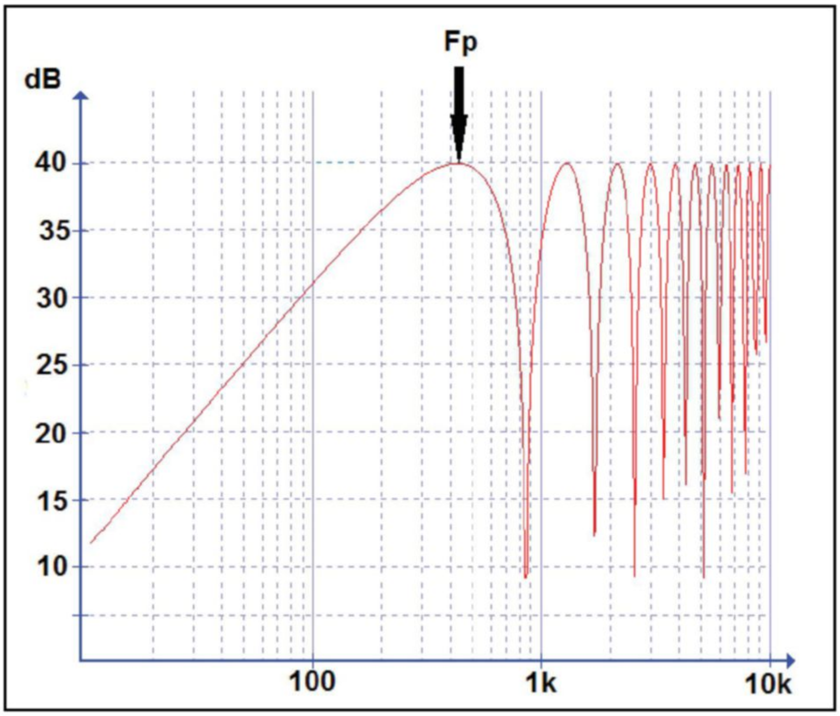


Figure 3: This is the far-field response of the H-frame.

cavity that is created in front (and similar behind) of the woofer. In the near-field measurement the negative side radiation of the woofer does not come into play and the frequency response below the cavity resonance frequency remains flat (if Qts of the woofer is high enough) down to the woofer resonance frequency.

The far-field response is shown in **Figure 3**. The frequency Fpk where the first peak occurs is given by:

$$Fpk = (0.5 \times c)/D$$

where c is the sound's velocity and D is the distance between the openings.

The peak in the frequency response results when the back (negative) radiation, after 180° of phase shift due to the distance D, becomes positive and adds to the front (also positive) radiation. The resulting sound pressure level (SPL) is then twice that of the single-source SPL (+6dB). Above the peak frequency, there are also a lot of other similar peaks and notches due to the same reason.

Below the peak frequency Fpk, the sound pressure falls with 6dB/octave due to the cancellation that is caused between the two radiation sides. At the low frequencies, the distance D is small when compared to the wave length of the sound and cannot provide enough isolation between the positive and negative radiation.

The frequency at which the open-baffle speaker has the same output level with a single similar monopole (e.g., a closed-box subwoofer) is given by:

$$Feq = (0.17 \times c)/D$$

Above this frequency the open-baffle speaker has more output than a single monopole and below this frequency it has less output that falls with 6db/octave where the single monopole has a flat response.

In order for the open-baffle subwoofer to be useful for music reproduction, it is absolutely necessary to equalize the peak in the frequency response that it is due to the cavity resonance using a notch filter and then the far-field low-frequency roll-off by boosting the low frequencies with 6db/octave to flatten the frequency response. This will increase the woofer's excursion and the linearity will be worse. That is why, in order to get high volumes of sound at the low frequencies with an open-baffle subwoofer, you need a lot of woofers.

The directivity of the open-baffle subwoofer is also maintained at the low frequencies. The response drops -3dB when the listening point is

45° off the dipole axis and -6dB when it is 60°, thus for maximum sound pressure you should remain in front of its axis.

A "Novel Variation of the W-Frame"

I already mentioned that I intended to install the subwoofer in the corner of my living room, and unfortunately the H-frame was not suitable since it requires a lot of free space in front and in back of the cabinet.

After several thoughts and attempts, I designed the cabinet shown in **Photo 2** (pictured without its top cover). This cabinet has three openings: the common, the left and the right opening. The two woofers were placed one facing the other. The common opening of the cabinet was placed in the corner, while the right and left openings were placed at the sides of the corner.

The result was that the positive radiation of the woofers comes out from the left and the right opening while the negative radiation of both woofers comes out from the common opening. Mr. Linkwitz characterized this frame (in a personal communication with e-mail) as a "novel variation of the W-frame."

In **Figure 4**, the top view of the cabinet is shown with detailed dimensions for the construction. The W-frame can be considered as equivalent to two dipoles at almost right angles to each other. The response at a point that is in the far field will be equivalent to the sum of the responses of the two dipoles.

Photo 2: To fit the space, I designed a unique cabinet, the W-frame subwoofer cabinet, with three openings: the common, the left, and the right opening.

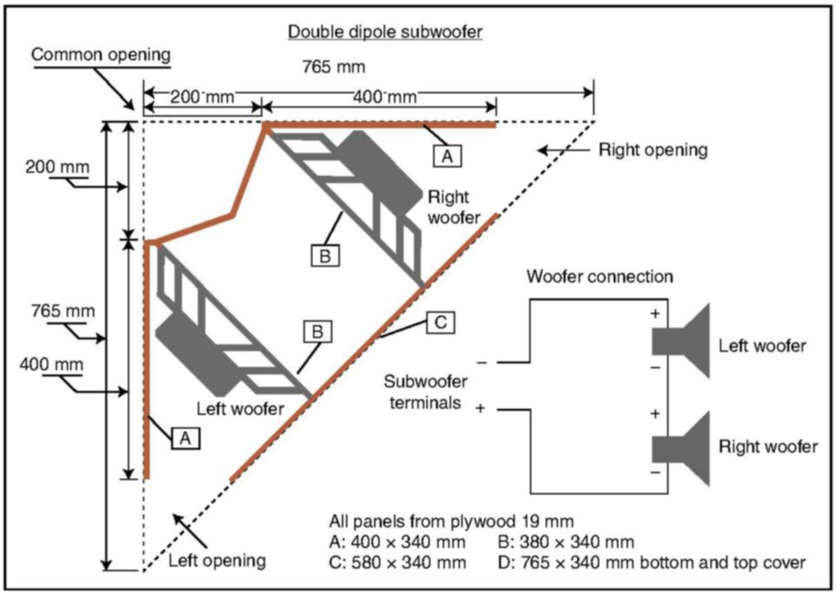


Figure 4: The top view of the cabinet is shown with detailed dimensions for the construction.

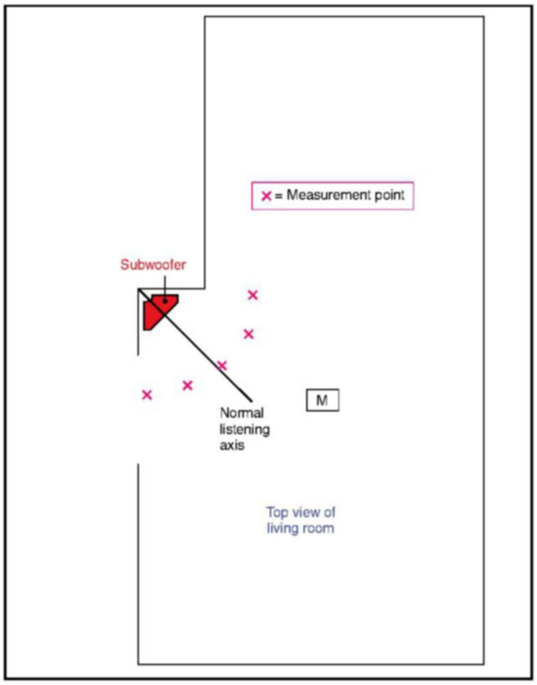


Figure 5: Here is the double dipole directivity measurement.

Frequency	Maximum SPL (In half-space with 4 woofers)
20Hz	87dB
30Hz	97dB
40Hz	105dB
50Hz	111dB
60Hz	115dB
70Hz	119dB
80Hz	123dB
100Hz	129dB
150Hz	139dB

Table 2: Considering the parameters, I calculated the maximum SPL in half-space with four woofers.

Since the nominal listening axis will have an angle with the axis of each dipole, in theory, a drop in the amplitude response of the dipole will occur. Fortunately, this is valid when the subwoofer is installed in the free field and not in the corner of a real room. I verified this by measuring the response of the subwoofer as shown in the **Figure 5** where the top view of the listening room is shown with the subwoofer in the corner. The subwoofer was driven with pink noise, band filtered between 45Hz and 85Hz. I used a RadioShack SPL meter for the measurement. At all five positions that I measured, the result was the same, meaning that the directivity of the subwoofer when installed in the corner would not be a problem.

Some important factors of the subwoofer's operation include: the distance between the two openings of opposite phase of the W-frame is not constant; it starts from 40cm and goes up to 48cm. For the analysis, I took as distance D the mean value of 44cm. The peak frequency is calculated as follows:

$$F_{pk} = 0.5 \times 344/(0.44) = 391\text{Hz}$$

The frequency at which this open-baffle subwoofer has the same output level with a single monopole is calculated as follows:

$$F_{eq} = 0.17 \times 344/(0.44) = 133\text{Hz}$$

Another important factor for a subwoofer is the maximum SPL that can produce with the frequency. Mr. Linkwitz provides an Excel file that can be found at: www.linkwitzlab.com/spl_max1.xls for calculation of the excursion limited RMS (SPL) for a driver in an open baffle, in half space.

In my design, I used four woofers, each of them having Sd = 530cm^2 and Xpk = 6.75mm. The effective path difference was D = 0.44m.

Considering the parameters, the maximum SPL is shown in **Table 2**. Since the subwoofer would be installed in the corner of the room, I expected additional gain of several dB at the low frequencies.

The Construction

For the construction of the subwoofer, I used 19mm plywood for every side of the cabinet. (Editor's Note: For construction details please refer to the original article published in *audioXpress* November 2004.)

The two woofers of each cabinet are connected electrically in series due to their low nominal 4Ω impedance as shown in Figure 4. The minus (-) terminal of the left woofer is connected to the positive (+) terminal of the right woofer. The (+)

terminal of the left woofer is connected to the (-) terminal of the amplifier and the (-) terminal of the right woofer to the (+) terminal of the amplifier. I did this to maintain the absolute phase of the subwoofer with the rest of the system. I built two such subwoofers and the whole subwoofer system included four 12″ woofers. **Photo 3** shows the complete subwoofer cabinet before the two drivers were installed and without the bottom cover.

Photo 3: This image shows the W-frame subwoofer cabinet with the two woofers.

The Measurements

After I completed the construction of the cabinet, I proceeded with the equalization of the response of the subwoofer by measuring the subwoofer's near-field response. For all the measurements I used the JBL Smaart Pro Real Time Module version 2.1. The frequency resolution of all the measurements was 1Hz.

First, I placed the subwoofer on the floor in the middle of my listening room (position M shown in Figure 5) with a distance greater than 1m from any nearby obstacle. The result of these measurements are shown in **Figure 6**. The measurement on the top was taken with the microphone placed in the front of the common opening of the subwoofer. The other two measurements below are the near field response with the microphone on the left opening of the subwoofer and on the right opening of the subwoofer. (Please refer to Figure 4 for the naming of the common, left, and right openings.) The measurements on the left and right openings are similar and this proves the symmetry of the design.

The near-field response measured at the common output of the subwoofer has some differences from the measurements on the left and right opening. The left and right opening have a more broad peak response and the peak frequency is different.

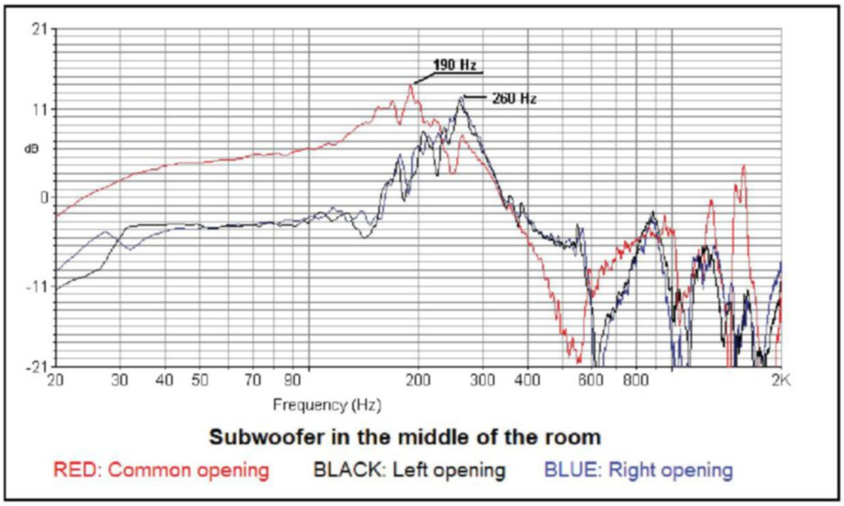


Figure 6: This measurement is of the dipole subwoofer in the middle of the room.

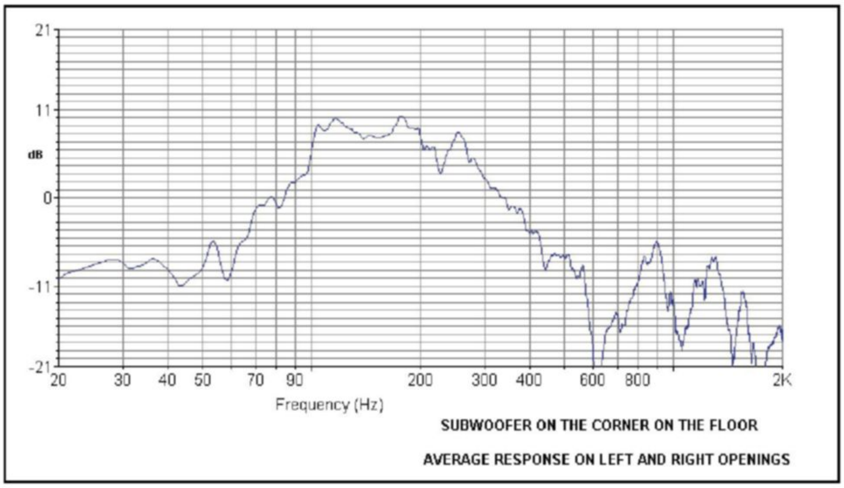


Figure 7: Here I measured the dipole subwoofer in the corner of the room and on the floor.

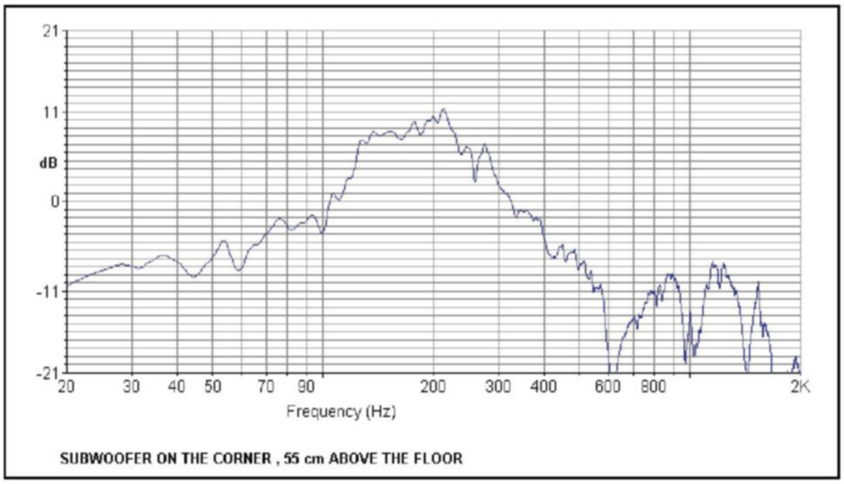


Figure 8: This is the subwoofer response in the corner at 55cm above floor.

Resource

G. Ntanavaras, "The Double-Dipole Subwoofer," *audioXpress*, November, 2004.

It is 190Hz for the common opening and 260Hz for the other openings. This can be explained because the left and right openings of the subwoofer have different shape from the common opening and thus different cavity resonance frequencies. From about 30Hz to 120Hz, which is the range that the subwoofer will reproduce, the responses are similar at all three openings.

After these measurements, I put the two subwoofers one above another in the corner of the room behind the TV box to measure how different the near-field frequency response will be from that taken in the middle of the room. The result is shown in **Figure 7**. This response is the average of four measurements taken on the two left and the two right openings of the subwoofers. As you can see there are a lot of differences. The peak at 260Hz is still present but as expected, the corner has broadened significantly the peak of the response, so now it starts from about 70Hz and goes up to 200Hz. Also it has much more output at these frequencies. Actually from 100Hz to 200Hz it has a flat response.

I was thinking of possible ways to make the response of the subwoofers in the corner similar to the one in the middle of the room. A good solution was found when I put the subwoofers 55cm above the floor. The responses are shown in **Figure 8**. The main differences are in the range between 70Hz and 120Hz. The subwoofers in the new position have less output, about 4dB at 90Hz and 9dB at 110Hz.

The response was not the same as the one I took in the middle of the room, and which I believe had the lowest resonance, but it was the best I could get after placing the subwoofer on the corner. So I decided that this would be the final position of the subwoofers, and I used this response for the design of the equalizer.

Photo 4: I installed the subwoofer in the corner of the listening room with the equalizer and the power amplifier underneath.

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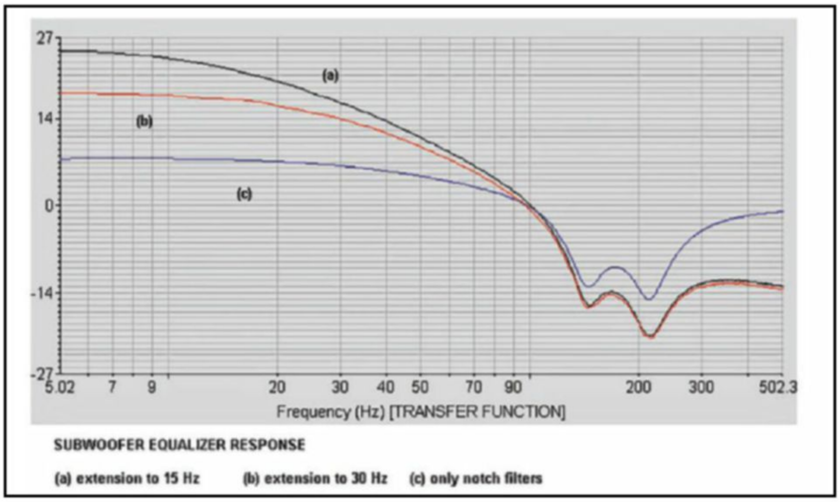

Figure 9: This graph shows the measured response of the equalizer.

Photo 4 shows the two subwoofers as finally installed in the corner of my living room, 55cm above the floor. On the right side you can see the projection TV that was slightly removed from its position. Below the subwoofers, are the equalizer, its external power supply, and the power amplifier that drives the two subwoofers. (Editor's Note: details of the original equalizer construction from the original article not included here.)

After the construction, I measured the frequency response of the equalizer. **Figure 9** shows the result. Three curves are shown. The top curve is the total response of the equalizer when the bass extension is set to 15Hz. The middle curve shows the total response set at 30Hz. The bottom curve is the response with a low boost circuit. All the measurements shown in Figure 9 were in excellent

Appendix

The notch filter as used in the subwoofer equalizer uses the topology shown in **Figure A**. A typical frequency response of the circuit detailed in Figure A is shown in **Figure B**. The parameters for the design of the notch filter are:

FR = the notch frequency
Q = defines the width of the notch
a = defines the attenuation at the notch frequency

The components of the above notch filter can be computed as follows:

Select R1
Then R5=R1 × a / (1-a)
Then C1 = 1 / (Q × R5 × 2 × π × FR) where π=3.14159
Select R4 > 10 × Q^2 × R5
Select R3 <= R5
Then C2 = (Q × R5) / (2 × π × FR × R3 × (R4-R3))

As an example let's compute a notch filter with the following parameters:

FR = 150Hz
Q = 8
a = -15dB = 0.1778

Then we have:

Select R1 = 2kΩ
Then R5 = R1 × a / (1-a) = 432.5Ω
Then C1 = 1 / (Q × R5 × 2 × π × FR) = 306.7nF
Select R4 > 10 × Q^2 × R5 = 277kΩ. Let R4 = 300kΩ
Select R3 <= R5. Let R3 = 390Ω.
Then R2 = R5 - R3 = 42.5Ω and
C2 = (Q × R5) / (2 × π × FR × R3 × (R4-R3)) = 31.4nF

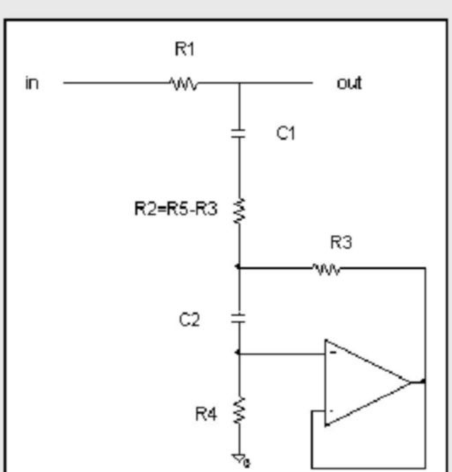

Figure A: This is the topology for the notch filter used in the subwoofer equalizer.

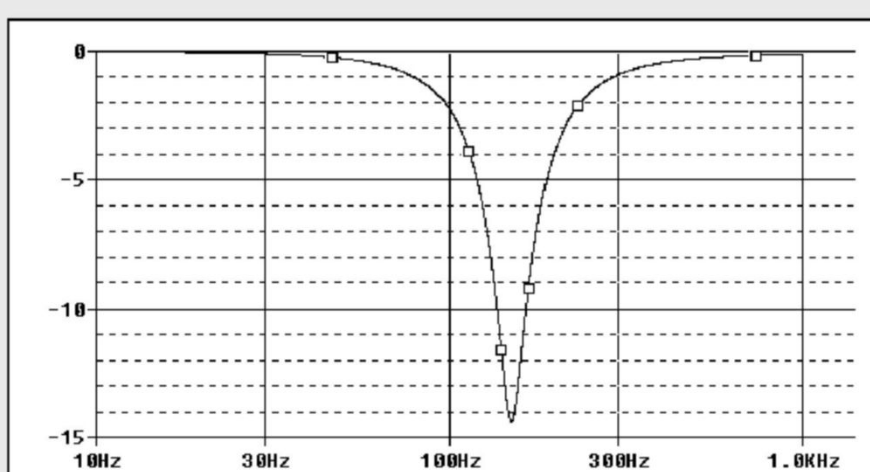

Figure B: Here is a typical frequency response of the circuit shown in Figure A.

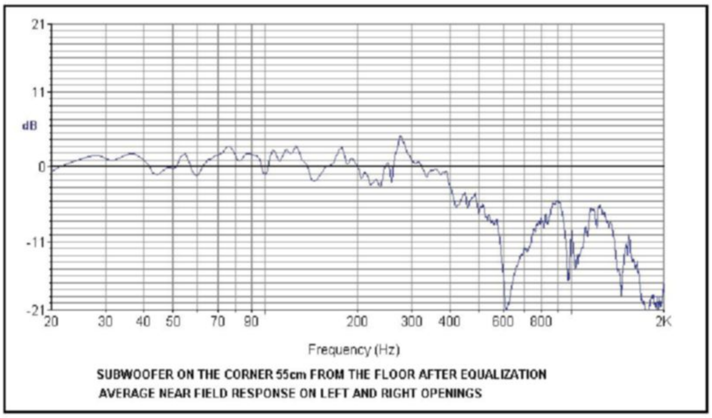

Figure 10: The low-frequency limit was set at 15Hz. The average response of the four measurements is shown here.

agreement with the simulated responses and no corrective actions were needed.

It is easily observed in Figure 9 that the difference between the low and the peak value of the response between 10Hz and 230Hz is about 45dB for the top curve and about 39dB for the middle curve. This difference is very large and great care should be taken on the construction of the circuit to avoid possible noises and disturbances.

After the testing of the circuit, I connected it before the power amplifier and I measured the near-field frequency response of the subwoofer without the circuit around U3B (the low frequency boost for the far field response). The measurements were taken as before at the four openings of the two subwoofers (left and right openings) with the subwoofers placed at the corner and above the floor. The low-frequency limit was set at 15Hz. The average response of the four measurements is shown in **Figure 10**. The response remains flat within ±2 dB between 20Hz and 350Hz. This was an excellent result if you take into consideration the fact that the frequency resolution of the measurement is 1Hz.

Dipole or Monopole Subwoofer

There is a lot of discussion concerning the differences in the bass reproduction between a dipole and a monopole subwoofer. After living with this open-baffle subwoofer for quite some time, I fully agree with the opinion that the dipole subwoofer is much better than the corresponding monopole subwoofer. Since I have constructed both kinds of subwoofers using the same woofers, I can say that the most important difference that I heard between the closed-box subwoofer and the open-baffle subwoofer, installed in the same listening room, is more clarity in the reproduction from the dipole subwoofer. I believe that the open-baffle subwoofer will remain in my listening room for a long time.

Editor's Note: All audioXpress *articles from 2001 to present can be found on the aX Cache, a USB drive available from www.cc-webshop.com.*

About the Author

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

300B Triode Single-Ended vs. Push-Pull vs. a 6CA7 Power Pentode

In this article, our hollow-state expert compares the performance of a Western Electric Model 91A using 300B triodes against a push-pull amp with no overall negative feedback.

By
Richard Honeycutt

The Grandaddy of Excellent Single-Ended Triode Audio Amplifiers

Probably the most highly regarded classic single-ended (SE) triode audio amplifier is the Western Electric (WE) Model 91A, introduced in 1936 by Electrical Research Products Inc., Western Electric's film sound subsidiary. This amp served as the inspiration for the recently introduced WE Model 91E integrated amplifier. (Western Electric has been re-established and is now a manufacturer of electron tubes and high fidelity, www.westernelectric.com/91e.) This article compares the LTSpice-modeled performance of the original 91A, a push-pull hollow-state amp using 300B triodes, and a 6CA7 (EL34) push-pull amp with no overall negative feedback.

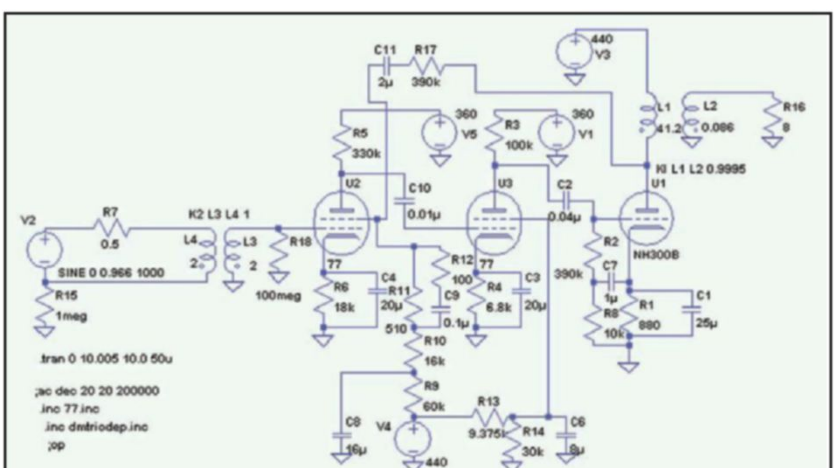

Figure 1: This LTSpice model of a Western Eelectric 91A amplifier has a few changes from the original, as explained in the text.

Figure 2: At 14.97W output, the sine wave is very clean.

Figure 1 shows the LTSpice model for the 91A. Tubes U2 and U3 are 6C6 triodes in the original amp. However, no LTSpice model is available for the 6C6. The 6J7, the 77, and a few other tubes are essentially equivalent to the "triple-grid" 6C6, according to the *RCA Tube Handbook*, and there are LTSpice models for the 6J7 and the 77. (The LTSpice models appear to be identical.) As with many manufacturer-supplied schematics of hollow-state equipment, the WE 91A schematic includes the normal DC operating voltages.

When I built the LTSpice model for the amp, the simulation did not produce those voltages. Therefore, I tweaked the values of the cathode resistors R6 and R4 from the original values of 2kΩ and 1.2kΩ to 18kΩ and 6.8kΩ, respectively. The simulation then failed to indicate the expected gain, so I tweaked the negative feedback resistor R17 from 100kΩto 390kΩ. As you can see from **Figure 2**, the maximum unclipped sine-wave output voltage is a bit over 15Vpeak, for a maximum output of 14.97W into an 8Ω load. The fast Fourier transform (FFT) shown in **Figure 3** indicates a very low total harmonic distortion (THD) of about 0.3%, primarily second and third harmonics. Although triodes are prone to produce more even harmonics,

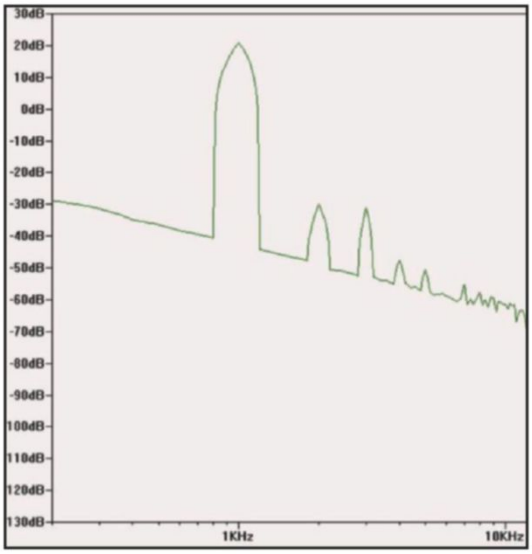

Figure 3: The FFT of the output wave shows low second harmonic distortion, and only slightly higher third harmonic.

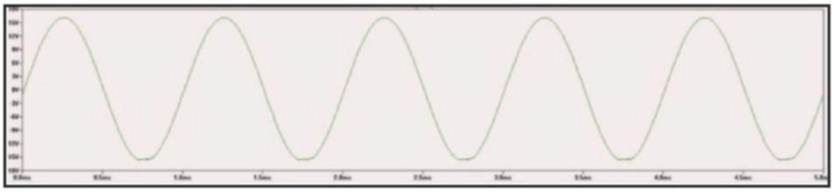

Figure 4: The output wave shows slight clipping on negative half-cycles.

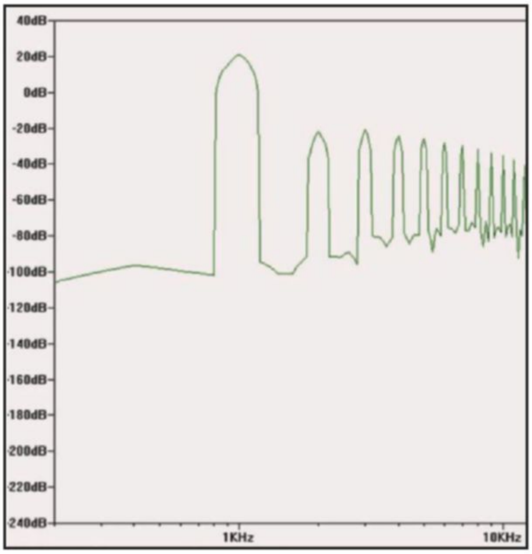

Figure 5: With slight clipping, all harmonic levels increase.

the negative feedback brings the second-harmonic component down slightly below the third.

Another component of the model that may not match the actual 91A is the input transformer. Having no specs on the actual Western Electric transformer, I chose to model an ideal transformer. Not knowing the original's impedance ratio or secondary inductance, I chose a 1:1 ratio and a secondary inductance of 2H.

Another predictable effect of negative feedback is that when the amp begins to clip (in **Figure 4** the amp is about 4% overdriven), a crop of higher-order harmonics begins to sprout. (See the FFT in shown in **Figure 5**. Although the THD for the slightly clipped wave is only 1.05%, the significant increase in upper harmonics would likely make the amplifier sound harsh.

Even though the Western Electric Bostwick horn tweeter used in the best movie theaters in the mid-to-late 1930s had an upper response limit of 8kHz, the WE 91A's response was only 4.4dB down at 20kHz, and was flat down to 20Hz.

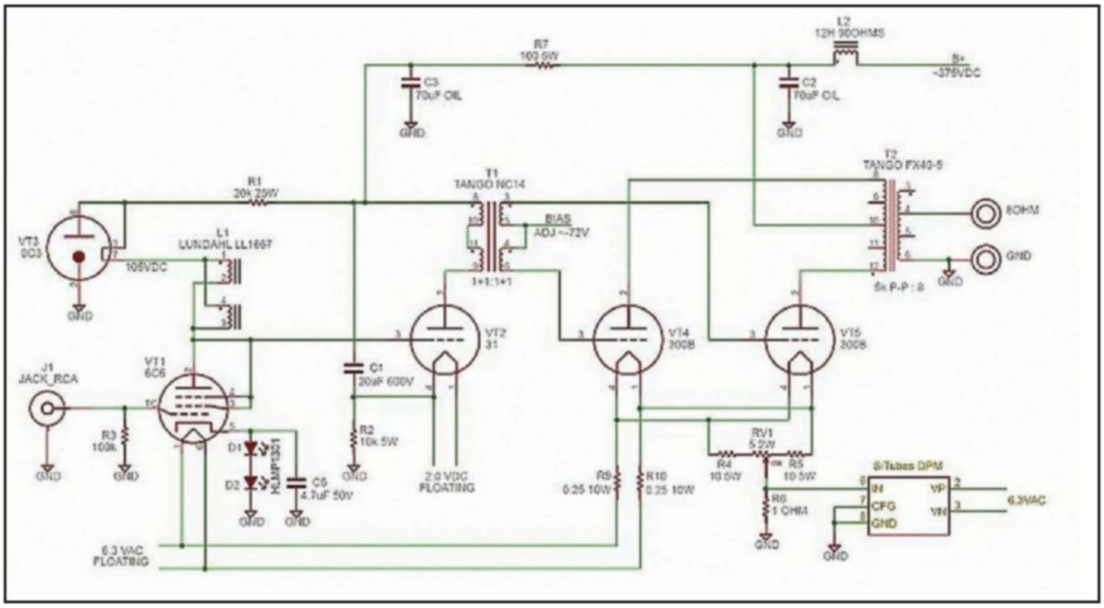

Figure 6: Pete Millett's "Fatboy" PP 300B triode amplifier is truly state-of-the-art!

It is important to note that because of the accuracy limitations of the vacuum tube and transformer models, and the resulting circuit tweaks necessitated for the simulation to give credible results, plus the improvements in transformer technology and performance since the 1930s, the frequency response and distortion data given for the 91A can only be considered to approximate true performance; To the extent that these results may differ from the manufacturer's specifications, the latter should be given more credence.

300B Triodes in a Push-Pull Circuit

The logical push-pull (PP) 300B triode circuit to compare with the WE 91A would be the 92 A or B, introduced shortly after the 91A. However, building an LTSpice model of the WE 92 plunges us into the same difficulties as we found when modeling the 91: tubes for which there are no LTSpice models, and transformers for which insufficient data is available to model them in LTSpice.

However, *audioXpress* contributor Pete Millett has designed a superb "Fatboy" 300B Push-Pull Amp using 300B triodes and posted the design, comments, and test results on his website (http://pmillett.com). **Figure 6** shows the schematic of Pete Millett's push-pull amplifier.

Unlike the WE 92, the Fatboy uses transformers that can handle the full audio frequency range. The choke-loaded 6C6 "triple-grid" preamp tube is the same type used in the WE 91; however, the 92 used WE 262A triodes as preamp, predriver, and driver tubes, and did not use choke loading. The advantage of choke loading is that it stabilizes the DC plate voltage of the preamp stage via the VR tube, preventing small operating-point (OP) shifts in the preamp from causing large OP shifts in the driver stage.

The Fatboy uses a type 31 triode for the driver, and—like the 92, uses a transformer phase splitter. My own research has shown that as the input signal voltage of most hollow-state PP amps is increased, the first serious distortion usually appears in the phase splitter, not the output stage. Although there are differences in distortion performance among the different types of phase splitter—paraphrase, long-tailed pair, or split load—a hollow-state phase splitter is often the weak link from the standpoint of distortion. Using an excellent driver transformer gets around this weakness. (However, there was one widely-used guitar amplifier in

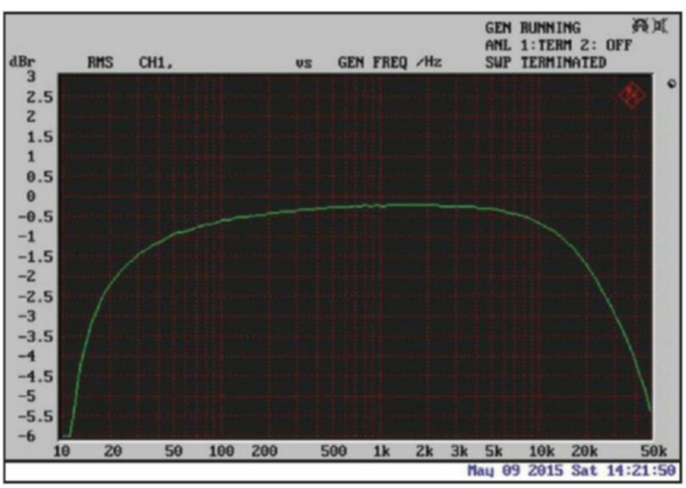

Figure 7: The Fatboy's frequency response is far better than the original WE 91s.

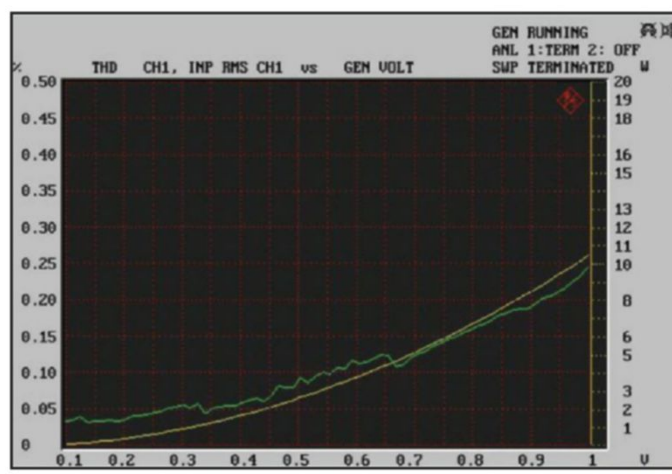

Figure 8: The Fatboy's distortion performance is stellar!

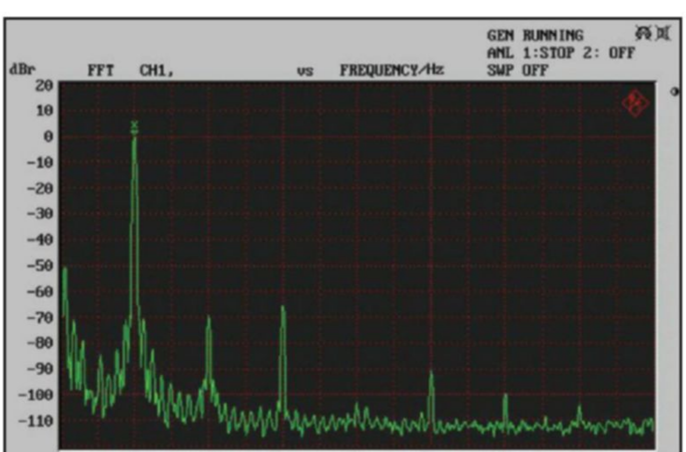

Figure 9: The Fatboy's FFT for a 1-kHz input and 1W output shows mainly second and third harmonics.

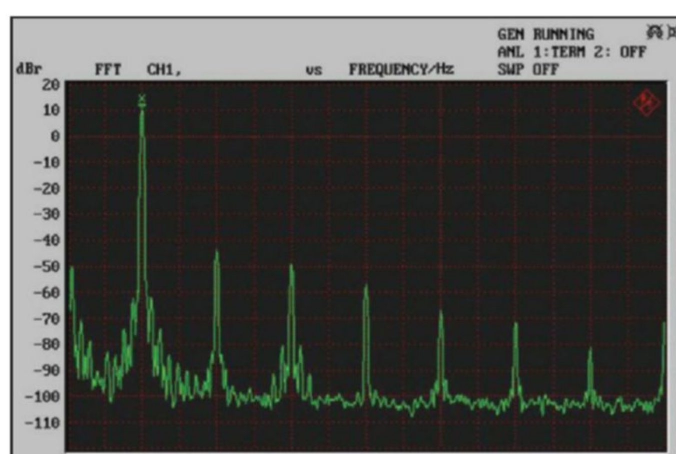

Figure 10: At 10W output (1.6 dB below clipping), additional harmonics appear, though the overall distortion is still quite low.

the early 1970s that used a transformer phase splitter which was under-designed, resulting in a tendency for the primary or secondary to open, especially in the event of an internal short of an output tube; hence the importance of using an excellent driver transformer—not only for low distortion!)

Performance Measurements

On his website, Millett discusses various design decisions he made in creating the Fatboy, so I won't go any further here; instead, I'll move directly to the performance measurements Millett provided. Note that these are actual performance measurements, not computer simulations as I've used for the other amps in this column.

Figure 7 shows the frequency response at 1W output. You'll note that it is -2.2dB at 20Hz, and -1.7dB at 20kHz; -3dB at 15Hz and 30kHz. The amp

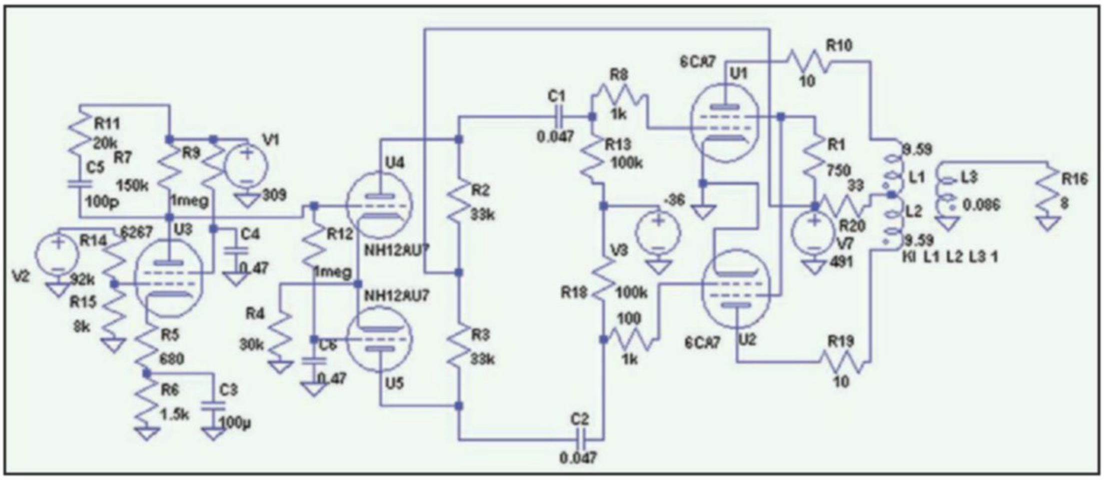

Figure 11: This LTSpice model shows a 6CA7 PP amplifier.

puts out 12W at the onset of clipping, increasing to only 5% at 19W. We don't have an image of the output at various levels, but the linearity and distortion are illustrated by the curves shown in **Figure 8**, which indicate performance in the range below 10W. The THD is only 0.2% at 10W output; note that this performance comes from a circuit using no negative feedback (NFB)—not even unbypassed cathode resistors. The lack of NFB accounts for the slow increase of THD from 12W to 19W output. I will mention that the THD at low frequencies increases to 1% at 1W output at a frequency of 20Hz, because no transformer is ideal.

As shown in **Figure 9**, the FFT reveals bit more third harmonic than second, with no visible fourth harmonic, resulting from the PP circuit's cancellation of even-order harmonics. Fifth and higher-order harmonics are over 20dB down from the third-harmonic level. At 10W (**Figure 10**), the second harmonic is the highest, but still 43dB below the fundamental, and higher harmonics decrease steadily as harmonic order goes up, except for a 10dB increase in the eighth harmonic compared to the seventh, at which point the harmonic level is 70dB below the fundamental (0.03%).

Figure 12: The output wave looks pretty clean at 75W.

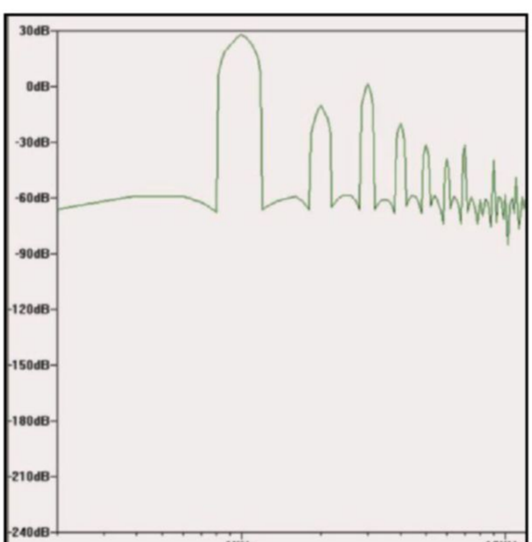

Figure 13: The FFT shows significant harmonics through the seventh.

6CA7/EL34 PP Amp with No Overall NFB

Figure 11 shows a PP amplifier using 6CA7/EL34 output tubes. The only NFB is the partially unbypassed cathode resistor of the 6267 pentode preamp tube. The output tubes' operating point is close to that specified on the Tung Sol datasheet for Class-B operation with a 500V B+. The frequency response is about 1.2dB down at 20Hz, and 7dB down at 20kHz. Since the LTSpice model assumes an ideal transformer, the response of an actual amplifier built with this deign would be a bit less flat.

The output wave at 75W shown in **Figure 12** looks pretty good, considering that there is no overall NFB. The Tung Sol datasheet gives a maximum output power of 70W at a THD of 5%. The FFT shown in **Figure 13** indicates significant harmonics up to the seventh, dominated by the third. Overall THD is 4.73%.

Comparisons

We can compare these amps' frequency responses, maximum power outputs, THD, and distortion signatures. Frequency responses are broadly similar, except that the Fatboy measures flatter at high frequencies than any of the others, even though their (the others') flatness is probably overstated by the LTSpice models assuming ideal transformers. The 6CA7 PP amp has the highest maximum output power, at 75W. The WE 91A below clipping has the lowest distortion, but its output power is also the lowest: just under 15W. At 10W, the Fatboy's distortion is very low, but as would be expected for an amp having no NFB, the distortion increases to roughly the same level as that of the 6CA7 PP amp at full output power. For all the amps except the Fatboy at 10W output, the third harmonic is dominant. Only the Fatboy and the 6CA7 amps have significant harmonics above the third, for unclipped output waves. And as mentioned earlier, the level of the higher harmonics in the Fatboy's 10W FFT is well under 0.1%.

In a future article, we'll examine the performance of two "favorite" PP amps that use NFB: the Williamson and the ultralinear designs.

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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