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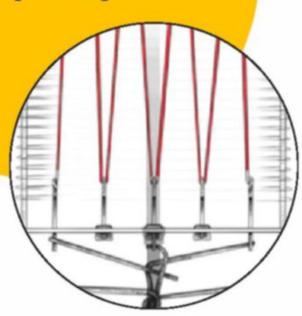

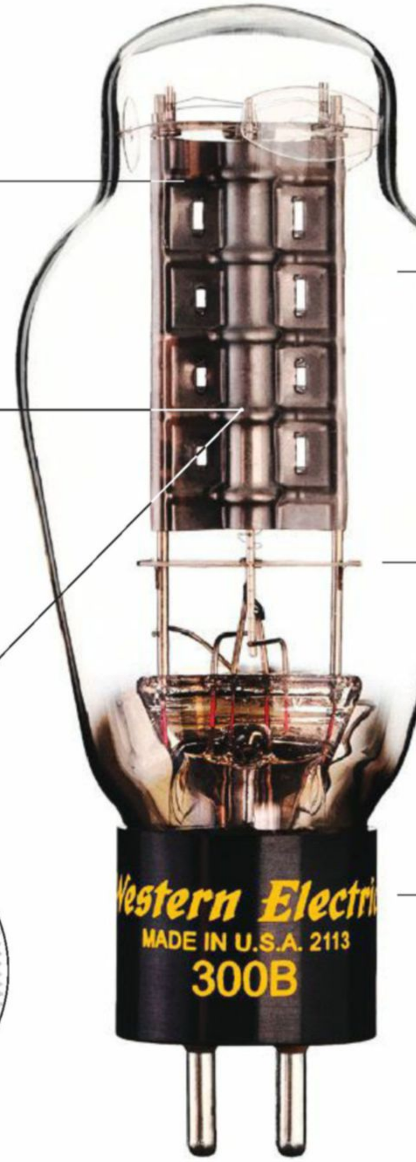

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A New Chapter in Audio Development

This December 2021 edition closes an important chapter for this magazine, having concluded another successful publication year under extraordinarily difficult circumstances globally. This was the year when we celebrated our 50 years from the company's origins and 20 full years of audioXpress magazine. I believe we celebrated in the best possible way—by releasing our best editions every month—even under significant challenges.

Unfortunately, as a result of nearly two years of global pandemic, our own readership and author communities have suffered unfathomable losses—and every day the entire audio industry continues to mourn the departure of more industry legends and more precious talent. And I believe the only way to honor their memories and counteract the feeling of loss is to highlight the work of new talented builders, developers, researchers, and entrepreneurs.

Many of our readers still clearly remember when a magazine was something that arrived in the mail every month and was supposed to provide knowledge, education, and entertainment, fulfilling a variety of needs in order to be perceived as valuable. That included a generous serving of specialized news updates that weren't available elsewhere, reference articles for beginners, and even crossword puzzles to challenge the readers' devotion to the passion that united the magazine's community. I remember including advertisers' business reply cards with all the numbers of the ad pages, so readers could conveniently express their interest.

When I joined *audioXpress* and transitioned from the previous model, this magazine needed a new format and new direction. As I wrote back in December 2013, I tried to combine "more (of what our readers expect), electronics (our roots), and audio innovation (our focus)," respecting the publication's heritage.

We've been doing that since then and I am proud of the publication's evolution within that spirit.

But now it's time to enter a new chapter in our mission of "Advancing the Evolution of Audio Technology" as a global magazine and website. We are not planning any immediate, drastic redesigns or changes. We will simply reinforce the innovation focus and expand the coverage of all audio disciplines, expanding the development, design, and engineering angles.

We believe that part of that evolution is to bring audio enthusiasts, developers, and the industry closer, by promoting the value of everyone's contributions and efforts. And we do that every day. It's not something we state. It's something that needs to be done in practice. Every day.

Among the greatest challenges during these pandemic years was the need to keep the community together and expand on its purpose by offering content of common interest, while continuing to reflect an industry in accelerated evolution. We felt that pressure as our online community expanded, and through the boost in our magazine and newsletter subscriptions—and we could see how our own readership's expectations also evolved.

audioXpress is increasingly relevant because it reflects the frontiers of what needs to be uncovered and shares the details about the products and technologies that are being unveiled and discovered, as well as the journeys of the companies and individuals who are exploring new ground. That is how a trade publication fulfills a unique role—and in our case, that of "Advancing the Evolution of Audio Technology."

These are very exciting times. Audio content is being produced and experienced in multiple new ways, and audio technology is more relevant than ever to multiple industries, and serving many exciting new applications. The audio industry is reflecting that excitement.

In this edition, we addressed speaker optimizations, room correction, new sound response emulation concepts, variable acoustics, and sound calibration for any targeted audio reproduction system. In this last domain, I'm particularly glad to feature a contributed article from Lars-Johan Brännmark, Chief Scientist at Dirac Research, discussing the challenges of applying DSP room correction to complex multichannel setups, intended to meet the requirements of object-based immersive audio formats. His article discusses a completely new frontier for audio, with the new concept of spatial room correction, going beyond frequency domain and time domain optimization. And it hints at the need for a new approach to the architecture of speakers as well, which need to cooperate, allowing room-correction filters to operate simultaneously.

J. Martins
Editor-in-Chief

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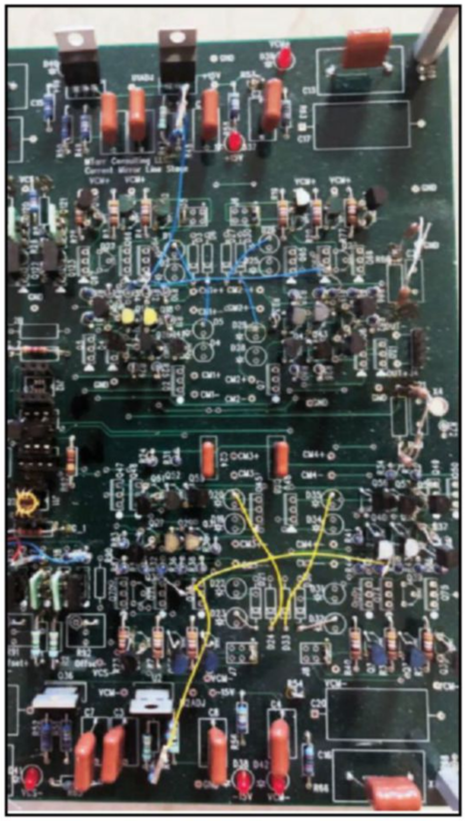

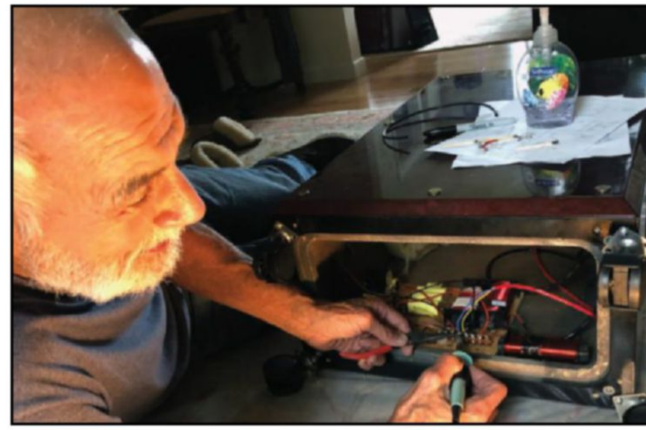

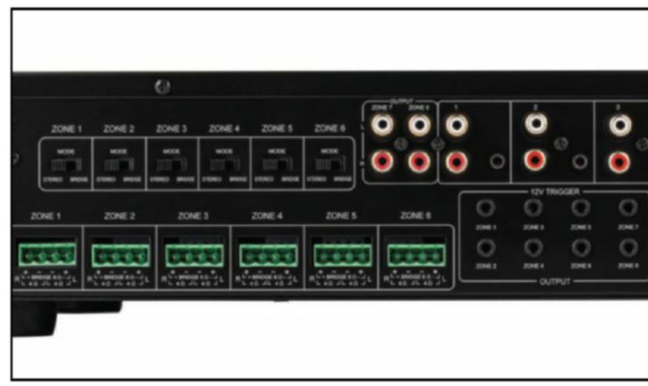

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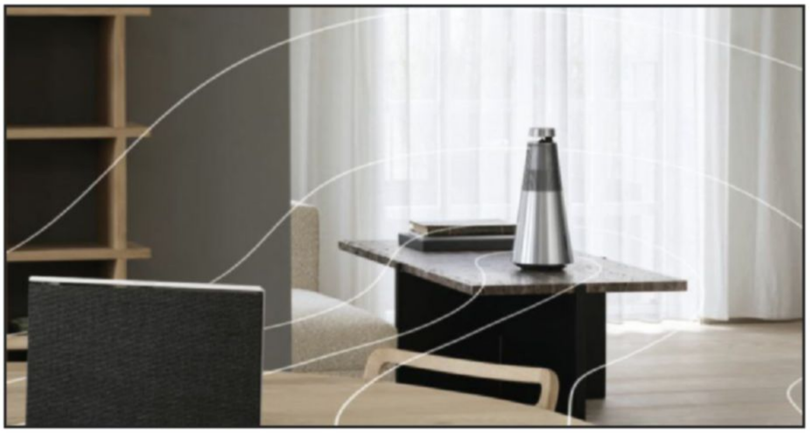

Market Update—Room Correction and Speaker Calibration Technologies

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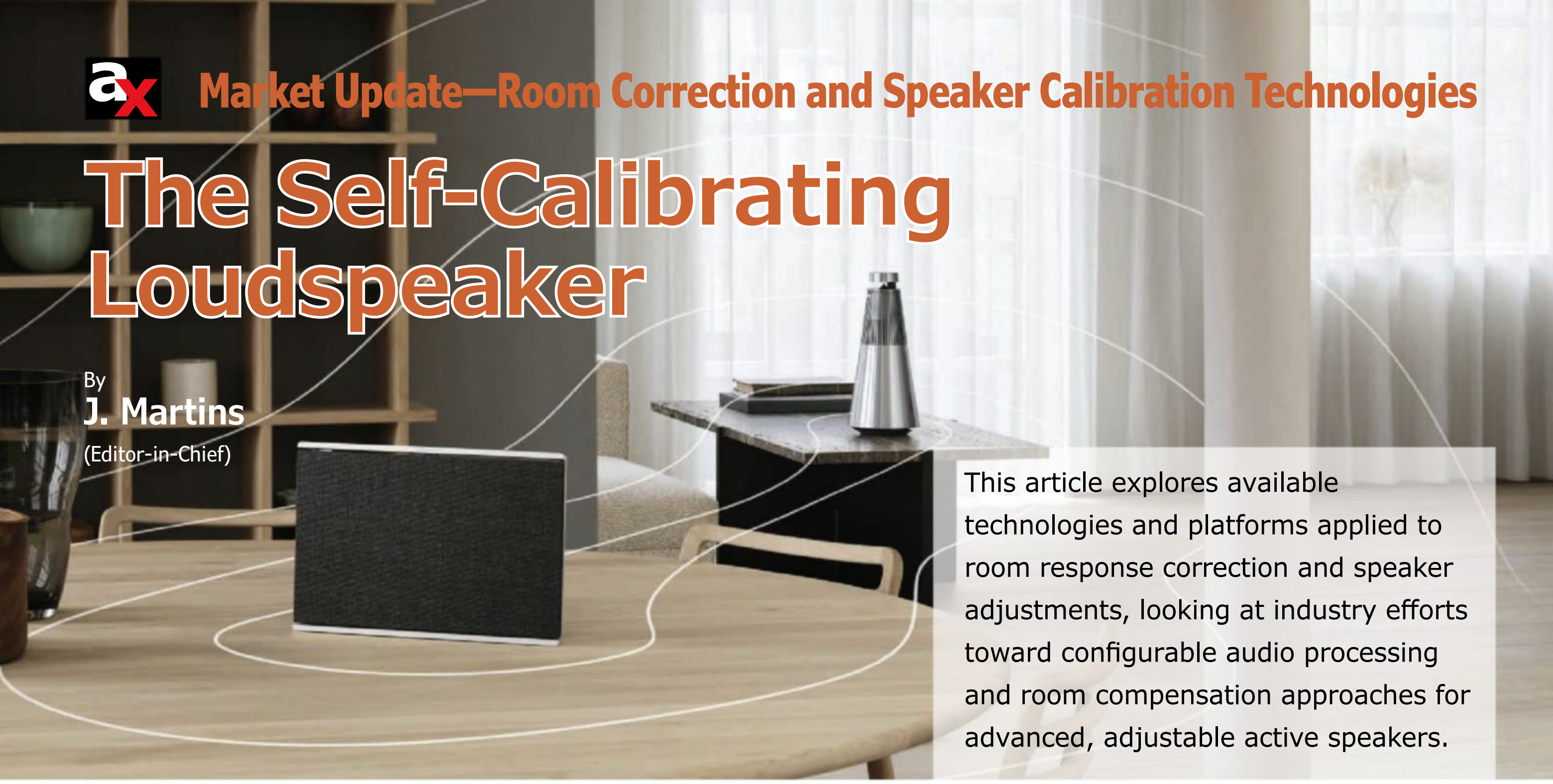
The Self-Calibrating Loudspeaker

By
J. Martins
(Editor-in-Chief)



This article explores available technologies and platforms applied to room response correction and speaker adjustments, looking at industry efforts toward configurable audio processing and room compensation approaches for advanced, adjustable active speakers.

Among the most critical areas of modern audio product "intelligence," exploring built-in DSP to optimize the responses of drivers and amps in connected speakers, continues to be a focus for the industry. The speaker optimization approach is consensually what audio designers mostly associate with truly "smart" audio devices—devices that are not only able to use DSP to optimize the system's response, but are increasingly able to compensate and adjust to the surrounding acoustical conditions or room response.

When discussing emerging applications for digital audio processing and the possibilities of integrated platforms combining speakers, amplifiers, DSP, microphones, and sensors in general, one of the most exciting possibilities is that of the self-calibrating loudspeaker—the ultimate goal for the product category.

In 2017, Apple managed to somewhat realize that vision with the Apple HomePod, designed as a high-quality wireless speaker for the home that shamelessly made use of all the best available technology that the audio industry had been developing. The price of the HomePod was criticized at the time as being outrageously more expensive than a smart speaker from Amazon or Google, when in reality the HomePod was much more affordable than speakers in the same form factor from Sonos, Bose, and many others. And yet, the quality of the HomePod was surprisingly good for its size.

Apart from the impressive high-excursion 4" top woofer and seven-tweeter array, each with its own amplification channel, to maximize the sound quality from such a compact design, Apple integrated "spatial awareness," enabling the HomePod to sense its location in a room and automatically adjust the audio. The design used the same six far-field microphones that were required for Siri voice control and the power of its built-in A8 processor to tune the "direct and reflected audio" from the drivers. This enabled adjusting the response to the room acoustics, its relative position to reflective surfaces, and even optimized the configuration as a stereo pair with two HomePods.

Combining this "automatic room-sensing technology," the HomePod was able to quickly learn its position in a room, and within seconds, optimize the driver's response "to deliver the best possible music listening experience." Apple also used an internal low-frequency calibration microphone for automatic bass correction, with direct and ambient audio beamforming. As Apple described it, "managing all variables through real-time software modeling that ensures the speaker delivers the deepest and cleanest bass possible, with low distortion."

In part, that approach remains implemented in the latest HomePod mini, with the difference that its much smaller size makes the effect less obvious. As Apple details, the HomePod mini uses an Apple S5 chip to enable "computational audio," processing complex algorithms in real time to balance and fine tune the sound at any volume. With lessons learned from the HomePod, the new mini uses its "room-sensing" and "spatial awareness" abilities to maximize the user's perception, adjusting frequency response according to the sound signal (music, voice, and media). To compensate for the reduced driver size, Apple also designed custom force-cancelling passive radiators that give the HomePod mini its impressive bass extension.

Before Apple, Bang & Olufsen had long been exploring the concept, and Sonos had already introduced its own spatially aware tuning technology in 2015. The Sonos Trueplay is a speaker-tuning software first implemented in the impressive Sonos PLAY:5 to improve speaker performance, or the perception of poor speaker performance in comparison with traditional high-fidelity systems. Since Sonos' main focus is in whole-home audio, the approach was very different. Its Trueplay software was designed to improve all its speaker range, including different connected models, allowing listeners to easily tune their existing systems to optimize the sound no matter where the speakers were placed.

The challenge—since Sonos speakers are not designed with the sophisticated microphone arrays, sensors, and processing power that Apple used in the HomePod—was to be able to customize speaker sound without requiring complex and expensive professional calibration.

Sonos designed Trueplay so users can tune speakers with the press of a button, knowing that few homes are acoustically perfect,

The original Apple HomePod had the most complete implementation of "spatial awareness" so far, enabling the speaker to sense its location in a room and automatically adjust the audio.

and that the placement of the speaker in the room impacts the way it sounds. Using the Sonos app combined with the existing microphone(s) on an iPhone or iPad, and a series of special tones emitted by the Sonos speaker, the system is able to analyze how sound reflects off walls, furnishings, glass, and other surfaces in any given room. Sonos then loads an optimized profile for the speaker so the music sounds its best. Instead of requiring calibrated measurement microphones, Trueplay relies on the consistency of Apple's microphones in the iPhone and iPad—and that is why it is iOS-only.

Sonos has been highly praised by this solution and the results it enables in its speakers in difficult rooms. Of course, very few users understand what they are doing while they follow along the instructions to move the iPhone to different locations—and very often people question the whole process and don't do it properly.

More recently, Sonos implemented a similar feature to tune the sound of the speaker automatically in portable speakers (e.g., the Sonos Move or the Sonos Roam), requiring 30 seconds to complete the process. Ideally, that should be the way to go in consumer products, but as we know, things are not that simple, and we still need to step-up on technology integration to make this possible. But we are getting there fast, as smart home integration and AI computation on-device are now making possible what the pioneers in room correction could have never imagine.

For that envisaged evolution to take place, product designers need to start embedding these optimizations directly on the device, analyzing sound sources, and optimizing signals between amp and the speaker drivers.

Much smaller in size, the Apple HomePod Mini is able to adjust its response but focuses much more on the type of sound signals it plays, including voice, and performs as a networked system of multiple units.

And we also need to get away from the concept of "room correction software" that runs on a PC with wires connected to an audio interface and more cables for measurement microphones, and the whole old-fashioned ritual that has characterized those approaches. Neither the simple installation of a dedicated processor running equalization and filters—which are always going to be optimized for a single position—delivers the benefits of optimizing and maximizing the speaker's response for variable conditions and with different format sources, stereo, or multichannel.

There is no reason why the "room correction software" can't run directly on-device these days, reacting in real time to a number of microphone and sensor inputs. Also, because the benefits for consumers are not in obtaining a theoretical "flat frequency response" or reference target curve out of the speakers, but making sure that whatever is being reproduced sounds as closely as possible to its source with minimal detrimental effect from the room conditions.

Compensation and Calibration

Many companies that have previously worked with speaker calibration and room compensation technologies have envisaged that automatic speaker calibration according to the room acoustics would be a possibility one day. Some are starting to converge over that vision.

One of the very first companies to gain recognition in this field was Audyssey Laboratories (https://audyssey.com), a company created in 2002 as a spin-off from the USC Integrated Media Systems Center, the National Science Foundation engineering research center at the University of Southern California. It was founded by Prof. Chris Kyriakakis from the USC Viterbi School of Engineering (now Chief Audio Scientist at Syng, a company created by ex-Apple employees) and Prof. Tomlinson Holman from the USC School of Cinematic Arts (still working with Apple) along with two former USC students and researchers, Sunil Bharitkar (currently working at Samsung) and Philip Hilmes (now heading Amazon Lab126).

The first implementation of Audyssey's technology was released for the CEDIA market and consisted of a dedicated multichannel processor designed for home theater using a Texas Instruments DSP platform. After measuring the room—setup time in 2003 was already less than 10 minutes to obtain the exact room and audio system characteristics—the system addressed the negative effects of room acoustics on sound reproduction.

The company's first approach to room correction, Audyssey MultEQ, basically corrects the effect of room reflections by separating the source signal in the time domain and acting on each channel's frequency response. MultEQ creates a room equalization filter for each speaker and subwoofer in the system.

Since then, Audyssey delivered other solutions to overcome acoustical limitations in audio systems and better match human perception, creating different implementations for consumer, professional, and automotive products.

Other Audyssey tools include Dynamic EQ to compensate for the problem of perceived loudness and deteriorating sound quality as the playback volume is decreased—particularly important in noisy environments (e.g., the car); or Dynamic Volume to manage undesirable variations in volume across program material.

Sonos Trueplay enables users to use optimize the sound of their speakers using the dedicated app and the microphones of iOS devices. Auto Trueplay can also automatically adjust the sound of Sonos speakers. As soon as audio begins playing, the speaker automatically re-tunes the sound, adjusting for the room acoustics.

Audyssey continued to license those technologies successfully to multiple manufacturers, and eventually evolved to apply the solutions in integrated "auto-calibration systems" for TV sets and promoting built-in processing for home theater AVRs with the support of mobile devices and apps. More recently, the company turned its focus to explore those same tools to signal processing of voice capture and recognition applications, including noise-cancellation and dereverberation tools to increase voice recognition accuracy and intelligibility.

Many other companies, including Pioneer and Yamaha, have developed a similar approach to Audyssey's room equalization on the home theater front. This also included Onkyo and Integra, which adopted the AccuEQ solution—temporarily switching from Audyssey. Apart from adjusting the frequency response of the system, the AccuEQ approach promised to eliminate standing waves from a specific listening space but apparently the results didn't impress users. More recently, Onkyo (and sister brands) switched to Dirac Live.

Anthem Room Correction (ARC) is another familiar reference in advanced room correction, offering effective results for high-fidelity applications. The solution, developed by Anthem Electronics and implemented in products from Anthem, MartinLogan, and Paradigm, combines a measurement microphone and software to perform the measurements, calculate corrections, which are uploaded to the audio devices. ARC can be as "automatic" as the user wishes, but also allows a good degree of customizations for advanced correction, ideal for integrators. The software packages, ARC Genesis (for Mac and Windows) and ARC Mobile (for iOS and Android), are free for download.

The latest generation of ARC Genesis software introduced more advanced measurement functions, acoustic correction algorithms, and user-configurable target curve customization. The solution makes it easy to perform corrections in any room and system configuration, up to 7.1.4-channel home theaters. The user positions the calibration microphone as instructed, ARC measures responses and adjusts the equipment performance to suit the space and given conditions, focusing on removing "negative room contributions," and retaining the "favorable acoustic signature" of the speakers. The custom filters generated by Anthem's algorithms are used to program the embedded DSP in compatible audio products.

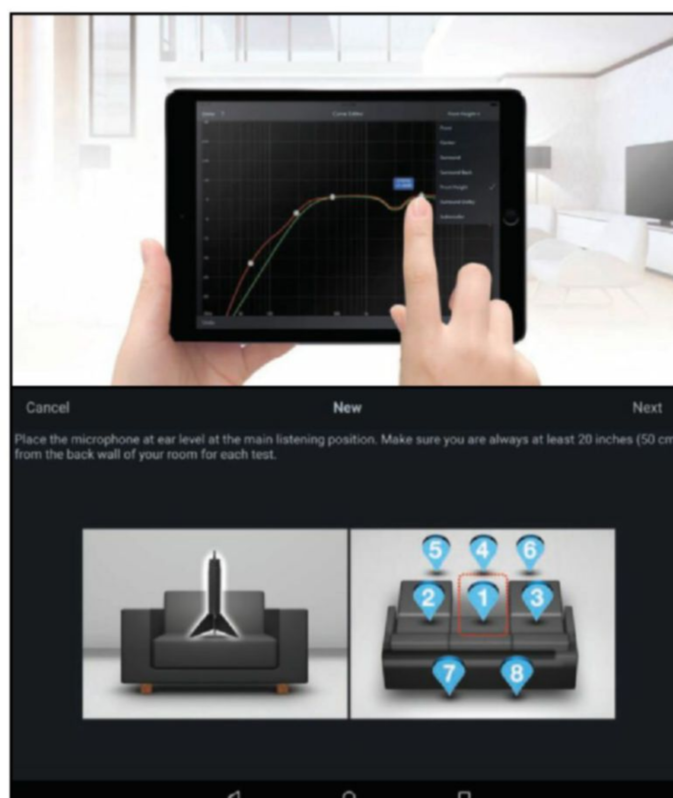
Ideal for most consumers, ARC Genesis' Auto mode asks a few simple questions about a system, runs a full set of measurements, automatically calculates the correction and bass management settings, and uploads them to the ARC-compatible device. The Professional mode offers full control and a granular level of customization, including crossover adjustments, before uploading corrections and bass management settings.

Whatever the approach, the ARC software is always measuring the difference between an ideal lab-standard speaker response, and the actual conditions in a given room.

Taming the Modes

To deal with room modes and reflections, a typical room correction software or system measures the impulse response at multiple positions in order to calculate a combination of correction filters, normally FIR or IIR to reverse the room effects, while maximizing the speaker response with lower distortion. Things get complicated when we start acting in the phase and time-domain, in order to compensate for variable positions and sound propagation at the different frequency bands. As with flat linear frequency response, time alignment at all frequencies remains an elusive target where any changes immediately originate an effect elsewhere, sometimes worse than the problem we are trying to compensate. And that is often the case with room modes or standing waves.

Without interfering directly with the reproduced sound from the loudspeakers, we need to enter the realm of active acoustics, which implies applying convolution algorithms to the measured



Audyssey MultEQ established the room correction as an integrated solution in AVRs and pre-processors (there is also a DAW plug-in for studios), addressing acoustical problems measured in the time domain. As a fully-integrated solution it ships with its own microphone, calibrated to a 1/4" industry-standard measurement microphone.

sound and generating a reversed-mirror signal of what we want to cancel, or compensate acoustic signals—an approach normally effective over room modes.

There are products in the market that take that approach, and it is an area that promises to greatly expand in the emerging field of virtual acoustics (a topic for another article).

A simple active solution for small rooms and studio environments exists from Swiss company PSI Audio, in the form of the Active Velocity Acoustic Absorber (AVAA). Inside what seems to be a standard trapezoidal speaker, a microphone measures the acoustic pressure in front of an acoustic resistance. The acoustic resistance is designed to let air through but significantly reduces the pressure. Behind the acoustic resistance, a transducer membrane is driven to absorb the volume of air going through the acoustic resistance as well as ensuring a specific acoustic impedance in front of this acoustic resistance. As PSI details, when in function, this acoustic impedance in front of the acoustic resistance is significantly lower than in ambient air and therefore acts as a pressure sink. The acoustic impedance of the air is affected typically over a radius of 1m to 1.5m around the AVAA. This explains how the AVAA can be more absorbent that its actual surface of perfect absorber.

An effective active "bass trap," the AVAA has the advantage of being easy to set up and move, no calibration is required, and it is able to take care of any room modes that stubbornly remain even after basic acoustic treatment is applied, between 15Hz and 150Hz. Positioning two AVAAs makes a significant difference in normal-sized rooms and the concept shows that a combination of actual speakers and microphones, permanently active, is an effective way to address acoustic problems.

The Anthem Room Correction (ARC) helped establish sound correction at home, being supplied with a microphone and software, with corrections uploaded to the Anthem Electronics or Paradigm processors.

For the past two decades, multiple companies have also worked to perfect the speaker calibration process taking into account the room response. Harman International Industries did extensive work on that front and owns patents in this domain.

A specific example of a Harman patent submitted originally in 2006 (US8577048B2) describes systems and methods for calibrating a loudspeaker with a connected microphone. "The loudspeaker includes self-calibration functions to adjust speaker characteristics according to effects generated by operating the loudspeaker in the room. In one example, the microphone picks up a test signal generated by the loudspeaker and the loudspeaker uses the test signal to determine the loudspeaker frequency response. The frequency response is analyzed below a selected low-frequency value for a room mode. The loudspeaker generates parameters for a digital filter to compensate for the room modes. In another example, the loudspeaker may be networked with other speakers to perform calibration functions on all of the loudspeakers in the network."

This very detailed and extensive patent—valid until 2029—basically covers what the whole industry is currently attempting, which is to move room compensation DSP to the speaker and fully integrate speakers as "self-calibrating." And yet, one of the critical missing areas that needs perfecting is the "room-sensing" part.

RoomPerfect Measurements

The companies that are targeting the very high-end of audio reproduction and even cost-is-no-object systems, understand how challenging the room correction approach is. Some are even determined to push it as far as it can go.

One of the companies determined to perfect the results for its own high-end audio products, in any room, is Lyngdorf Audio (and its luxury branded extension Steinway Lyngdorf). Peter Lyngdorf, the company's founder stated, "At Lyngdorf, we recognized that just focusing on creating processors and amplifiers with a linearity of ±0.2 dB makes no sense when the music is being played in a room with a linearity of perhaps ±10dB.

"Our products are designed to make speakers perform perfectly in rooms full of furniture, curtains, and bookshelves, because we create products for use in everyday life."

To pursue that goal, Lyngdorf explores the full potential of digital signal processing in its processors and amplifiers and all its products are able to load filters for room correction. But earlier on the company realized that taking multiple measurements in the room with a microphone introduces many variables, requiring a long effort to fine-tune and optimize. Lyngdorf did the research to identify the basic errors in this approach: "You don´t get sufficient data through a single-point measurement, and it is very hard to evaluate which problems can be corrected and which cannot. The acoustical problems of a room are in all three dimensions." Lyngdorf also added, "the perfect one-size-fits-all target curve does not exist."

The company's research focuses on room correction technology covering the full frequency range, adjustable to all types of speaker and subwoofer setups, and optimizing the impulse response to

allow for the most dynamic music reproduction. For this approach, which involves several patented technologies, the company created a fully integrated and user-friendly solution called RoomPerfect, which doesn't need an external computer.

The RoomPerfect calibration process is fully guided instead through the display of the Lyngdorf processor/amplifier, and all the equipment required for calibration is included with the system. The calibration process is based on a single measurement, taken at the listening position to instantly capture all the specific speaker configuration properties, followed by a large number of random measurements taken throughout the room to determine room modes and overall room response.

To create an effective solution for RoomPerfect, Lyngdorf also perfected the use of specific tones, designed a dedicated microphone, and designed a progressive system of measurement and evaluation for the multiple tones. After each measurement, the system evaluates the information received and decides if it needs more information to continue, or stores the current values. The company's patent covers the whole measurement method and approach, providing a different level of performance and consistency Besides the level of minutia and control, Peter Lyngdorf believes the RoomPerfect approach can be adapted also for use in affordable stereo systems.

As an active system, PSI Audio's AVAA is an efficient solution for room mode problems. It can be turned ON and OFF, and can easily be deployed into a different room when required.

Perfecting the Science

The same challenges that Lyngdorf identified are being approached in multiple ways in different market verticals. We already discussed the consumer level-approach, which is being attempted by companies such as Apple and Sonos. And we mentioned the room correction software solutions that have helped popularize the concept, which were mainly designed to deal with the complexities of home theater, and gradually evolved to fine tune high-end, high-fidelity home audio systems. The digital signal processing required for room correction has also been taken to new heights in the installation and residential integration sectors, where solutions from Dirac or Trinnov are highly recognized.

French company Trinnov Audio has been taking room correction technology very seriously since its foundation in 2003, and its five main inventions are protected by more than 50 international patents. The company resulted from fundamental research on high spatial resolution audio that was conducted at the IRCAM institute in Paris, and its focus has been on creating very high-quality solutions optimized for immersive audio. Because of this focus, Trinnov understands the challenge of creating ideal conditions for content enjoyment—at the commercial, professional or home levels—requiring a complete approach, starting with traditional acoustics. And being familiar with production environments, Trinnov understands how digital room correction can also help create the required consistency from production to reproduction.

And with the introduction of object-based, spatial audio content such as Dolby Atmos, that requirement was reinforced. "As you increase the number of speakers in your room, you increase the amount of acoustic interactions and therefore problems. This makes Loudspeaker/Room Optimization even more critical to create a realistic and precise tridimensional sound bubble."

Trinnov took a very well-integrated approach toward optimization of the larger number of speakers, by using semi-automatic and advanced systems to achieve the required accuracy. For its room correction and loudspeaker optimization systems Trinnov developed the multiple components required, starting with a unique, four-element 3D calibration microphone, which measures and identifies the location of all speakers in the room in three dimensions. This microphone is the sensing extension of the loudspeaker/room Optimizer software, which in turn is powered by a unique hardware platform, built around a multicore Intel processor and the TrinnovOS 64-bit operating system, which also adds support for scaling computing power and software upgrades.

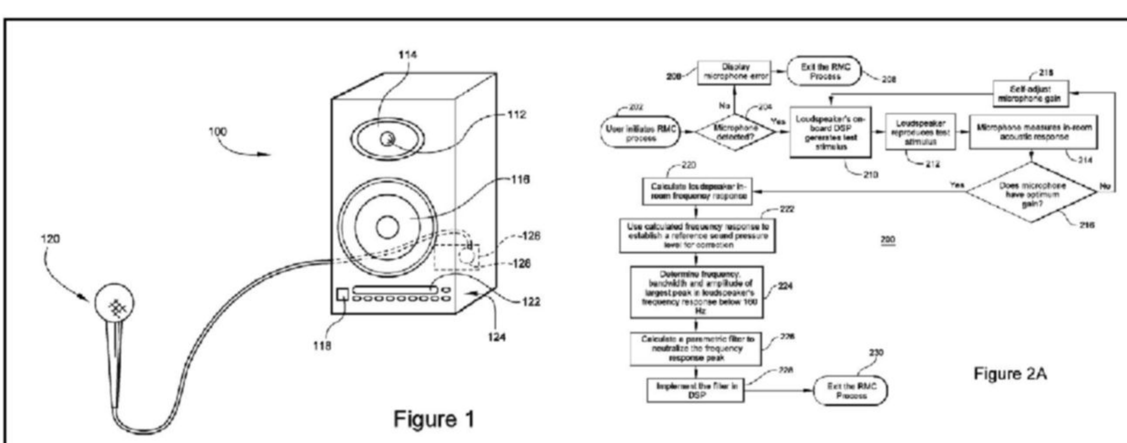


Figure 1 and Figure 2A of the extensive US8577048B2 patent awarded to Harman International Industries, describing a self-calibrating loudspeaker system.

Market Update—Room Correction and Speaker Calibration Technologies

Since the company's inception, Trinnov developed its own loudspeaker/room optimization technology, helping redefine audio calibration standards across multiple applications in the professional audio industry as well as in consumer environments. The Optimizer was primarily intended for studios and the company built its reputation in broadcast, music, and post-production applications. Gradually, the Optimizer was adopted also in premium commercial and private cinemas. Currently, Trinnov's Optimizer is used in thousands of installations worldwide.

Lyngdorf RoomPerfect explores the full potential of digital signal processing in the company's processors and amplifiers and all its products are able to load filters for room correction.

Extremely detailed and accurate, Trinnov's Optimizer technology was primarily intended for sound engineers and evolved into other applications where the company's sophisticated algorithms proved very effective, such as high-end home theaters.

The unique Trinnov 3D Measurement Microphone was developed to increase the accuracy and accelerate acoustic measurements, using four calibrated microphone capsules in a tetrahedral arrangement. This allows feeding timing information from a single sweep and helping to create a tri-dimensional acoustic mapping.

The software addresses the frequency and time domains separately, allows semi-automatic optimizations, but also supports detailed adjustments when required. It measures the direct sound of the speaker, the early reflected sound, and the overall energy response in the room, and addresses the time domain by correcting impulse response, phase, and group delay. Optimizations can be applied for a stereo pair, or to all channels of a multichannel audio system.

The Optimizer relies on Trinnov's unique microphone to feed tridimensional measurements, simultaneously feeding more reliable information but also simplifying the process. The microphone uses four elements in a precise tetrahedral arrangement. Each capsule is individually calibrated at the factory (±0.1 dB from 20Hz to 24kHz), and directly connected to the Trinnov processors, keeping the entire signal path within a fully integrated system for accuracy.

Trinnov offers dedicated entry-level processors for the Optimizer, with the ST2-HiFi designed to be inserted between preamplifier and power amplifiers in stereo systems, while the ST2-Pro version, adopted by many studios around the world, supports up to four simultaneous processing channels.

Trinnov Altitude processors are designed according to demanding professional audio specifications and recognized as a premium solution for high-end home cinema and immersive audio installations, while the Amethyst preamplifier is Trinnov's flagship high-end stereo product. On a single chassis, it combines an audiophile preamp, integrating the Optimizer advanced speaker/room correction with a simplified calibration wizard based in presets for different settings. "Instead of repeatedly integrating third-party, short-lifespan technologies into new hardware products, Trinnov opted for true innovation, upgradeability, and longevity," the company states.

There is no question that Trinnov created a flexible architecture where decoding, processing, and operating functions are all handled as elegantly as possible, with benefits well-demonstrated by the company's ability to constantly support new formats, design new features, and update the software with improvements, some of which are free. Eventually, if and when the company faces the need to streamline its approach to enter new product segments, it can simply port its technology and experience to new processing platforms. If it so decides.

The amount of DSP power required for these processors to handle the tasks required for speaker and room optimization is one of the challenges for the manufacturers that traditionally took the dedicated processor route. Adopting a specific platform, and facing the need to migrate its code to new processors is a critical decision for these companies.

An example that comes to mind is that of DEQX, the well-respected Australian DSP company that pioneered an integrated approach at digital loudspeaker correction, crossover, and room compensation technology for studio and high-end audio applications. The company's founders were pioneers in the concept of low-latency impulse response correction for speakers (and the first to offer remote online speaker analysis, advice and installation support).

The processing algorithms in the DEQX-Cal software help deal with room reflections, adjust phase and timing errors at different

frequencies, adjust crossover frequencies and frequency response to the subwoofer level and finally help detect and minimize room modes.

Now, the company is developing a completely new hardware platform with the new DEQX Gen4 speaker calibration and room correction processors to be available in 2022. As the company explains, using powerful hexacore ARM processors, ESS Sabre Pro ADCs and DACs, and an AMOLED touchscreen, the three new models will "feature an order-of-magnitude improvement in resolution, all-new digital and analog hardware and unparalleled connectivity." More importantly, the DEQX Gen4 processors will interface over the network to the DEQX Cloud, creating "cross-platform, zero-install speaker calibration and room correction," as the company describes.

The Software Approach

Other companies, such as Dirac, the Swedish digital signal processing and optimization company founded in 2001, have been evolving from this same type of exhaustive and detailed procedure—practiced and praised by the many users who have adopted Dirac Live and the manufacturers who integrated Dirac Live Room Correction for home theater and high-end audio applications.

Dirac Live is a patented room correction technology that not only corrects the frequency response, but also the impulse response of the loudspeakers in a room. In terms of its processing algorithms, Dirac Live is considered the most advanced room correction technology available on the market, and is unique in that it provides true impulse response correction over a large listening area, improving the depth, the positioning and distinction of individual voices and instruments. Using multiple measurements and mixed phase correction, Dirac Live helps audio systems to create a natural, realistic and transparent sound with tighter bass and reduced room modes, in a way previously not possible.

The software itself is powerful enough to satisfy the needs of integrators, but can also be implemented in modern consumer AV receivers (AVR), with the possibility to deliver more complex optimizations from the user's measurements through cloud processing, in systems connected to the Internet. In this way, the computational power is almost unlimited and not restricted by the chipset that a manufacturer designed into their equipment. Dirac offers specific

The result of the research in digital signal processing initially conducted at the Uppsala University in Sweden, Dirac established Dirac Live using patented room correction technology, measuring and processing the frequency response and the impulse response of the loudspeakers in a room.

Australian audio signal processing company DEQX is developing a completely new ARM-based hardware platform for its new DEQX Gen4 speaker calibration and room correction processors, scheduled to be available in 2022. The image shows a preview of the front panel.

Dirac Room Calibration OEM packages for these implementations.

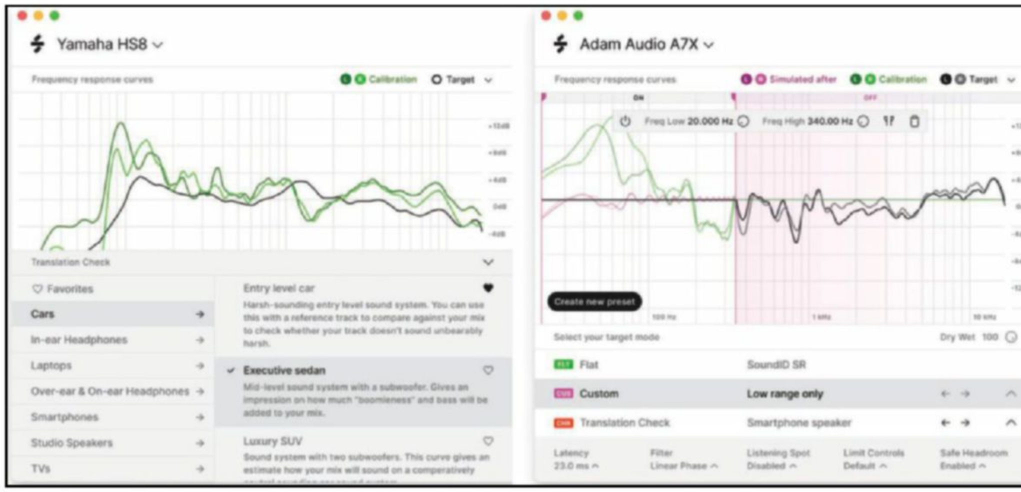
In 2017, Dirac announced Mobile Dirac Live—an end-user version of its integrator-based Dirac Live room correction solution—as an app for iOS and Android devices. This enabled the same benefits of the installer version of the Dirac Live platform for more affordable home theater AVR products, configured directly by the end-user.

A wizard assistant provides the user with step-by-step instruction and feedback throughout the calibration process. The Dirac Live audio filters and algorithms are embedded in the AVR products, and are simply configured remotely, enabling also users to activate different licenses (e.g., evolving from stereo to 5.1 or more channels) and new features. The latest feature available is Dirac Live Bass Control, which measures and phase corrects both the speakers and subwoofers across all frequencies to produce enhanced bass clarity and tone.

This level of integration is available on multiple home theater systems and in a range of products from a series of leading AVR manufacturers. Harman, Sound United, Rotel, Onkyo, Pioneer, Pioneer Elite, Integra, and Bryston all are now offering this level of Dirac Live integration even in some of the most affordable receivers and integrated amplifiers.

The market recognition led Dirac to expand to other markets, including calibration for studio systems, where things have become quite active in the last few years. Creating different front-ends for its Dirac Live software, the Swedish company also brought the same technology to consumers who listen to music streamed simply from a tablet or computer.

Part of its new 360 Reality Audio range of wireless speakers, Sony's SRS-RA5000 was designed to enhance spatial audio from multiple drivers, filling a room with sound both vertically and horizontally. To optimize its response, the RA5000 features Sound Calibration. Users need to hold the Immersive Audio Enhancement button on the RA5000 and the speakers conduct a detailed sound calibration adjustment, including loudness, for optimum audio performance for the room in which it is placed.



With the the 2020 version of SoundID Reference, Sonarworks introduced the ability to make custom adjustments to a frequency target curve and simulate different sets of studio monitors and speakers with "Translation Check".

The Dirac Live Room Correction Suite software consists of a tool to take measurements and design digital filters, and a processor that stores the filters and processes the audio signal. Studio users can choose to run Dirac Live as a DAW plug-in (VST2, VST3, AU, or AAX format) or use the Standalone version that processes all audio applications. There's also a choice between stereo and multichannel licenses that can be activated.

Users simply need to connect an omnidirectional microphone (the system accepts calibration files to be uploaded) to the computer, follow the instructions to make the acoustic measurements, and the room correction filters are created in Dirac's cloud server. The user is always able to adjust the target frequency response curve if needed and the procedure can be repeated anytime.

The company's approach has also been to leverage the fundamental processing technology to create software that is able to run in multiple platforms, enabling it to adjust fairly quickly to new market opportunities.

Available for audio manufacturers and OEMs, Dirac Room Calibration is now being adopted for consumer audio products such as Bluetooth and Wi-Fi speakers and soundbars. The principle remaining that users can activate Dirac's room correction technology directly by running the app on Android and iOS devices. Like Sonos already does, the microphones on those same mobile devices can be used to feed room sweeps and impulses that can be combined with measurements performed directly by microphones built into the speaker being optimized.

The full potential of "room-sensing" and "spatial awareness" is being pushed in multiple consumer categories, and Dirac is adapting its technologies and suite of tools to optimize digital audio and perfect sound for better listening in any environment. Current focus for Dirac includes embedded platforms for automotive, computers and tablets, headphones, and even mobile devices. In automotive audio, Dirac can be applied to optimize the acoustic environment in the cabin, including at different occupation levels. And learning to deal with the challenges of vehicle cabins on the road is probably a great way to perfect the kind of integration that one day could be present in every single speaker, home, portable, or PA.

dSONIQ Realphones software re-creates the acoustic environment of a real recording studio control room in headphones and corrects the headphones' frequency response for accuracy.

From Target Curves to Response Emulation

Since this Market Update is mainly focused on the installation and home applications, we had to leave out studio reference and calibration applications. We will briefly address some examples, simply as a way to illustrate how these solutions are paving the way for a completely new approach—already a market trend in music production and home studios.

In 2020, Dirac also decided to expand the application of its pioneering room correction solution, offering its patented impulse response and frequency response correction to help tackle the accuracy of sound reproduction in studio environments. Dirac Live is now even available with room correction for multichannel and surround sound studios in various configurations.

Dirac was motivated to enter this space by Latvia-based Sonarworks, a company that completely disrupted the room correction market with its Reference monitoring calibration software, introduced earlier in 2015 for headphone calibration. In 2018, Sonarworks even proposed a digital sound reference standard response curve called Sonarworks SR (Studio Reference), attempting to deliver the same accurate studio reference sound on speakers and headphones.

The latest version of the Sonarworks software, now called SoundID Reference, continues to deliver a simple and affordable approach for reference sound, and introduced the ability to make custom adjustments to the target curve in real time or even define a Custom Target.

But the most "intriguing" new feature was the introduction of a "Translation Check" feature, realistically simulating a plethora of different devices and device classes—which means emulating both speakers and particular room acoustics. This enables users to check their mix in different listening environments without having to leave their studio chair, bedroom, airplane seat, or hotel room—typically via headphones.

This, in turn, is leading the home studio market toward a completely new direction: If you can adjust the speaker's response to the room acoustics, and now you can simulate any listening environment...then it doesn't matter what speakers or headphones you use.

Multiple companies have already explored this approach for headphones. Just recently, California studio monitor company Fluid Audio has ventured into headphones and announced the release of its new Fluid Audio Focus headphone playback and mixing system. To combine the headphones with correction and processing software, Fluid joined forces with Russian software house dSONIQ, and bundled its new Focus headphones with the company's Realphones headphone correction and binaural room simulation solution. This is becoming a serious trend in the studio world.

More recently, multiple companies have started to add binaural simulation of real room responses, enabling musicians and music producers to enjoy a realistic emulation of being in an actual studio environment while wearing headphones. Waves Audio, Embody, Scaeva Technologies, and Slate Audio with its VSX Headphone and Modeling System, are among the notable companies that have entered the space.

dSONIQ's approach was to combine both techniques, and the dSONIQ Realphones software re-creates the acoustic environment of a real recording studio control room in headphones, and corrects the headphones frequency response for accuracy. This enables a standard headphones design with decent frequency response and dynamics to turn into a reliable monitoring tool that can be used even with a laptop in a hotel room. And there's no way this approach will be restricted to binaural reproduction through headphones.

Another pioneer in acoustic correction software for home studios, IK Multimedia offers a complete solution to correct poor room acoustics and improve the audio monitoring accuracy of studios with its ARC System 3. The solution, which combines an ultra-accurate measurement microphone, room analysis software, and a correction plug-in, is a simple-to-use package that delivers professional-level results for home studios. With its 2020 update, IK Multimedia introduced an all-new analysis algorithm, Volumetric Response Modeling (VRM), which measures acoustics at three different heights around the listening position to deliver superior accuracy, and also allows time domain adjustments through Natural and Linear phase modes.

The novelty in the latest version is the virtual monitoring feature that lets users intentionally shape their sounds to emulate devices (e.g., TVs, smartphones, and car audio systems, as well as many popular studio monitors), to quickly and conveniently listen to the mix on other "virtual systems" to ensure sound translation across common listening devices. And naturally, IK Multimedia recommends that home studio users add its own iLoud MTM high-resolution compact studio monitors, to create a fully integrated, consistent emulation solution. These monitors feature built-in DSP and Physical Response Linearization (PRL) to correct the system's performance in real time and, of course, are bundled with the ARC software and reference microphone for built-in acoustic self-calibration.

Users can just set up these compact monitors in an hotel room, put the microphone in the listening position, press a button, and automatically adjust the frequency response to its placement and obtain a custom calibration for the environment. Next, the user can choose custom response curves that emulate devices, allowing it to audition its work to ensure sound translates perfectly.

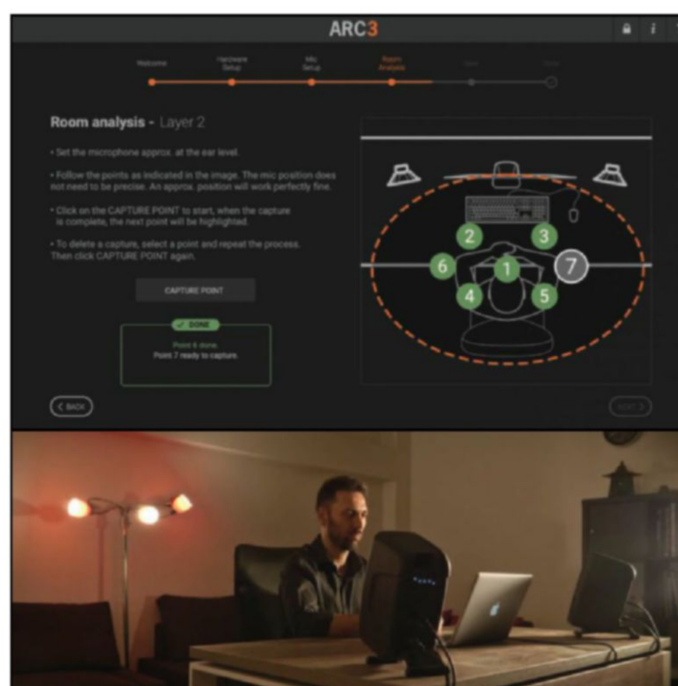
Where Next?

Without yet jumping into the tempting concept of acoustics emulation, which is motivating major investments by many audio companies, software companies in the pro audio studio market have been already selling the "room simulation" concept as a plug-in.

Latency with always be an issue when applying these room emulations from a computer, but what's stopping anyone from embedding something like this on dedicated hardware?

After all, Abbey Road Studios has partnered with Bowers & Wilkins to bring the Studios' unique acoustic personality to the in-car listening experience. These automotive applications, also explored successfully by Dirac, typically use low-power SoCs combining dedicated DSP and ARM processors. And there's probably enough power in any of the microcontrollers being used in the next generation of smart-speakers to run the required convolution filters.

A real demonstration that this is possible is available in the latest



IK Multimedia offers a complete solution to improve the audio monitoring accuracy with its ARC System 3 software and calibration system. Users can just set up the iLoud MTM compact monitors, put the included microphone in the listening position, and obtain a custom calibration for the environment. Next, the user can choose custom response curves that emulate other target devices to ensure sound translates perfectly.

Market Update—Room Correction and Speaker Calibration Technologies

The Merging+Anubis audio interface and controller is able to run Sonarworks' SoundID Reference correction software without a computer, with the lowest possible latency, right out of the box.

The latest Bang & Olufsen Beosound Level is a portable wireless home speaker that is able to adjust its sound to compensate for any position or room. The design features next generation active room compensation using the built-in microphones and sensors to intelligently analyze the behavior and continuously adjust speaker's response. The speaker can easily be carried around the home and even taken into the garden thanks to its wireless technologies and IP54 dust and splash water resistant rating. The audio architecture automatically adjusts its acoustic tuning from 180-degrees to 360-degrees based on the way it is positioned, optimizing any listening experience.

integration effort between Merging Technologies and Sonarworks. Thanks to its serious on-board DSP/FPGA processing capabilities, the Merging+Anubis audio interface and controller is now able to run Sonarworks' SoundID Reference correction software without a computer, with the lowest possible latency, right out of the box. The goal for Sonarworks and Merging Technologies was to simplify the music production process with the first hardware able to run the popular calibration software outside of a computer, providing a practical solution for music creators and engineers to apply accurate sound correction at the lowest possible latency on multiple outputs simultaneously. Users of this compact AoIP solution can get immediate headphone and room correction for studio monitors, anywhere and anytime. The Anubis integration supports different SoundID Reference correction curves to multiple sets of connected speakers, recalling different room profiles for a specific speaker set (e.g., flat or XCurve) at the touch of a button. And SoundID Reference even offers its various speakers, headphones and listening environment emulations, which users can switch at the flip of a button.

The difference between these production environments and consumer applications lies in the fact that while audio engineers want to be able to tweak every parameter, consumers want things to be automatically optimized at the press of a button. And that is where the use of AI on smart devices, constantly measuring the environment and providing real-time adjustments makes sense.

As we've seen from the multiple "room-sensing," and "spatial awareness" approaches taken for speaker optimizations, the fundamental challenge toward the self-calibrating loudspeaker lies in connecting the sound sources, the speakers and the room itself. That is precisely what is being done with variable acoustics and the design of performance spaces using systems such as Meyer Sound Constellation. Using an array of ambient sensing microphones, sophisticated digital signal processing, and loudspeakers, Constellation modifies the reverberant characteristics of a venue and redistributes sound throughout the space, ensuring a natural acoustic experience in every seat.

A similar, streamlined approach will be viable in the smart home, where distributed "invisible" speaker panels and microphones in the walls and ceiling are able to completely redefine the acoustics, while the connected smart speakers used for music, movies, or any other content reproduction are fully optimized for the ultimate listening environment at all times.

While self-calibrating loudspeakers will continue to be perfected and make sense for mobile and portable applications, at home, the whole acoustic environment and the devices within can be fully integrated and optimized in the smart home.

Resources

Anthem Electronics | www.anthemarc.com

Audyssey Laboratories | www.audyssey.com

DEQX | www.deqx.com

Dirac Research | www.dirac.com

dSONIQ Realphones | www.dsoniq.com

IK Multimedia | www.ikmultimedia.com

PSI Audio | www.psiaudio.swiss/avaa-c20-active-bass-trap

Sonarworks | www.sonarworks.com

Steynway Lyngdorf (RoomPerfect)
https://steinwaylyngdorf.com/roomperfect

Trinnov | www.trinnov-audio.com

Introducing GLM 4.1 loudspeaker manager software

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15 years

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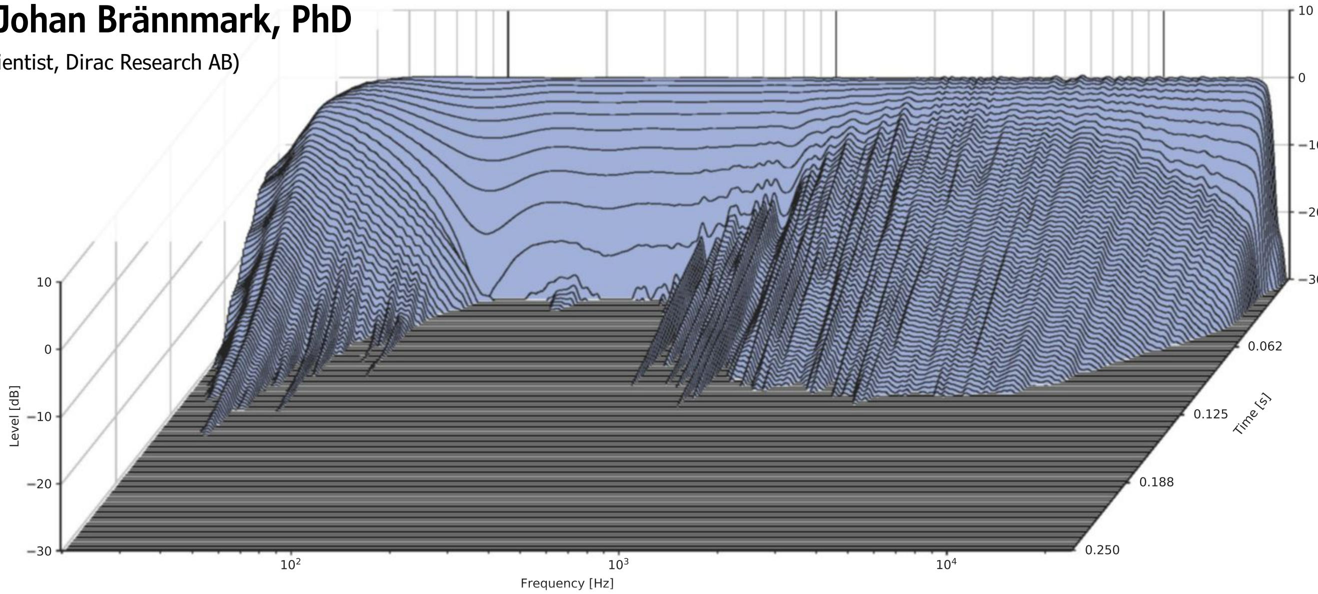
The Arrival of Spatial Room Correction Technology

By
Lars-Johan Brännmark, PhD
(Chief Scientist, Dirac Research AB)



This article describes an approach to the challenge of applying DSP room correction to complex multichannel setups, intended for immersive audio reproduction. Instead of one filter applied to a single channel, this Swedish company proposes a solution where all speakers work to optimize the reproduction of each input channel.

Over the past 20 years or so, room-correction technology has slowly but steadily made its way into the mainstream of high-quality home audio. Today, many leading audio/video receivers (AVR) come with built-in, DSP-based room-correction solutions. These solutions allow consumers to improve sound system performance in a given space, compensating for a room's unique characteristics.

Commonly, room-correction systems use a microphone to measure each loudspeaker and generate a filter to correct the speaker/room responses on a per-channel basis. In this way, the time and frequency properties of a channel are optimized independently of other channels in the system.

But home audio is evolving, and room correction must evolve, too. Consumers, content producers, and AVR manufacturers have begun to embrace immersive audio and multichannel formats for all manner of home entertainment, much of which is meant to be enjoyed by multiple people, positioned in multiple locations within a dynamic space. Established methods of room correction may not be sufficient under these circumstances.

With current room-correction systems, a filter can shape a signal in time and frequency, but the spatial response—the way the sound propagates and interacts with objects and room boundaries; the way sound pressure at one position relates to pressure at other positions—remains unaltered. In light of modern, immersive, multichannel home audio experiences, we need room correction to be spatially robust. That is, the system should be able to improve sound performance throughout a spatial distribution of listener positions, rather than at a single point in space. The more complex the room response is (e.g., if the frequency response varies very rapidly with small changes in listener position), the less improvement can be expected from a standard room-correction filter.

In short, a three-dimensional sound field cannot be re-shaped in space through traditional room-correction methods, with one filter applied

to a single channel. The solution is loudspeaker co-optimization. All the speakers in an audio system working jointly to reproduce each input channel in an optimal way.

The Possibility of "Spatial" Room Correction

When we think about it, spatial room correction, which is currently out of reach for single-channel techniques, would be possible if multiple speakers were allowed to cooperate (i.e., if room-correction filters operated simultaneously on several speakers located around a room).

To some extent, this is what happens in a few high-end bass management solutions: Multiple bass-capable speakers, typically subwoofers, can be equalized and phase-shifted with regard to their in-room sum response, thereby reducing seat-to-seat frequency response variations in the common bass channel. Such bass management solutions bring multichannel equalization and phase adjustment to bass speakers.

This is a well-accepted and useful solution, although limited. True spatial room correction applies not only to optimizing a common bass channel, but to all input channels and to a wider range of frequencies.

So, what can we do with multiple channels to stretch a sound system spatially and reach its full potential? A lot, it turns out.

During the 20-plus years of room correction development, the DSP and acoustics research communities have come up with a lot of theories about, and methods for achieving, sound field control. In various ways, such methods are based on multichannel (MIMO) filtering and the physical principles of sound field superposition. However, apart from a few beam-steering and binaural rendering applications (e.g., for soundbars, laptops, PCs, and TV sets), most of this research has yet to be productized for broader consumer markets.

In part, this technology lag is due to system complexity and sensitivity. Combining sound from separate channels certainly creates new possibilities, but it puts higher demand on system reliability and robustness (e.g., manufacturing tolerances and long-term stability of electro-mechanical components). To have the desired effect, the sound fields of loudspeakers, plus room reflections, must add up in exactly the right constructive/destructive way at multiple points in space and at exactly the right time.

Loudspeaker co-optimization and spatial room correction can achieve the right effect. What's more, through broadband sound field control, users can take control of the full sound wave in time, frequency, and space. They can control not only the way it propagates from the speaker through the room and to the listener, but also its interactions with the room boundaries, enabling all speakers in the system to work jointly to reproduce each input channel in an optimal way.

"Support" Speakers and "Super" Speakers

As we know, maintaining control over sound quality, channel levels, and room/speaker interaction becomes more difficult the more speakers are added to a system. But with a co-optimization correction scheme, the diversity in a system can actually be exploited and become a benefit rather than a nuisance.

With loudspeaker co-optimization and spatial room correction, each speaker in an immersive, multichannel sound system may simultaneously assume two roles: first, as a primary speaker to be corrected; and second, as a "support" speaker used to correct one or several of the other speakers in the room. Each speaker in the room is helped by all or a subset of the other speakers to reach its target response.

The aim is to get the best possible impulse response from each speaker, taking other available speakers into account as support speakers and creating a virtual "super" speaker. All speakers are used to reproduce each input channel. With a number of speakers placed around the listener, as in typical surround-system layouts, this co-optimization approach allows for the cancellation of reflections and the control of room resonances in ways that are not possible with traditional means.

In practice, co-optimization allows for not only performing impulse response correction of a speaker's direct sound, but also for enlisting additional speakers that can be used to correct for reflections that would otherwise not be possible.

About the Author

Dr. Lars-Johan Brännmark has served for nearly 20 years as an inventor and algorithm designer at Dirac Research AB in Uppsala, Sweden. He is currently Research Fellow and Chief Scientist in the same company, with particular expertise in room correction and sound field control. He holds a Diploma in Sound Engineering from Piteå School of Music, Luleå University of Technology and he has also studied Electrical Engineering at Purdue University and Musicology at Uppsala University. Before joining Dirac, Lars-Johan worked as a sound engineer at the Swedish National Radio. He holds a PhD in signal processing from Uppsala University.

For example, speakers located close to a reflecting surface are better suited than the source speaker to cancel out reflections from that direction.

In the case of stereo, traditionally each input channel is reproduced by one of the front speakers. If each of these front speakers work on their own, the only way to handle room modes is to add or remove power, however, this will only be valid for a very small region of the listening room and the speaker may not be able to handle the increased power. Using multichannel control, additional speakers are used to add and remove power in different positions—and with different timing—simultaneously. This allows, for instance, a more even bass reproduction, with better timing, in larger listening areas.

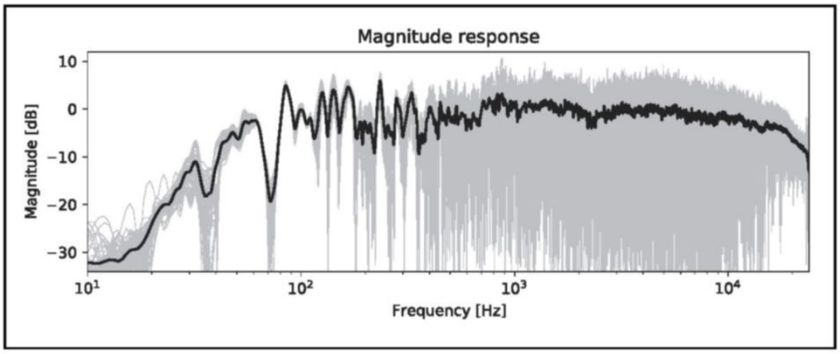


Figure 1: The measured responses of an uncorrected system, in 64 microphone positions (grey lines) and their average (black line).

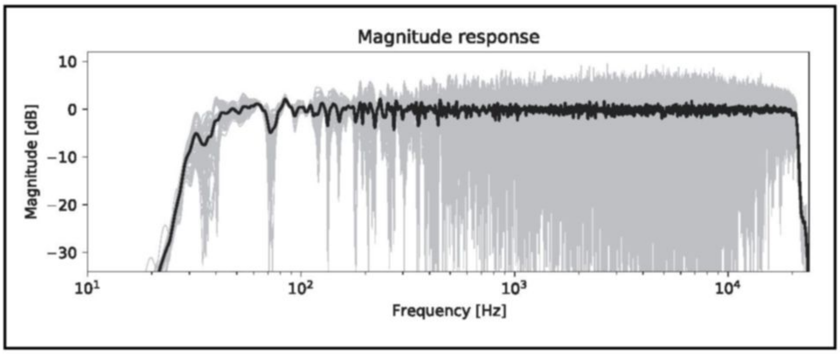


Figure 2: The measured response of a system using room correction based on single-channel technology. The average (black) is improved but spatial variations (grey) over different measurement positions remain.

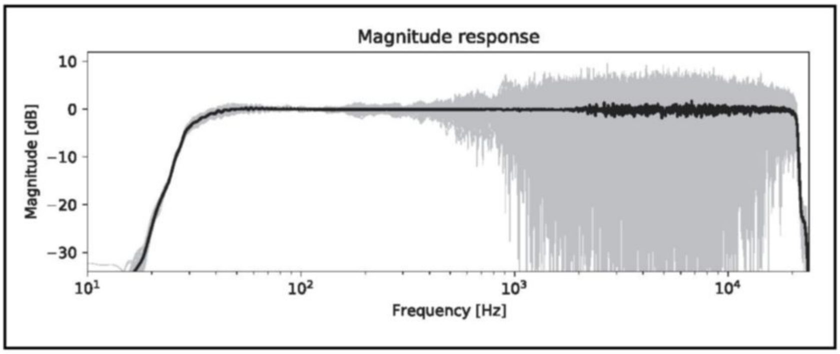


Figure 3: The measured response of a system using loudspeaker co-optimization for room correction, reflecting a flattening out, a reduction in spatial variations, and cancellation of room resonances.

The Benefits of Loudspeaker Co-Optimization

Loudspeaker co-optimization already exists in the market, although in targeted applications. The Volvo XC90 SUV, for example, has included an audio system that co-optimizes the car's loudspeakers in frequency, time, and space for optimal bass integration and clarity. The Volvo solution is factory calibrated to recreate the acoustic environments of performance venues. Like in the room solution we've discussed, the individual speakers act as one super speaker to create an immersive listening experience for every seat in the automobile.

And as already mentioned, some of today's bass management solutions offer an early example of co-optimization, although for just one input channel. But true loudspeaker co-optimization can actually improve bass management.

In the MIMO control paradigm, bass-capable speakers (e.g., subwoofers) also contribute optimally to the reproduction of speakers with less bass (i.e., they can extend their frequency range if needed). The system thus performs a sort of implicit bass management, but in this case, the bass content of separate input channels is not re-routed to a single bass channel. It remains intact and is fed to separate (corrected and bass-extended) speakers.

All told, the overall benefits of loudspeaker co-optimization for spatial room correction are notable. For today's immersive, multichannel content, the sound system enjoys significantly improved correction with control in time, frequency, and space, including a higher degree of control over speaker and room response. The system can achieve a tighter sound; low frequencies are extended; and spatial variations in a room can be minimized (**Figures 1–3**).

And then there is this benefit: Through loudspeaker co-optimization and spatial room correction, users can shape or design the immersive sound field they desire. Just as a target curve can be used to shape sound to a specific taste in the frequency domain, in multichannel design, it's possible to shape spatial properties (**Figure 4**).

Users of a multichannel audio system with a DSP supporting co-optimized, spatial room correction may wonder what room response properties might be interesting to add in order to create a signature sound in a space where they

enjoy immersive content. What if a completely anechoic "dry" bass response is not always the most desirable? With new spatial control, users may be able to design other types of sound fields. For example, they could copy the room response of a larger, more comfortable listening space into smaller-sized home cinema.

And because audio systems and the rooms in which they're enjoyed change over time, the nature of spatial room correction—like today's traditional room correction—allows for in-room recalibration by the consumer, even as they attempt to shape a room's spatial properties.

In the end, two inexorable forces in home audio are coming together to form an opportunity seeking a solution: Room correction has established itself as a means for perfecting speaker sound regardless of space and acoustics, at the same time immersive, multichannel audio content is becoming more popular and prevalent. The additional speakers required for immersive audio formats enable a new kind of digital loudspeaker and room correction, one which goes beyond frequency domain and time domain optimization: spatial room correction.

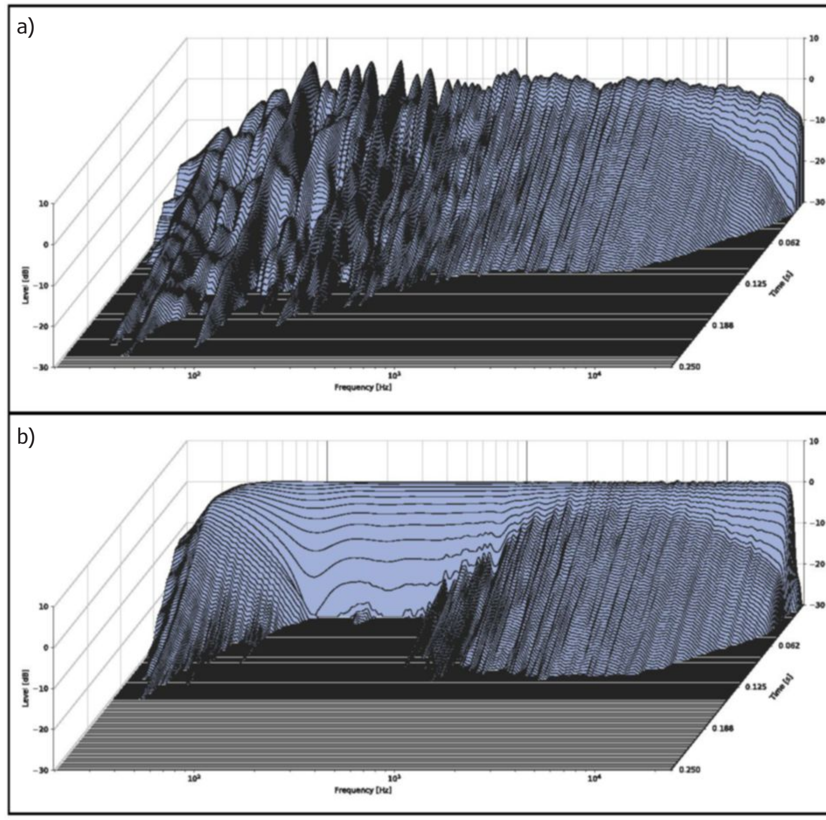


Figure 4: A waterfall plot of spectral decay shows how a room responds to sound before (a) and after (b) co-optimization. Notice how the solution removes low-frequency resonances.

Dayton Audio DAX88
An Eight-Source, Eight-Zone Distributed Audio Matrix Amplifier

Setting Up a Whole-House Audio System

By
Thomas Perazella

Photo 1: The front panel has a power switch and a series of indicator LEDs.

As Thomas Perazella was set to create a whole house audio system in his home, he learned about the impending launch of the Dayton Audio DAX88 Distributed Audio Matrix Amplifier and received one of the first units intended for beta testers. The article explores the system's main features from a very practical perspective.

The balance between listening to music you love wherever and whenever you want and spending time to improve the sound of the playback system to provide greater insight as to what the artist is trying to convey is a topic that has raged in audio discussions almost as long as reproduced music has existed. There is no doubt in my mind that the emotional effect of listening to the music I love is what drives me in my audio hobby.

Over the many years that I have been involved in audio, the main goal of my work has been to increase the contact with the artist during the performance. There are many factors that are involved, but one of the compromises of having such a revealing system is the requirement to be in the one room where the system exists, and often one location in that room.

For best results there also has to be considerable effort given to the construction and acoustic treatment of the room. The August 2021 issue of *audioXpress* has an article on the work that went into my listening room and the results. Interestingly, the end result is both expanding and restrictive at the same time.

Why a Whole-House System

Another fact in my music listening is that I spend a lesser amount of time with my reference system than I do listening to other sources. Probably the greatest amount of listening time is divided between working on my PC, in my workshop, sitting in the sunroom or on the screened in patio, cooking in the kitchen, and listening while eating. In those cases, compared to serious listening to my reference system, the primary difference is that the music is an adjunct to whatever I am doing rather the primary focus of my attention. In fact, having that close emotional contact to the artist in those cases can be a distraction to what I am doing. To sum it up, I spend a lot more time listening to a lot more music of all types in a lot more locations in my house than with my reference system. Thus, the importance of a whole house system.

In the past, I had multiple audio systems in various locations, each with their own sources and only local control. It worked but was somewhat restrictive and not cost effective. While in the process of designing a new house, I decided to

employ a whole house system that would give me greater flexibility. Requirements of that system included the ability to choose multiple sources and play them in different rooms with individual parameter controls for each room, the ability to provide music in at least six locations, have a network connection to take advantage of my music server and NAS storage, control capability via smartphone, and the ability to receive music both hardwired and via Wi-Fi. Researching the various options, I came across an announcement by Dayton Audio of a new whole house system called the DAX88. When I contacted the company, I found that it was in beta test stage so I volunteered to be one of the beta testers. They agreed to include me in the test and sent a DAX88 with a remote hub and keypads.

The DAX88 System

The center of the DAX88 system is an eight-source, eight-zone distributed audio matrix amplifier (**Photo 1**). If you look at that description, you might think that it is just another multichannel amplifier, but you would be wrong. Yes, it does contain 12 channels of Class D amplification rated at 75W per channel into 8Ω with all channels being driven, but that is only the beginning. I will attempt to provide insight into some of the main functions of the DAX88, but the list is way too long to include here. The Parts Express website (www.parts-express.com) has a lot of information on the system, including a very inclusive instruction manual (**Figure 1**).

Mounting

The DAX can be mounted in your system in two ways. The chassis has four large rubber feet for standard shelf or tabletop mounting. Also included is a rackmount kit containing two mounting tabs and hardware to allow fastening to any standard 19″ rack. Since the depth of the DAX is 14.25,″ a recommended rack depth would be 17″ or 18″ to allow for cable size.

Front Panel Controls

On the front panel is a power switch and a series of indicator LEDs. In addition to a power-on LED, there are six groups of two LEDs, one group for each zone. On the left of each group is a signal LED that illuminates blue if music is being played in that zone. On the right of each group is a status LED that is red if that zone is off and blue if it is switched on by means of the Matrio mobile app or a connected keypad.

Inputs

Input AC voltage is either 115V or 230V and is selected by a slide switch on the rear of the unit just

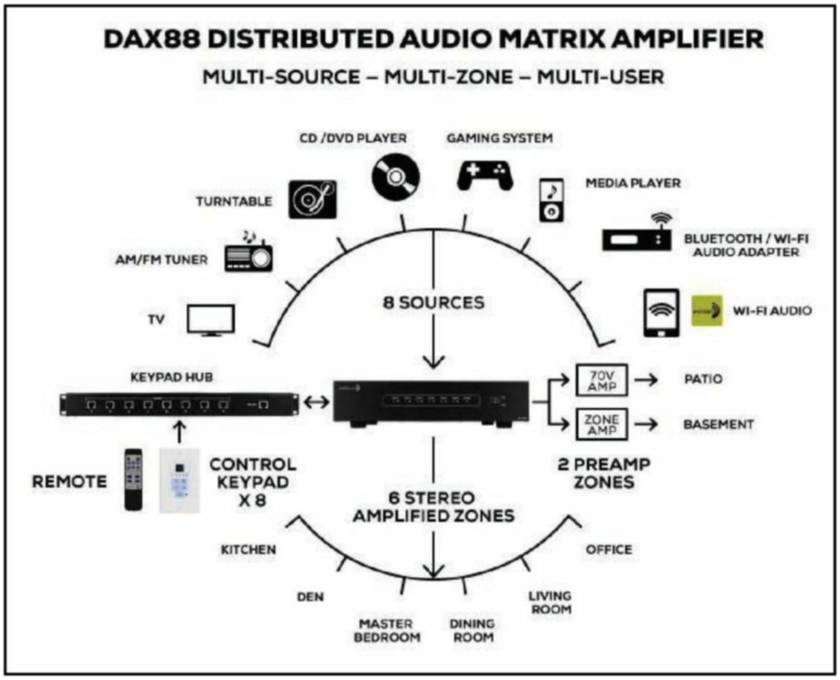


Figure 1: This is a source and output map for the DAX88 whole home system.

above the IEC power connector that has a built-in line fuse. Line level audio input is accomplished through seven sections of the rear panel. Sections 1–4 have both RCA input jacks and a single stereo 3.5mm jack. Sections 5–6 have a 3.5mm jack and Toslink digital optical inputs. Section 7 has a 3.5mm jack. In addition to the direct inputs, Channel 8 of the inputs is reserved for a Wi-Fi input music signal. There are four other control inputs including a 12V remote power on, a whole system mute, an RS232 connector for updates if ever needed, and an RJ45 jack to connect the remote hub to the DAX (**Photo 2**).

Outputs

The DAX has eight stereo output channels with six of the eight being driven by internal Class D stereo amplifiers. Each of the six stereo channels can be bridged by means of a slide switch to provide a mono channel that will increase the output power to 180W with the caveat that the speaker impedance must

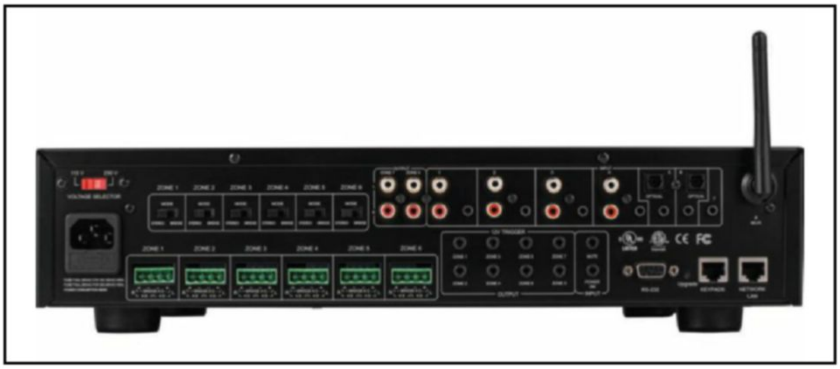


Photo 2: Rear panel connections include input AC voltage, an IEC power connector that has a built-in line fuse, and seven sections for line level audio. There are four other control inputs including a 12V remote power on, a whole system mute, an RS232 connector for updates, and an RJ45 jack to connect the remote hub to the DAX.

Photo 3: The keypad hub enables hardwired control of the DAX88.

be 8Ω minimum. The other two channels provide line level signals to external amplifiers, if needed.

The connectors for the six stereo amplifiers are four-conductor Phoenix-style types. Provisions are made for inserting bare wires into each of the separate but included Phoenix plugs and are fastened by set screws to prevent inadvertent disconnects. The plugs then feed into the chassis-mounted sockets. Using this system, the speaker wires can be fastened to the plugs first individually, in a space that is uncluttered before inserting them into the back of the DAX—a much easier method than trying to fasten wires directly to the rear of the chassis. Kudos on using this method. The line level connectors for output Channel 7 and Channel 8 are RCA type. Power control for associated equipment that uses a 12V signal is provided by 8 3.5mm jacks, one for each zone. These jacks will provide 12VDC whenever the zone is on.

Network Connections

The DAX88 can be connected to your network in two ways, directly wired or via Wi-Fi. As with any network device, direct wired is the preferred way and the method I used. I connected an Ethernet cable to the rear of the DAX using the RJ45 jack labeled Network LAN. The other end went to my gigabit switch that feeds all my wired devices into my router. For Wi-Fi connectivity, the DAX has an antenna on the rear panel.

Residential or Commercial Operation

The DAX88 system is equally comfortable working in a residential or commercial environment. Basic operation is the same for both, but there are a few functions that are better suited to either residential or commercial.

For larger commercial applications, multiple DAX88 control units can be interconnected. There is no limit to the number of units that can be added. The one caveat is that when using multiple units, they must be connected to the network via hardwiring. Wi-Fi cannot be used for multi-unit applications. When multiple units are connected to the network, the Matrio app will automatically recognize them and allow sophisticated control across all units.

Hardwired control of the DAX88 can be accomplished by means of the accessory hub, keypads, and IR remotes (**Photo 3**). This method seems most appropriate in commercial settings where direct control by many different users is possible with the least amount of conflict. For example, different DAX channels or zones can be directed to different conference rooms where the control of that room is possible only to people in that room.

Up to eight keypads can be connected via cat 5 or cat 6 cable to the optional hub, which then makes a single-wired connection to the DAX. This allows control in separate rooms directly without the need for the Matrio app. In a commercial application where it may be desired to keep people off the company network, this solution will provide the necessary control. The keypads allow control of most functions and, in addition to the onboard keys, can be controlled by an IR remote. If the keypad is in a location that is difficult for the remote to activate,

Photo 4: A remote keypad is linked to a specific output or zone of the DAX by configuration using the DIP switches on the keypad.

About the Author

Thomas Perazella received a Bachelor of Science degree from the University of California, Berkeley campus and he is now a retired Director of IT. Audio has been his passion for more than 50 years and he is a member of the Audio Engineering Society, the Boston Audio Society, the Philadelphia Area Audio Group, the DC HiFi Group, and the DC Audio DIY Group. He has written for *Speaker Builder* and *audioXpress* magazines. He has also authored several articles in professional audio journals and taught commercial lighting at the Winona School of Photography. Recently he received a patent on a cost-effective, high-efficiency LED lighting system for commercial and residential buildings. Thomas is also a Past President and Treasurer of the Rockville Chapter of the Izaak Walton League of America, one of the oldest national conservation organizations in the US.

an additional IR sensor can be connected to the keypad and mounted at a more convenient location.

A keypad is linked to a specific output or zone of the DAX by configuration using the DIP switches on the keypad. When using the wall-mounted keypad, if the DAX88 is in standby mode, depressing the select button turns the linked channel on (**Photo 4**). All keypads are backlit in blue when on. Depressing the select button for about 5 seconds turns the linked channel back off. The DAX itself remains in standby. When on, all indicator LEDs for active functions and the digital display characters are blue. When the keypad is off, only the select button is backlit.

When turned on, the keypad function defaults to volume. Depressing the select buttons scrolls across the functions from left to right starting with the default volume position, one position each time the select is depressed. Once in a desired function setting, the remote will stay in that setting as long as value changes are being made. If no changes are made for 5 seconds, the keypad returns to the default volume function.

The remote works in conjunction with the keypads to provide control over a large area using IR signals (**Photo 5**). Each keypad has an integral IR sensor and can also accept a second hardwired IR sensor that can be located in a different location for additional flexibility. The remote controls all the functions of the keypad plus a mute function not directly available on the keypad. The mute function is very convenient for temporarily stopping all sound from that DAX zone.

A limitation of using the remote compared to the keypad directly is that there is no visual indication of the changes being made by the remote. For example, when using the keypad, as you select a source, the source number appears in the display. With the remote, you have to scroll through the sources until you find the one you want by listening to the feed. I think the main use of the remote will be for volume control and mute.

The IR reception is quite good on the keypads. I got over 20′ in line with the sensor and it worked fine. At about 60° off-axis, it was still working at 20′. At 90°, it worked to about 10′.

Having two zones with line level outputs with RCA jacks allows connections to external amplifiers that may use 70V outputs to minimize power loss over long runs often encountered in commercial buildings.

Matrio

For residential applications, the simplest and most versatile way to work with the DAX is to use the Matrio app on a smartphone. I have an iPhone 8 that I use to control the DAX via the Matrio app, and it has proven to be an eminently useful way to control both what I am listening to and where (**Photo 6**).

Matrio provides control over the most commonly used features of music playback with the DAX. When the app is first started, it provides a home screen that lists the eight zones available. Although they are named ZONE1 through ZONE8, the zone names can be changed to match the locations in the house. For example, my ZONE1 is for my exercise room, ZONE2 is for my family room/kitchen area, and so forth.

Within each zone, the basic window shows a dropdown window of all the input sources. They are labeled Input1 through Input7 with Input8 fixed as Wi-Fi. Again, except for Input8, all the inputs can be renamed as desired. For example, in my case Input1 is called Music server. Input2 is called Blu for a Bluetooth to an audio adapter plugged into the 3.5mm jack.

Just to the right of the input window is a virtual slide button that turns that zone on or off. To the right of that is a grey speaker graphic with an "X" next to it. Touching that graphic will turn the speaker graphic green and mute the output for that zone. In the upper right of the zone window is a dropdown control that opens three other controls in the zone window. Those are Balance, Bass, and Treble that allow fine tuning of those parameters for the zone.

In addition to individual zone controls, Matrio has a function known as Group Control. You can put multiple zones into a group and then control power, volume, sync volume, mute, and select an input source for all the zones in that group.

Matrio also has a settings function where you can do a manual refresh of all connections, link to the

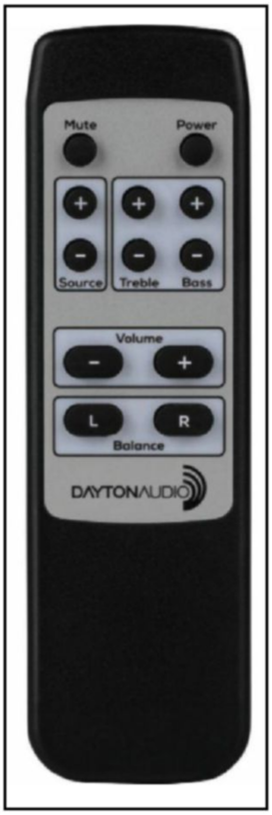


Photo 5: The remote works in conjunction with the keypads to provide control over a large area using IR signals.

Photo 6: For residential applications, the simplest and most versatile way to work with the DAX is to use the Matrio app on a smartphone.

Photo 7: The Dayton Audio model ME820C 8″ is a two-way with a micro-edge construction, which makes it blend well into the ceiling ME820C.

Photo 8: The Dayton Audio ME820C 8″ has four screws with four levers to mount the speaker frame to the drywall.

user manual, apply a custom name to the DAX, name the inputs, name the zones, do a factory reset, and see all DAX information.

Speakers

The majority of the rooms where I use the DAX have in-ceiling mounted speakers from Parts Express. They are Dayton Audio model ME820C 8″ two-ways with a micro-edge construction, which makes them blend well into the ceiling (**Photo 7**). In addition to great audio performance they are very easy to mount.

In the trade, there is a general separation of junction boxes and other electrical devices into two categories, new work and old work. New work means that they are installed before the wall or ceiling drywall is in place, which allows them to be mounted directly to studs.

For existing applications where drywall is in place and it would require a lot of work to open the ceiling, install the device, and then re-drywall and paint, old work devices mount by means of two or more levers attached to screws. When an appropriately sized hole is cut in the ceiling, the device is inserted into the hole and the screws are tightened. As the screws are turned, levers at the ends of the screws rotate so that the levers span the drywall. As the screws are tightened, the levers are pulled down tightly against the rear of the drywall holding the device firmly in place. These speakers use this type of mounting. They have four screws with four levers to mount the speaker frame to the drywall (**Photo 8**).

On the patio, which is covered and screened in, the speakers are still exposed to the weather. Mounted high on opposite walls are two Dayton Audio 6 1/2″ outdoor speakers that I have had for years. In my basement workshop, which has an exposed ceiling, I mounted two Polk Audio bookshelf speakers on brackets that I hung from the ceiling joists. The DAX performed exceptionally well with all of these speakers.

Overall Impressions

Convenience and flexibility rule. Having tried other solutions for music in various parts of my former houses, it is clear that the DAX88 brings a new dimension to whole-house sound. It is hard to remember all the machinations I used to go through to get the music I wanted at a particular time in various rooms. In fact, sometimes I would skip the music for a spur of the moment stop in a room because of the hassle of turning the system on, locating the music, setting the level, and so forth. Not a problem with the DAX. I simply take out my phone, start the Matrio app, and I'm off to the races.

It is also not only the convenience that is a winner, but the quality of the sound available from the DAX is great. I have never run out of power even when out on the screened in patio. It easily meets the standards of performance of any of the standalone systems I formerly used. Could you achieve a higher level of sound performance than the DAX? Yes, as my reference system can attest to. However, going that route is counter intuitive to the concept of convenient, flexible whole house music.

I can sum up my feelings in one statement. Installing the DAX88 is one of the best audio decisions I have ever made. Considering the how, when, and where I do most of my music listening, it is the perfect solution. I think you can tell I am mightily impressed.

Resources

T. Perazella, "Acoustic Room Treatments: Taming the Gorilla," *audioXpress*, August 2021.

Parts Express/Dayton Audio, www.parts-express.com

Directivity Measurement of In-Wall Loudspeakers

By
Christian Bellmann and Ruben Hauschild
(Klippel GmbH)
and Mattia Cobianchi
(Bowers & Wilkins)

This article details how Kippel's holographic measurement approach for directivity testing provides comprehensive, easy-to-interpret data that is relevant for transducer engineers as well as audio system designers.

Where should I place my speakers? In theory, I think I know the answer, but in reality, finding a good compromise between the optimal listening setup and the arrangement of the interior in a living room might become a controversial issue. With this issue in mind, in-wall loudspeakers are a very clever idea. The loudspeakers are integrated in the walls or the ceiling to be almost invisible. This can provide an optimal listening experience and doesn't interfere with the interior design.

Over the years, the applications, capabilities, and sound quality of in-wall loudspeakers have increased. Starting from single broadband transducers, often with medium audio quality, similar to those found in public places (supermarkets) or public transport (planes or subways), high-quality, multiway in-wall loudspeakers are serious alternatives to ordinary loudspeaker systems nowadays.

Also, in professional applications (cinemas or theaters) in-wall setups become more and more important and are scaled up to planar arrays with hundreds of distributed transducers used for modern 3D sound reproduction and virtual reality using ambisonics and object-based algorithms.

These applications are defining new requirements for the measurement process as well. The directional characteristics are an especially important factor, because the direction in which a loudspeaker emits sound highly affects the interaction with the listening room and the listening experience. Audio system developers need that 3D information, accurate in both magnitude and phase, to verify their designs and tune their DSPs. Moreover, the growing technical interest and knowledge in the hi-fi community is pushing the demand for accurate 3D data. Nowadays, new freeware simulation and auralization tools give almost everybody the opportunity to do simple acoustical simulations.

Measurement Challenges

However, the directivity measurement is still a challenge. Traditionally, these measurements have to be performed under free- and far-field conditions, which can be realized over a wide frequency range in an anechoic chamber. But for low frequencies (f<100Hz), most test rooms are insufficiently damped, which causes systematic measurement errors. Also, the large distance between loudspeaker and microphone (r>4m) makes an accurate phase measurement at high frequencies (f>10kHz) almost impossible, because small deviations in temperature affects the speed of sound and the propagation delay as well.

Coping with these limitations, the response of the device under test (DUT) is directly captured by moving the microphone at multiple positions in the sound field, typically placed on a sphere. Practically, this can be realized using either a microphone array or turntables.

In-wall speakers need to be measured in half space (2π) by mounting the DUT either on the ground floor of a semi-anechoic chamber or using a free-standing baffle in a full anechoic room. Unfortunately, a baffle of limited size may produce new measurement errors caused by diffractions at the baffle's edge and interference with the backward radiated sound. Using a rectangular baffle, as recommended by the International Electrotechnical Commission (IEC), can reduce but not avoid these artefacts.

Holographic Directivity Testing

Due to these known issues, Klippel has established a new approach for directivity testing that overcomes these problems. The system is designed to measure loudspeaker systems in almost any acoustical environment (e.g., offices) with the Near Field Scanner System (NFS).

Table 1: Klippel Scanning Solution for half-space testing

		
Limitations (size DUT)	<10″	<18″ (large custom baffle is applicable)
Applications	• Transducers • Single in-wall speaker	• Transducers • Single in-wall speaker • Large multi-way in-wall systems
Special Features	• Comprehensive testing combining acoustical, mechanical, and electrical measurements • Full Compensation of baffle effects, reduction of room reflections • Fast testing of round speakers by symmetry scans	• Most accurate acoustic half-space testing • Full compensation of baffle effects + room reflections • Flexible mounting of customized baffles

A robotic arm positions the microphone along two nested layers in the near field of the loudspeaker being tested. The DUT remains at a fixed position in the center of the scanner. This simplifies the handling for heavy devices and ensures a constant room excitation and reflections during the scanning process.

Based on this near-field measurement, the sound field radiated from the loudspeaker is identified using solutions of the wave equation (Spherical Harmonics, Hankel functions).

Due to the scanning along a double layer, the Direct Sound Separation (Klippel's patented technology) uses the additional phase information to detect the direction of the sound wave and can remove all room reflections from the direct sound of the loudspeaker.

This provides a new level of measurement accuracy, including a complete and comprehensive description of the entire sound field (near and far field). Moreover, it provides precise phase data also at high frequencies (f>10kHz) and a maximum robustness against external influences (e.g., reflections).

The holographic measurement approach is also adaptable to half space testing (2π). Half-space data can be acquired with different scanning solutions (**Table 1**). For small speakers, the multi-scanning workbench can be used and the acoustical measurement can be combined easily with electrical and mechanical measurements. Larger devices can be measured with the Near Field Scanner Carousel using an added baffle. Custom-made baffles are also supported.

We know from basic physics that a reflection on a plane can be modeled by a mirror image of the sound source (**Figure 1**). This virtual symmetrical sound field then can be modelled by using only a subset of spherical harmonics that matches this symmetry condition (**Figure 2**).

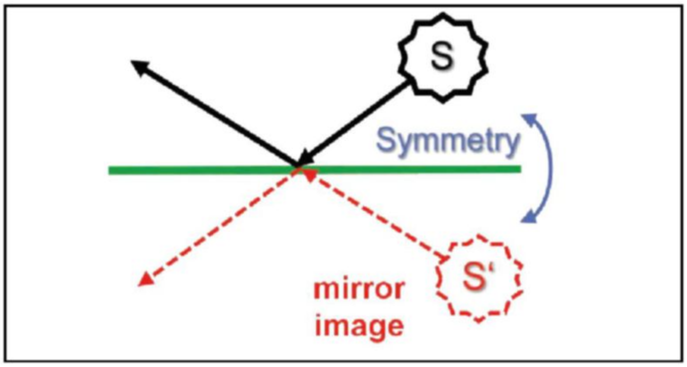


Figure 1: Sound reflection on a plate

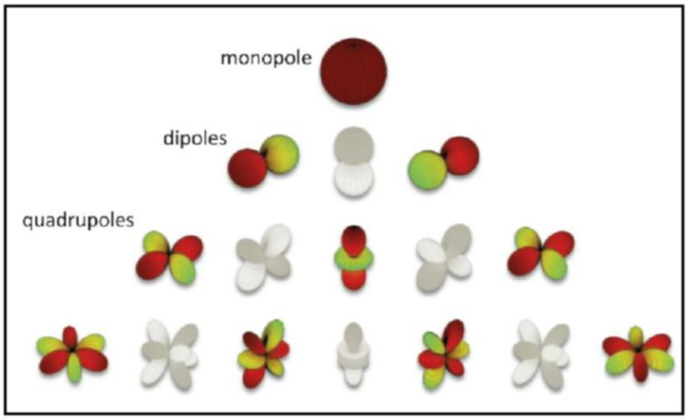


Figure 2: Subset of spherical harmonics for baffle symmetry

Finally, the holographic wave expansion can be applied. Furthermore, the Direct Sound Separation is opening new possibilities (**Figure 3)**. The measurement artefacts caused by the baffle (diffraction and backward radiated sound) can be identified and eliminated from the direct sound of the loudspeaker, because they are outside the scanning surface. This produces perfect half-space data as if measured in an infinite baffle.

Measurement Example

To illustrate the performance of this testing method, measurements were carried out on the Bowers & Wilkins CWM 8.5D (**Figure 4**). The CWM 8.5D is a high-performance compact in-wall loudspeaker that takes advantage of the technologies of the B&W flagship 800D3 series. It employs a 7″ bass-mid cone transducer and a 1″ diamond dome tweeter.

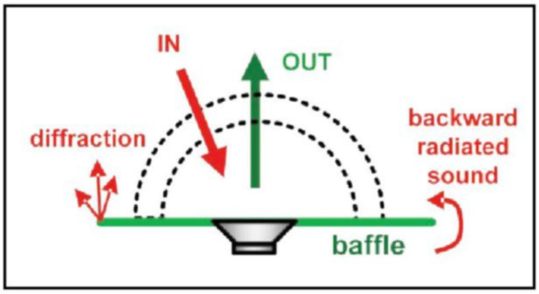


Figure 3: Direct sound separation for half-space measurement.

Figure 4: To illustrate the performance of this testing method, measurements were carried out on the Bowers & Wilkins CWM 8.5D, a high-performance compact in-wall loudspeaker.

The 7″ bass-mid cone breaks up in a controlled and progressive fashion from a relatively low frequency because of the cone's use of the proprietary Continuum technology. The speaker therefore maintains the ratio between the effective radiating area and the reproduced wavelength nearly constant across the passband. This results in an almost constant and very smooth horizontal directivity from 1.5kHz to 15kHz. This dynamic is visible in the -6dB iso-contour shown in **Figure 5**.

The damping component's careful optimization in Continuum's composite structure and the anti-resonance foam plug used in place of the dust cap also assures that the nonlinear distortion associated with break-up modes is very low. The CWM 8.5D's tweeter has a very low-resonance frequency. It is crossed over at around 3.5kHz, so the bass-mid reproduces most of the loudspeaker's critical midrange and blends seamlessly with the tweeter directivity. The low-resonance frequency of the tweeter also allows for an ultra-low distortion in the tweeter passband while the CWM 8.5's diamond dome guarantees that the first axisymmetric high Q break-up mode falls almost two octaves above the audible range at about 72kHz.

Measurement Setup

Due to the design, the CWM 8.5D needs a specific mounting. That's why a customized baffle was used to mount the speaker on the NFS Carousel. The scan was performed in a normal office room with about 1,000 measurement points and took about two hours. The spherical wave expansion was calculated with expansion order N=25, which was sufficient to characterize the entire sound field over the full audio band (20Hz to 20kHz). The accuracy is automatically self-checked during the identification using the redundant information of the raw measurement data.

Results

Finally, the CWM 8.5D's directivity can be analyzed by checking traditional far-field measurements (frequency response, sound power, and directivity index) or by simplified characteristics, such as the CTA2034 Spinorama. In addition, new near-field characteristics, such as the SPL distribution in a specified listening zone, can be analyzed.

The results show a very consistent horizontal coverage of ±50° with an almost constant directivity index of 6dB above 1.5kHz. The CTA2034 Spinorama

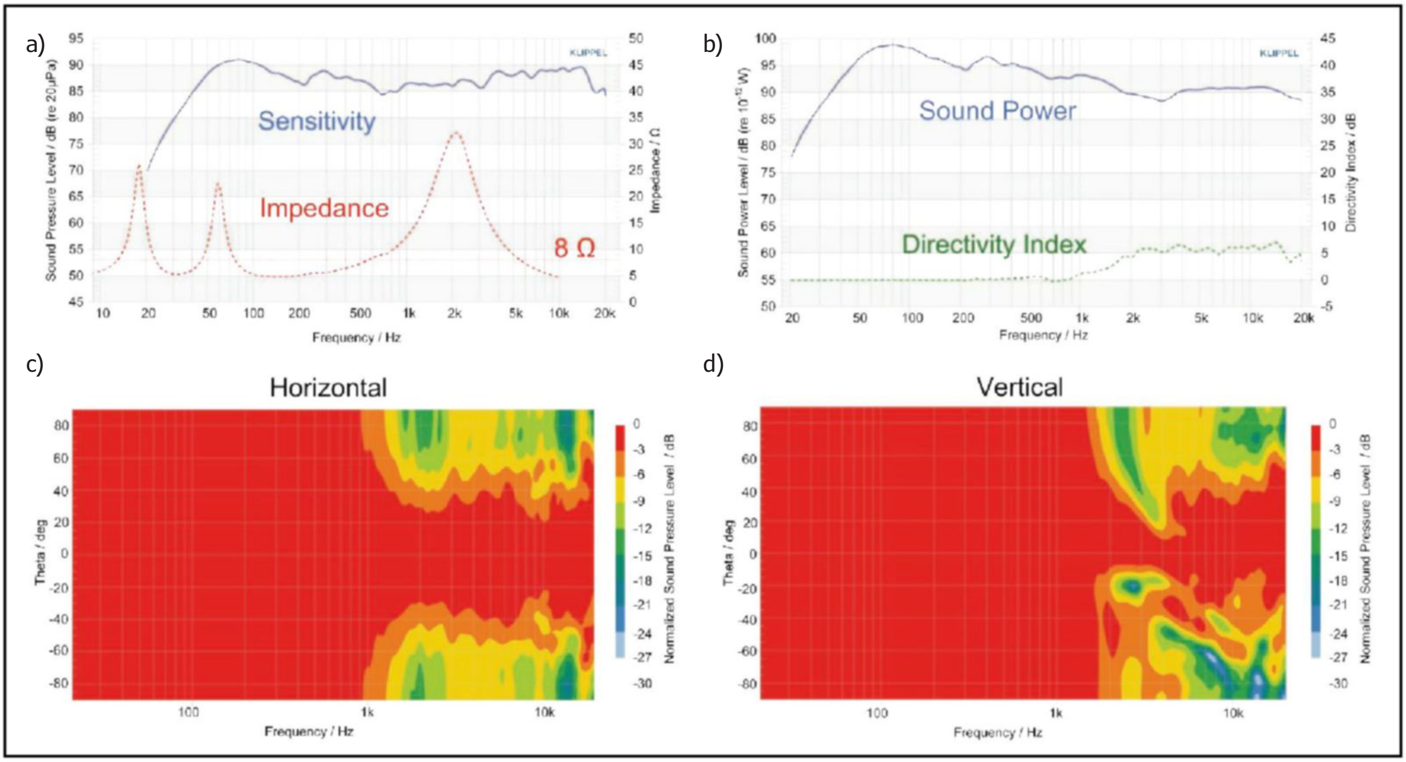


Figure 5: Far-field characteristics. (a): Sensitivity referenced to 2.83V @ 1m and electrical impedance; (b): Sound power referenced to 2.83V and Directivity index; Contour plots horizontal (c) and vertical (d), normalized to on-axis frequency response.

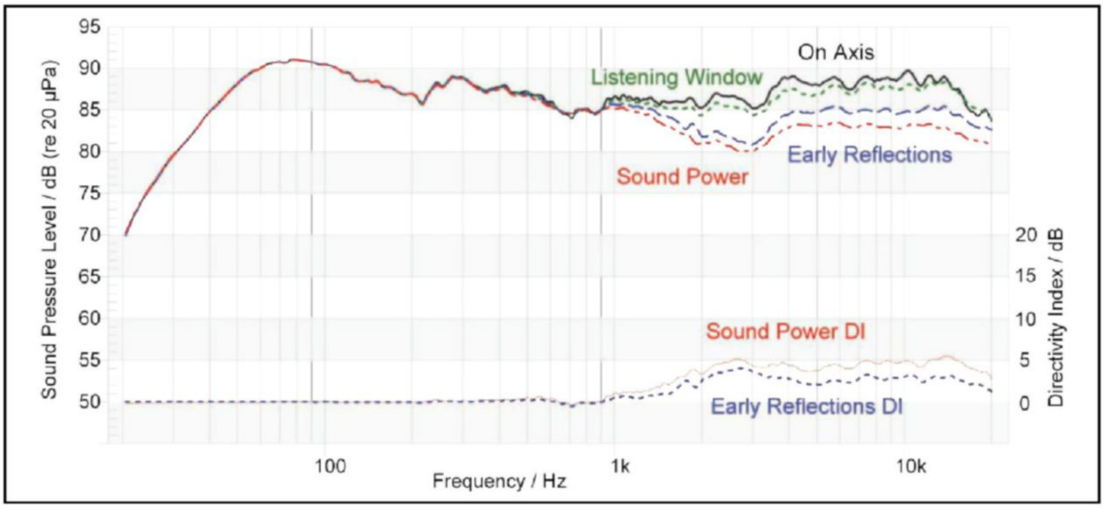


Figure 6: CTA2034 Spinorama referenced to 2.83V @ 1m

given in **Figure 6** shows a deviation of about ±3dB for the on-axis frequency response and is even smoother within the listening window. The comprehensive holographic parameter includes the energy transfer from near into far field as well. Thus, the precise sound pressure field can be calculated at any distance including the very near field of the device. In **Figure 7**, the sound pressure distribution of the CWD 8.5D at 2kHz is visualized. This accurate 3D data can be used for acoustic simulation software that predicts the performance of the loudspeaker in a listening room or any other application (public transport, alarm systems in buildings, etc.).

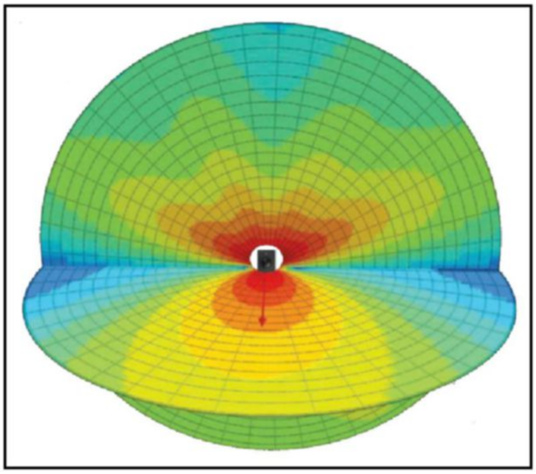


Figure 7: Spatial sound pressure distribution in the near field at 2kHz

Conclusion and Summary

The holographic measurement approach provides comprehensive, easy-to-interpret data that is relevant for transducer engineers as well as audio system designers. Hi-fi enthusiasts may also be interested in this information and motivated to learn about good sound quality and are willing to pay for it when evidence is provided.

Thus, objective testing is an important criterion besides the perceptual evaluation. In addition, the measurement produces the data that is required for room simulation tools, such as EASE, to simulate different test scenarios, including different loudspeakers placements, within an acoustically simulated space. It is well suited for professional or consumer applications.

About the Authors

Christian Bellmann studied mechatronics at Dresden University of Technology with a focus on control engineering, power electronics, and electrical drives. In 2013 he received a Diploma degree for his thesis "Separation of direct sound and room reflections using holographic methods," which was supervised by Dresden Technical University and Klippel GmbH in Dresden. After graduation, he joined Klippel, where he is currently engaged in the research and development of loudspeaker measurement systems with a focus on acoustical holography.

Rueben Hauschild studied electronic engineering at Dresden University of Technology with a focus on communication technology and acoustics. In 2019, he received a Diploma degree for his thesis "Maximizing the efficiency of active speaker systems with non-linear control," which was supervised by Dresden Technical University and Klippel GmbH. After graduation, he joined Klippel, where he is currently engaged in the support, research, and development of loudspeaker measurement systems with a focus on loudspeaker simulation.

Mattia Cobianchi is a transducer and acoustic engineer, specializing in drive units and cabinets modelling, innovative measurement techniques for loudspeakers characterization and psycho-acoustics. He has a MSc in Electronic Engineering and a Diploma in Acoustics. He's currently Senior Transducer Engineer for Bowers & Wilkins and a PhD student in the Music Department of Goldsmiths University of London.

You Can DIY!

Improved Loudspeaker Crossover

For the Parts Express "Solstice" Kit

By
Richard Modafferi

Richard Modafferi tests his invention, the two-way crossover, in a Solstice kit in an anechoic chamber on March 23, 2018.

In this project, the author describes a new crossover design and build project, originally created as a retrofit for the already-built Solstice Tower Speaker Kits sold by Parts Express. The project led to a patent application, issued in 2020.

This project began in early 2017, when an audiophile friend asked me to repair his speakers that were destroyed due to a failure in his transistor power amplifier. An internal short in the amplifier melted voice coils in both 6.5" drivers. The tweeters remain undamaged, but crossover parts melted. I seized that opportunity not only to repair his speakers but to try new ideas regarding crossover design. I was "forced" out of retirement… at age 81.

Three audiophile friends who, like me, own Pearl loudspeakers by Joseph Audio, had pressed me to "improve" on my late 1990's work for the company. This led me to return to my notes dating back to 1970s leading to an examination of network typologies, which may fix the (minor) faults in my previous work.

A review of Joseph Audio's "Perspective" speakers in *Stereophile's* July 2014 issue points to a "slight sound coloration" in the middle frequencies around 2kHz in what is an otherwise nearly flawless sound in listen tests. I knew the sonic problem was there, but based on the past performance of Joseph's speakers in audio shows where they consistently won first place for best sound, I saw no reason to attempt further research.

With that said, after examination, I had an idea of combining "constant-resistance" electrical network theory with my earlier work on my "infinite-slope" crossover ideas. After hearing about the project and listening to the result, other audiophile friends chimed in saying it was time to file a patent for my invention.

The patent expanded my previous work and crossover networks (US Patent No. 4407112 for "Phase-Shift Low Frequency Loudspeaker System" issued September 6, 1983; and US Patent No. 7085389 BI for "Infinite Slope Loudspeaker Crossover Filter" issued August 2, 2006). The new invention pertains to loudspeaker systems and, more particularly, to crossover networks displaying a combination of both steep and shallow slopes in filter amplitude responses and presenting substantially constant input impedances at their inputs.

It also provides improved crossover filter phase/delay responses relative to existing crossover filters. The resulting US Patent No. 10701487 "Crossover for multi-driver loudspeakers" was granted on June 30, 2020. This article details the project leading to the invention.

The Groundwork

After studying the sonic problem, I tried merging two separate concepts:

1. Infinite-slope crossover networks based on high-selectivity filters used in radio circuits, part of my original work and patents (Reference 1).
2. Constant-resistance electrical networks, as described in textbooks (Reference 2).

I first realized the crossover circuits merging concepts 1 and 2 using computer modeling software to develop the crossover schematic diagrams for two-way (woofer+midrange, tweeter) and three-way (woofer, midrange, tweeter) designs. That simulation work was done in December 2017. I obtained the components needed for building the two-way crossover and did the installation in a friend's "junk" box. The junk box worked so well that I tried the same two-way crossover in a Parts Express "Solstice" kit.

The Solstice is a two-way speaker kit designed by Jeff Bagby, a high-quality speaker using all Morel drivers in a MMT tower configuration: two TiCW 638Nd woofers, from Morel's Titanium series, and a CAT 308 tweeter, an updated version of Morel's classic MDT-30.

With the Solstice kit working, the test results and sound exceeded expectations. The prototype sounded so well that my friends came for a listening session at my home. Later listening sessions took place in their homes as well, comparing the sound of my prototype to several pairs of Joseph Audio's Pearl speakers. An astonishing result, with listeners recognizing they could easily hear a superior sound from the kit over the Pearls, which uses crossovers based on my earlier patents. The source material used for these listening test were audiophile-quality CD recordings and a *Stereophile* magazine test CD.

Upset that their $31,000 Pearls were upstaged by an inexpensive two-way loudspeaker kit, my friends demanded that I build and install a three-way crossover using the methods of my new invention. This three-way crossover design was installed in everyone's Joseph Pearls—including mine!

Photo 1 shows me installing a bass crossover in a neighbor's Pearl loudspeaker. As expected, the result was again a sound improvement. With my patent application validated, it was filed on June 28, 2019 and issued on June 30, 2020.

Meanwhile, I presented the two-way prototype installed in the Parts Express kit and my updated Pearls at an audio symposium in August 2019 (**Photo 2**). With many of my audiophile friends in the audience (**Photo 3**), the listening session once again showed how the inexpensive "Solstice" kit's sonic performance compared against the updated Joseph Audio Pearls.

The sonic signature of the two-way and three-way systems using my crossover topology being similar, a slight advantage was audible in midrange frequencies, as the three-way crossover removes bass from midrange drivers. The audience was clearly excited by the sound from both pairs of speakers,

Photo 1: Installing the bass crossover in a friend's Joseph Audio Pearl loudspeaker.

Photo 2: The author is shown with his two-way invention prototype (inside pair) and modified Joseph Audio Pearls (outside pair) at a public demonstration on August 15, 2019.

Photo 3: This is the audience at a public demonstration of the author's invention on August 15, 2019, at RIT-inn in Henrietta, NY.

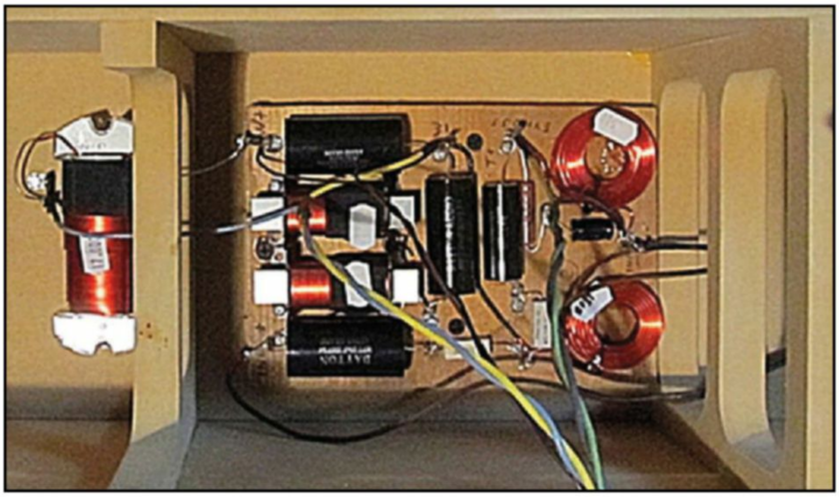


Photo 4: The two-way invention crossover installation is shown in a Solstice box with the 17mH coil next to the brace shown at left. NOTE: Damping resistor is absent here. The resistor is shown in Photo 8.

with most agreeing that the three-way Pearls had a "bigger" sound on loud orchestral and jazz music filling the large meeting room well.

The Solstice Crossover

Building the crossover project into the Solstice Parts Express kit requires rebuilding the kit box as received. To replace the original crossover kit with my two-way crossover you can use the original Parts Express kit box and order all the other parts needed directly from the Parts Express website. See the Parts List for the required part numbers.

I built my original crossover on a 7" by 5-1/2" by 1" (nominal thickness 3/4") pine board. The finished crossover will fit in the rear of the cabinet, behind the top 6.5" driver, second compartment from top of box (**Photo 4**). The crossover will fit through the 6.5" driver hole in the finished box, if removal is needed for error correction or repairs.

Begin the crossover assembly by attaching solder terminals around the board using #6×1/2" sheet metal screws, as shown in **Photo 5**. Be sure to attach jumper wire between the two terminals shown in **Photo 6**. Use #6x1/2" sheet-metal screws to attach both 1mH coils side-by-side centered as in Photo 6, leaving space to mount both 20μF capacitors alongside coils. Observe coil lead dress in Photo 6, and do carefully duplicate it exactly so mutual coupling between coils produces proper "infinite slope" function.

Both 1mH coils shown in Photo 6 have a "start" and a "finish" to winding. The coil shown at the bottom of Photo 6 has its "finish" wire connected to the crossover input terminal. The coil shown at top of Photo 6 has its "start" wire connected to W+, which feeds energy to both 6-1/2" drivers. Between the 1mH coils, the "start" of one connects to the "finish" of the other and both wires form a twisted pair for connection to remainder of crossover circuit (**Figure 1**). This arrangement of connections produces the opposing-field magnetic coupling transmission-zero "infinite-slope" function.

Attach the remaining components, including the two air coils, and make the solder connections to terminals following the Figure 1 schematic so that when you are finished it looks like the image shown in **Photo 7**. Note the "optional" 22Ω (damping) resistor—after building five kits, so far most users like the sound with that resistor included, but one person liked the "bright" sound with it omitted (Photo 4). Attach the air-core coils to the board with "amazing

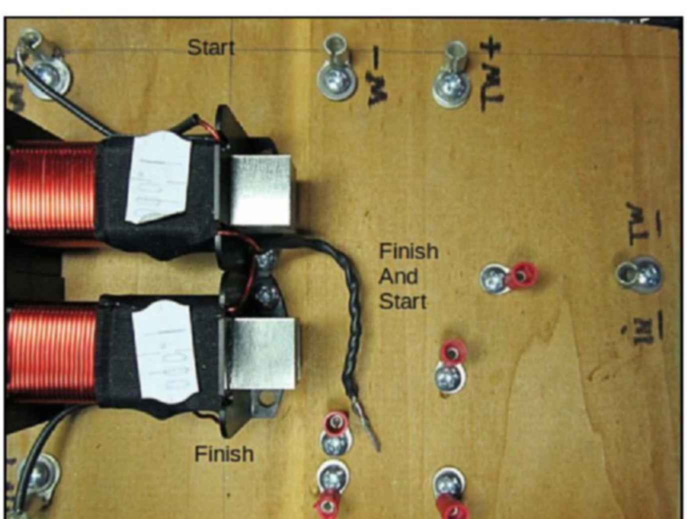


Photo 5: Here is the crossover board showing dress of 1mH coil leads and 10 solder terminals.

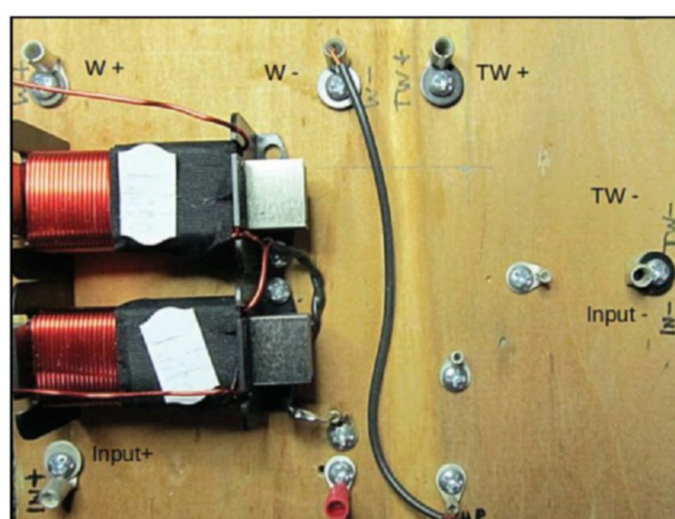


Photo 6: This is the position of jumper connection on crossover board.

goop," construction adhesive, or silicone cement. Recheck the jumper-wire "W-" to the junction of 0.56Ω/2Ω resistors. Also recheck the polarity connections to the coils to insure correct magnetic coupling with coil leads dressed, as shown in Photo 6.

The finished crossover can be tested by connecting an audio oscillator to input terminals, and the kit speaker drivers to the woofer and the tweeter crossover terminals (Photo 7). Check the woofer first, you should hear the response to 2kHz and a very fast drop-off (infinite slope!) to no sound at 3kHz and above. With the driver connected to tweeter terminals, measurements will show a flat response above 2kHz and slower drop-off below 2kHz. Once the crossover passes the test, it will be ready for installation in the kit box.

For those builders without test equipment available during the assembly, I recommend building the entire speaker system (all the drivers and the crossover) temporarily *without the box*, with all the pieces including the 17mH coil, spread out lying flat on table.

Attach the drivers to the crossover with clip-leads (Photo 7). Feed the music at a low level, check sound from the 6" drivers (low-mid frequencies), and tweeter (high frequencies). This procedure will be useful for finding errors before the final box assembly.

Speaker Kit

The Solstice loudspeaker kit is a two-way system with a 2kHz crossover, offered by Parts Express. The kit comes with all the wood parts in a "knocked-down" form, required for the cabinet construction, which the builder glues and clamps together to form the complete box.

The downloadable PDF instructions are well-written, and a video on the Parts Express website (www.parts-express.com) shows the box assembly method. Missing from the kit are input terminal blocks, wire, grills, the base, and hardware—all available from Parts Express or other sources separately. A bag of acoustical stuffing is supplied with the kit.

Required for the cabinet construction are 12" bar clamps, wood-glue (Elmer's) for assembly, as well as the usual tools, cabinet finishing tools, a jigsaw, a drill, a soldering iron, and so forth. After assembling five pairs myself, I have come up with some suggestions regarding assembly—expert builders modify as necessary.

The kit build starts from the bottom of the box up, following the instructions on the PDF, available from the Parts Express website. Note that in in the "Cabinet Assembly Guide Instructions" (page 3), the instructions seem to suggest the builder should

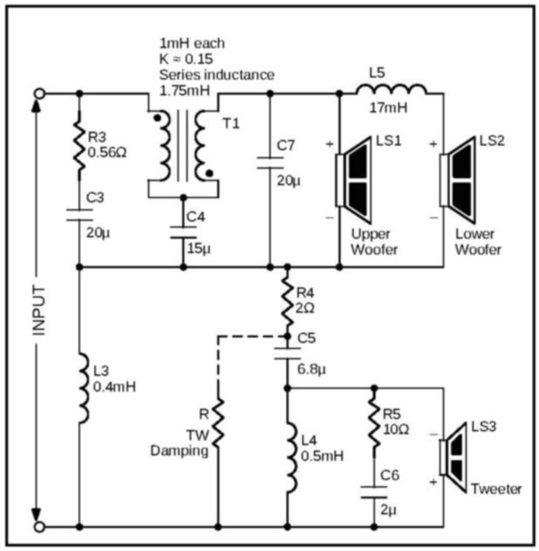


Figure 1: This is the crossover schematic for the two-way invention. The optional tweeter damping resistor is shown.

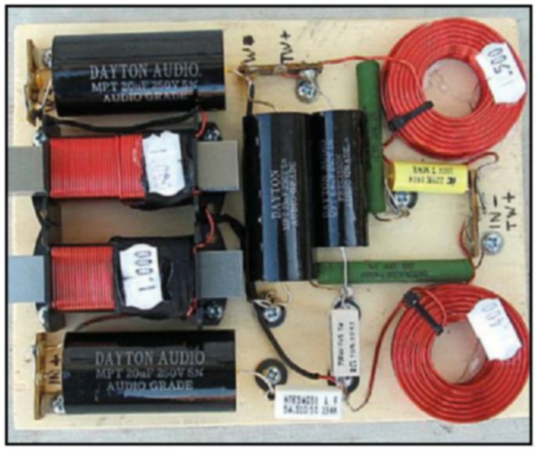


Photo 7: The complete crossover is shown with the 25Ω damping resistor and the tweeter control installed.

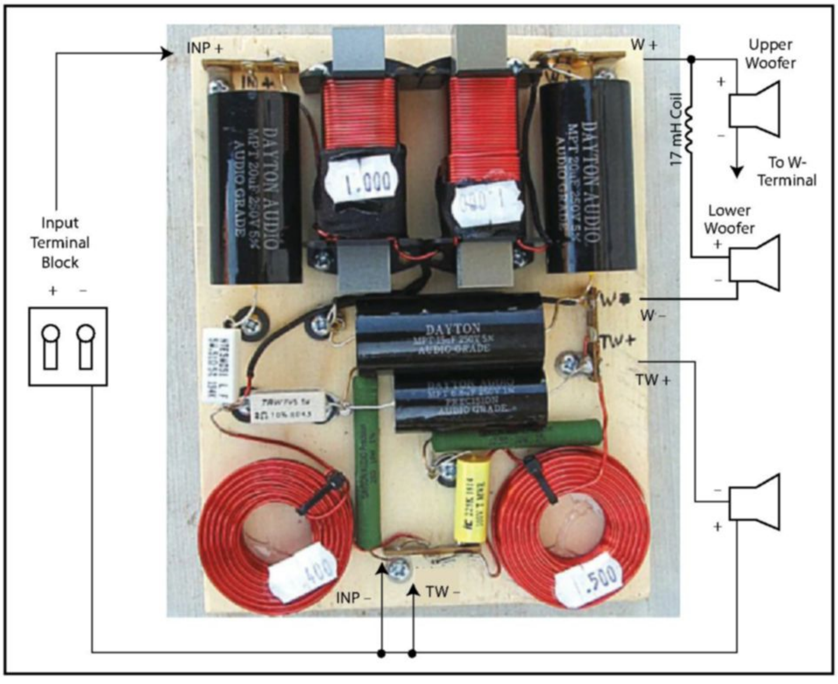
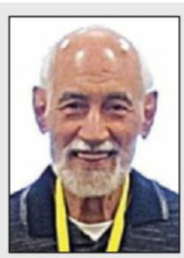


Photo 8: The complete box wiring diagram, observe the tweeter connection reversed phase.

Photo 9: The open Solstice box shows the acoustic stuffing two-thirds full, which yields "boom" in bass. Add all the stuffing provided for "honest" bass.

Photo 10: These Solstice speakers (shown here side by side with the Joseph Audio Pearl speaker) use fan guards to protect the drivers during transport.

About the Author



Richard Modafferi received his B.S.E.E. from Manhattan College in 1960, his M.S.E.E. in 1965, and M.S. Computer Science in 1968 from New Jersey Institute of Technology (NJIT). Richard worked for Blonder-Tongue Laboratories from 1960 to 1966 and McIntosh Laboratory from 1968 to 1974. He has been an independent inventor-consultant since 1974. He has also been employed from 1986 to present at Audio Classics in Vestal N.Y., performing restoration of vintage mostly McIntosh products (including some of his own design). He is the inventor of the "Infinite Slope" loudspeaker, which is currently in production by Joseph Audio. Today, Richard is still in running competitions at medal-winning levels in State and National Senior Games at age 83. He firmly believes that exercise floods brain, causing inventions, and he has nine patents as proof.

Resources

Parts Express, www.parts-express.com

Reference 1: US speaker patents issued to Richard Modafferi: US4,403,112; US4,771,466; US7,085,389B1; and US10,701,487B1

Reference 2: E. A. Guillemin, *Introductory Circuit Theory*, p. 307, John Wiley & Sons, 1953, www.josephaudio.com and www.audioclassics.com

apply glue and clamp the entire box in one operation. Since the glue sets quickly, making adjustments to the "square" parts after clamping might be difficult or impossible. I suggest the following sequence:

(1) Make sure all dado slots and surfaces are clear of dust, etc.
(2) Carefully check the fit without glue first, then glue and clamp the three window braces, and bass-reflex little shelf.
(3) Next, do the top and bottom cabinet pieces to insure they are both vertical and square with rear panel—check by "lining up" with temporary placement of one side panel—then glue/clamp these.
(4) Stop, let glue dry.
(5) Then glue/clamp the large side panels, leaving the top of the box open for installation of the crossover and the rear terminal block.

Apply your choice of finish to the "baffle" and curved "fascia" panel. (I have painted the baffle brown and the fascia black per photos on the website). Be careful not to paint or finish the bare wood where baffle and fascia join—the glue will not "stick." Glue these pieces separately after the paint dries.

Install the crossover and the 17mH coil into the open box as shown in Photo 4. The crossover fits into second compartment from the top of the box (located behind the top 6-1/2" woofer-midrange driver in the finished system). Drill three "pilot" holes in crossover board to clear three 1-1/2" "deck" screws; then attach the crossover to the rear of box. Three screws are visible on the crossover shown in Photo 4. If the deck screws look as though they will slightly puncture behind rear of box, grind off any excess length using motor-tool cut-off wheel. Attach the 17mH coil as shown next to brace, use two #6x1-2" "Drill" screws. Note that the crossover is installed using screws, not glue or adhesive! The crossover fits through the existing 6.5" driver hole in the finished box.

It is safe to first perform a temporary assembly of the front board (with the drivers) to the box without gluing things together (**Photo 8**). Lay the box upright and after clamping the front board to the box, make the connections to the drivers (don't stuff the box with damping material yet). Note that the tweeter must be connected "out of phase" since the infinite-slope crossover transformer inverts woofer-midrange phase at 2kHz frequency. (See Photo 8 showing the box connections and wiring.) Connect the system to a music source and listen for sound—perform acoustic tests if suitable test equipment is available. Determine if the sound is OK. If so, then have a

helper available to finish the kit by stuffing the box (**Photo 9**). Here, the Solstice box is open, showing the acoustic stuffing provided with the kit. The box is stuffed using about two-thirds of the stuffing provided yielding a slight "boom" to the bass.

That is followed by gluing and clamping the front board-driver assembly to the box. I used fan guards to protect the drivers (**Photo 10**).

Observations and Measurements

I performed electrical and acoustic measurements to my two-way Solstice kit. Figures 2–6 illustrate the advantages resulting from the application of the patent to this speaker design. **Figure 2** shows the frequency response test at three microphone positions. Note all three are closely similar.

Figure 3 enables readers to observe flat time delay in the author's new invention vs. his old one. Group-delay bump in old technology (Joseph RM20 from early 1990s) is audible causing a "lack of focus" in sound. The new design shows audible improvement in "focus/clarity" of sound.

Figure 4 shows the impulse response in the author's new vs. old technology. A Joseph Audio speaker model—1994 RM20 two-way system—shows a "smeared" impulse, tweeter first sounding then the woofer-midrange next. In the two-way Solstice

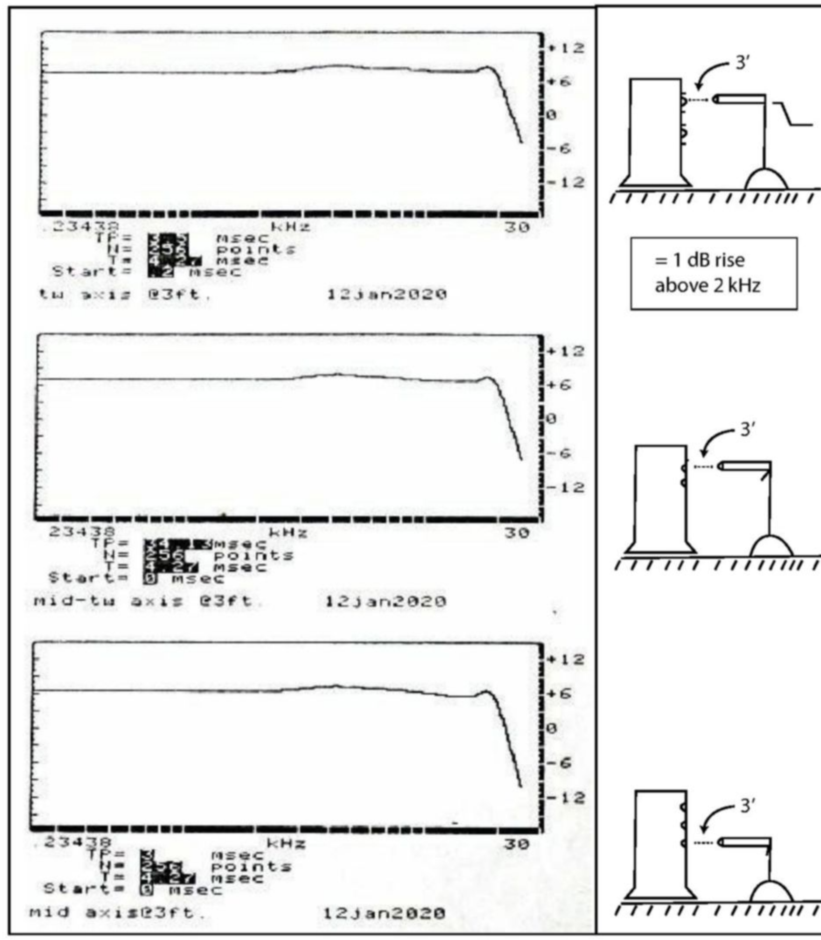


Figure 2: This frequency response test was taken at 3' with three microphone positions, and the Solstice box. Note: The 3' distance from microphone to box is not to scale.

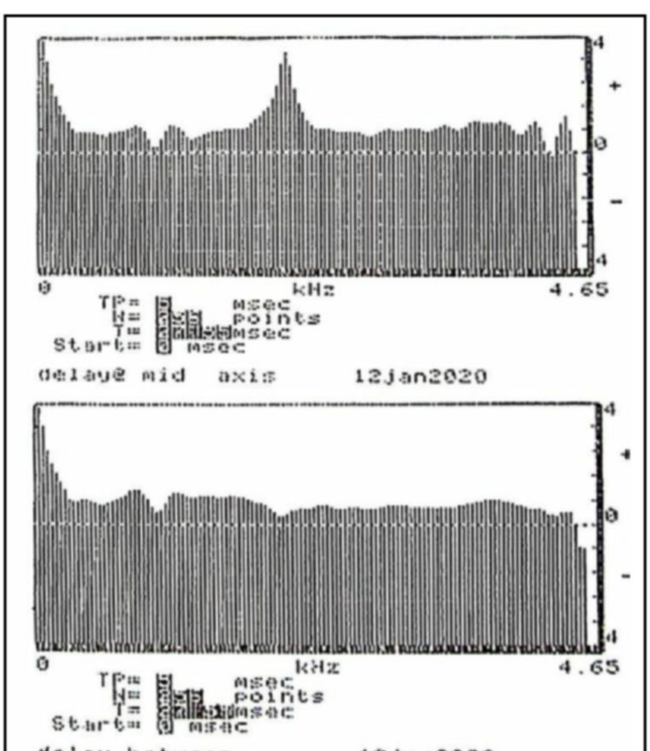


Figure 3: This figure shows the 2kHz region group delay in the crossover's new vs. old technology, as used in a Joseph Audio RM20 model speaker, early 1990s.

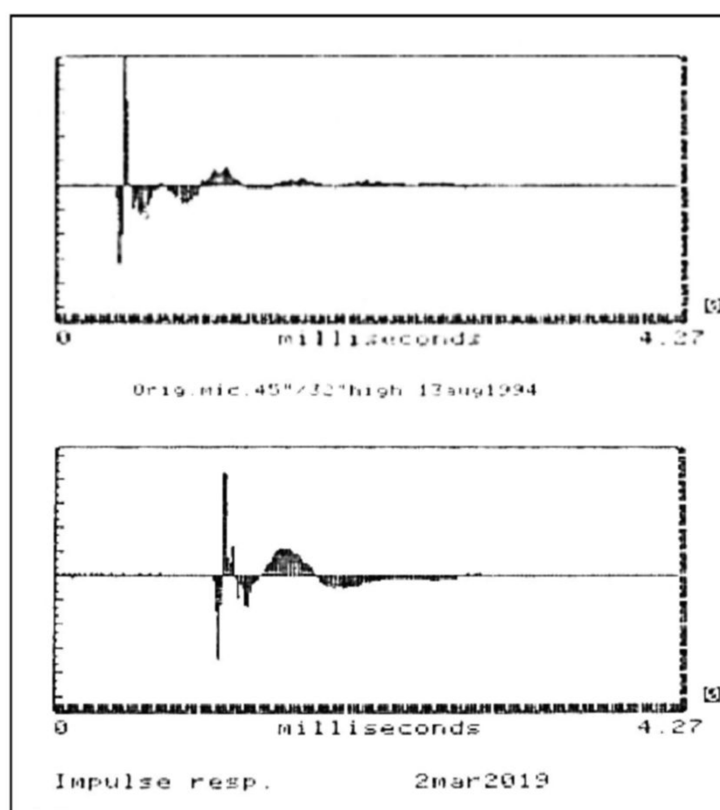


Figure 4: Here is the impulse response in the author's early technology vs. new.

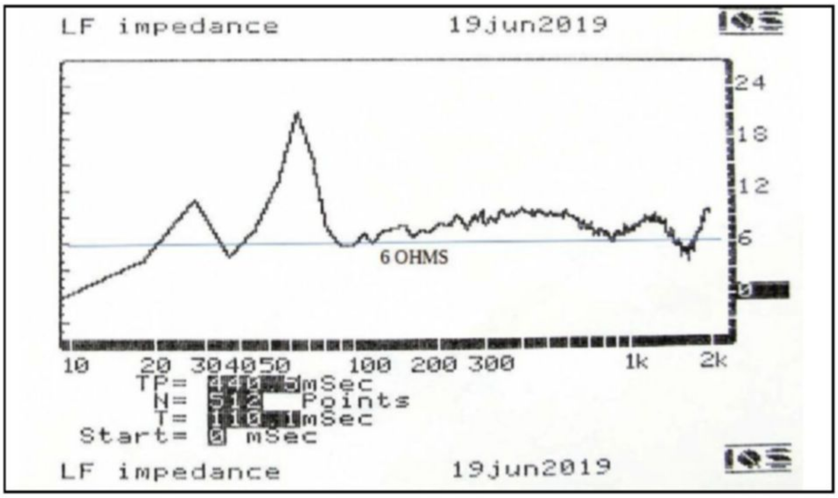


Figure 5: This figure shows the Solstice kit's input impedance in the bass-reflex region.

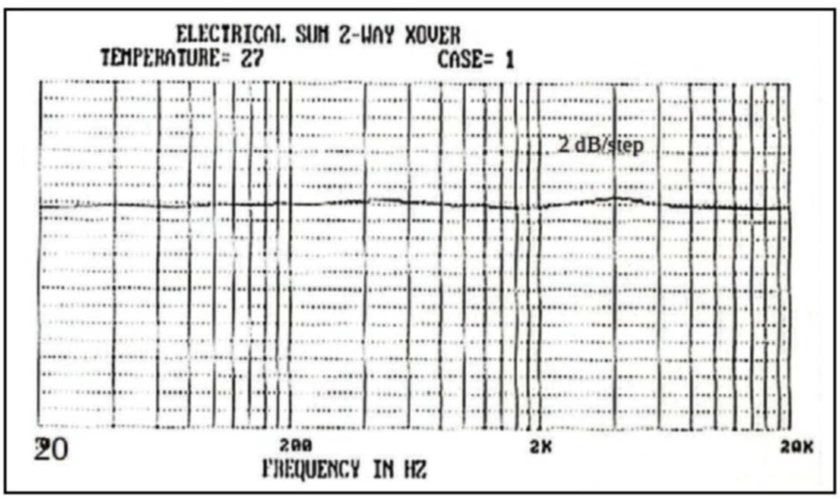


Figure 6: Here is the crossover electrical sum from the model. Note how closely the response here matches the acoustic measurements shown in Figure 2.

Parts List

Parts Express part numbers for each speaker
(for stereo, order two sets of parts)

Solstice MLTL Reference Tower Speaker Kit
Complete as listed on Parts Express website: Part# 300-708

Amount	Part	Part Number
2	1 mH coil	Part# 266-550
1	0.5mH coil	Part# 266-816
1	0.4mH coil	Part# 266-814
1	17mH coil	Part# 266-956
2	20µF capacitor	Part# 027-436
1	15µF capacitor	Part# 027-432
1	6.8µF capacitor	Part# 027-238
1	2.2µF capacitor	Part# 027-216
1	2Ω resistor	Part# 004-2
1	0.56Ω resistor	Part# 004-.56
1	12.5Ω resistor	Part# 006-125
1	22Ω resistor	Part# 255-956
1	Input connector back cup jacks	Part# 260-309
1	tweeter guard,	Allied Part # 609-5540
2	6.5" woofer-mid range guard	Allied Part # 609-5592

kit using the new crossover, the sound from all the drivers happens simultaneously with a smooth, symmetrical decay. Impulse response is a soft, clean "tick," while in the old one it sounds more like a "plop."

Figure 5 confirms that Jeff Bagby's bass reflex loading design is well done, showing a typical bass two-peak impedance characteristic in a critically damped bass response. Building a system using all the stuffing provided with the kit—completely filling the top three compartments—yields a flat, honest, critically-damped bass response. Leaving out about one-third of the stuffing yields an audible "boom" to low bass, giving the system a bit more authority; some listeners like this, especially in a room with "dry" acoustics.

Figure 6 shows the electrical (not acoustic!) response of the two-way crossover in my own build. The damping resistor for the crossover is missing in this model, yielding a slight treble boost (about 1-1/2dB in the 2kHz to 4kHz frequency range). Observe that the same treble boost is visible in the response graphs shown in Figure 2. This similarity shows correspondence of model and physical crossovers.

These measurements validate the absence of acoustic wave interference and also uniform acoustic delay in the midrange. Characteristic to the sound of my crossover design is an audible "coherence" and "clarity" in the sound. Absence of the so-called "sweet-spot" makes the sound have uniform spectral energy in the entire volume of the listening space, sounding the same anywhere in the room. This is limited only by the polar response characteristics of the drivers.

The frequency response measurements (Figure 2) show a rise of about 1dB in the tweeter range with the tweeter damping resistor omitted. Some listeners like the sound to be brighter, while others enjoy the flat response obtained by the addition of one shunt resistor in the tweeter circuit, as shown in the schematic (Figure 1). One could also install a "tweeter control" for adjusting tweeter level damping on the speakers, possibly a switch with several settings.

It's fun to switch drivers off in systems using my infinite-slope crossover. In a two-way infinite-slope system with a woofer-midrange driver operating by itself (no tweeter), there is almost no sound above 2kHz. Anything above 3kHz is gone, down 60dB to 80dB.

During a demonstration at a symposium (Photo 2), I disconnected the top sections of the Pearls, keeping the sound from the bass section only, using the 200Hz infinite-slope crossover using a CD playback of Diana Krall's "Live in Paris" concert, track one. She disappeared! Only the bass and drums remained!

Miking the Audience

By
Richard Honeycutt

Photo 1: Simply hanging mics above the audience is seldom the best approach.

Ever since the beginning of modern architectural acoustics, experts have focused on reverberation time as the most important parameter in auditorium performance. However, Richard Honeycutt believes that determining the direction from which sound arrives at the listeners' ears might be the real key to great acoustics.

We've all heard it said that there's a right way and a wrong way to do any job. Actually, there are often several right ways and several wrong ways! Often it depends on the purpose for which you're doing the job. Having identified the purpose, we can then progress to factoring in all the considerations that will lead us to the optimum way of doing the job.

One job that often raises "how-to" questions from audio technicians is miking audiences. There are at least four common reasons why you might want to mic an audience:

- To amplify audience questions and/or responses so other participants can hear them
- For recording or streaming questions/comments from audience members
- To pick up applause for recording or streaming
- To pick up audience singing or spoken responses

Amplifying the Audience

My first consulting job that involved miking audience members involved a conference room in which participants might sit anywhere in the room and could ask questions or make comments at designated times. The room was reasonably dead acoustically, so speech intelligibility was good, but only the conference leader on the stage benefitted from helpful early reflections to assist the projection of his voice to the audience. Audience members just had to "speak up" in order to be heard, presenting an obstacle to both speaking and hearing for those with weak voices or hearing challenges. The facility's staff had tried hanging mics above the audience (**Photo 1**), but this approach proved ineffective for three reasons.

The human voice is directional, as illustrated in **Photo 2**. Directivity is greater at higher frequencies than at low ones, which explains why it is easier to understand speech when the talker is facing us. Thus, an overhead mic will not pick up speech so that it will be optimally intelligible.

If you estimate the distance from a mic to an audience member shown in Photo 1, you'll see that only a few people in the sparsely populated front row are reasonably close to a mic. Remembering that sound level decreases as talker-to-mic distance increases, you will readily understand that few talkers in this audience will be adequately picked up by the mics.

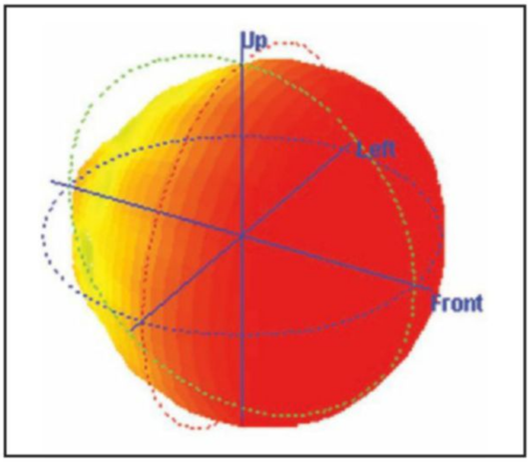


Photo 2: This balloon shows the frequency-averaged directivity of the human voice (Red: 0dB, Yellow: -10dB).

Adding more mics might seem to be the simple solution. Simple it is, but not a solution. Whenever there is an open mic in the same room as an active sound system, some of the signal from the speakers will get back into the mic, and if there is enough of that feedback signal, banshees invade the room! Doubling the number of open mics reduces by 6dB the permissible sound level before feedback squeal begins.

Another obvious trick is to use directional mics. In the equation for

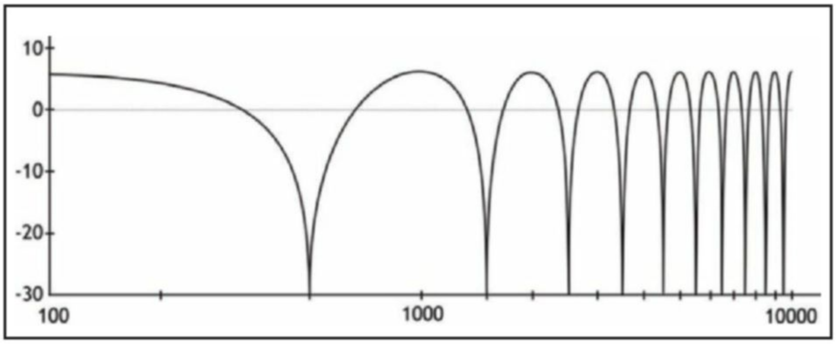


Figure 1: This frequency response, called "comb filtering," results from multiple microphones picking up the same sound source from different locations.

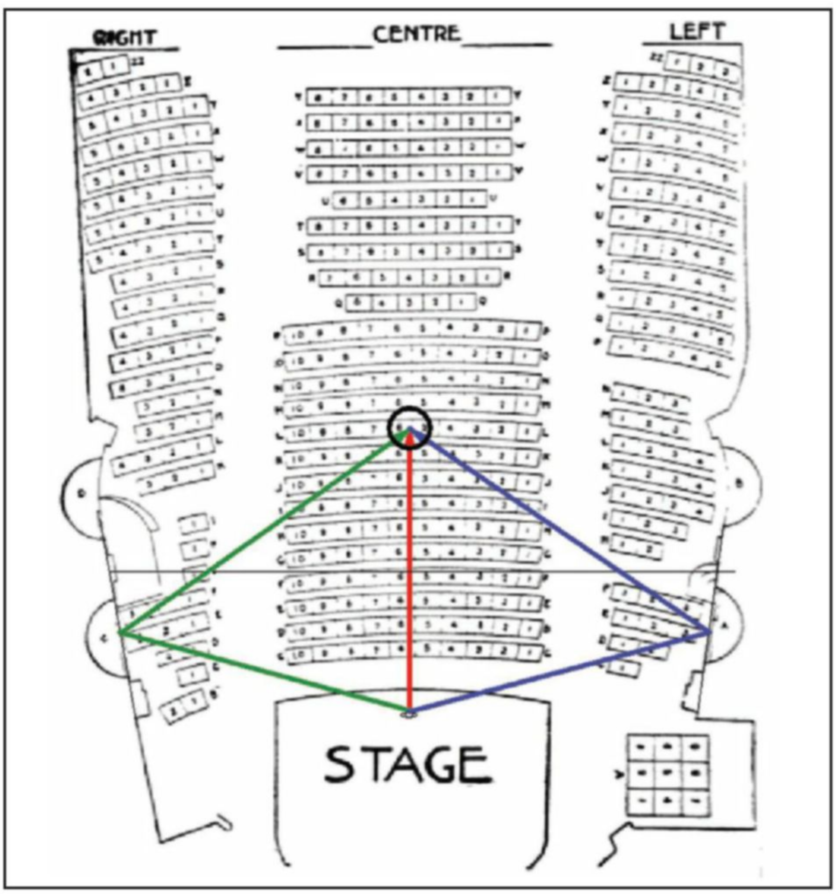


Figure 2: The red line represents sound arriving at the listener from in front; blue and green lines represent sound with laterally arriving components.

About the Author

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

calculating gain before feedback (GBF), the directivity of both the mics and the speakers can be seen to improve GBF. However, acoustical reality provides a "gotcha!" In any room that has reverberation—that is, any practical room, since we do not hold events in anechoic chambers—there is a distance at which the level of the direct sound from a source (e.g., talker, singer, or loudspeaker) equals the level of the reverberant sound. This distance is known as the critical distance. Reverberant sound is diffuse: It has no consistent directivity, no matter what or where the source is located. Thus, a directional mic located more than the critical distance away from a source will not pick up the source more effectively than an omnidirectional mic.

The equation for critical distance is:

$$D_C = 0.03121\sqrt{\frac{QV}{T}}$$

where:
D_C = critical distance,
Q quantifies the effect of mic and loudspeaker directivity
V = the volume of the room
T = the reverberation time of the room

If a conference room is 20′ wide by 30′ long by 12′ high and is fed by a loudspeaker having approximately a 90°H by 40°V directivity (Q≈5.6), with a reverberation time of 1 second, D_C would be about 6.3′. Thus, directional mics are not the panacea they may seem to be.

For conferences in which audience response must be clearly heard throughout the venue, two approaches have proven successful. Either mics with switches (either on the mic barrels or controlled by an operator) can be placed on stands where audience members can easily stand and walk to a mic to speak, or a handheld wireless mic can be passed around among audience members. The former approach is more practical for large gatherings.

Recording or Streaming Without Amplification

In cases not requiring audience members' speech to be amplified for the benefit of other audience members, matters are greatly simplified. With potential feedback squeal removed from consideration, the design problem becomes one of (1) making sure that the direct-to-reverberant ratio of sound picked up by each microphone is high enough for good speech intelligibility; (2) controlling audience mics' levels so as to prevent the sound of the presenter's mic from having a "barrel" effect due to excess reverberation

being picked up by audience mics that are open when they need not be; and (3) mic placement is chosen so as to avoid the "muffling" effect of miking from behind talkers.

Applause

The main requirements for audience mics used only to pick up applause, laughter, and so forth are that they not hotspot one or a small group of audience members. If the applause of a dozen audience members in a room of 1,000 is allowed to dominate the sound of audience response, the crowd will sound much smaller. The "3:1 rule" is helpful in these cases—the distance from a mic to the closest adjacent mic should be at least three times the distance of each mic to the nearest audience member. This placement also reduces comb filtering (**Figure 1**) that always results whenever two or more mics pick up the same source from different positions. Although it might seem counterintuitive, using more mics actually reduces the audibility of comb filtering by decreasing the width of the frequency band between minima, thus making the irregularities in response less noticeable.

Audience Singing or Speaking in Unison

Although miking choirs singing with or without accompaniment can be challenging (see the Sound Control articles in the February and March 2020 issues of *audioXpress*), miking audiences' singing and/or speech responses is less so. This is true because listeners know that understanding lyrics and the spoken word is unlikely when such a large group is singing or speaking. (When the bar is lower, it's easier to get across it!)

Ever since the beginning of modern architectural acoustics, we have focused on reverberation time (RT) as the most important parameter in auditorium performance (when your only tool is a hammer…). However, during the last decades of the 20th century, other metrics were studied. Arguably the most important of these relates to the direction from which sound arrives at the listeners' ears. Apparent Source Width (ASW) and Envelopment have been shown by some studies to correlate better with listener preferences for concert halls than any other metric. Both of these are related to the sense of being surrounded by the performers rather than hearing them from afar. Both are related to the Lateral Energy Fraction (LEF), which is the fraction of the sound energy reaching the listener's ears that originates from the sides rather than the front (**Figure 2**).

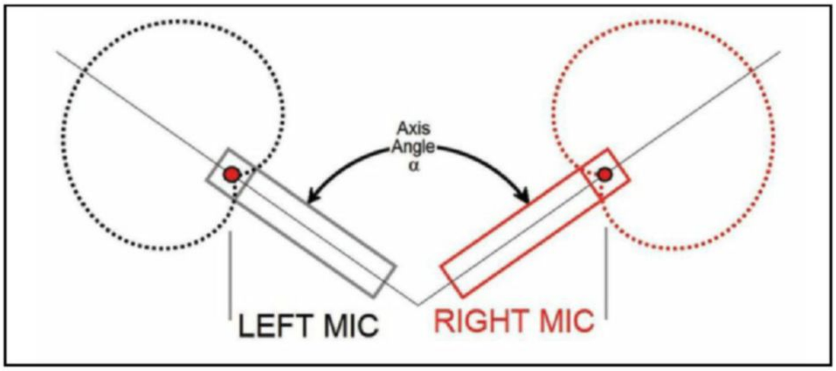


Figure 3: Miking an audience using coincident cardioid mics can increase the sense of envelopment.

While a greater LEF increases the ASW, LEF is not the best metric for assessing envelopment. Not only is the proportion of laterally arriving sound important, but the difference between the sound at the right ear and that at the left ear affects the sense of envelopment. While LEF quantifies the lateral vs. front sound levels, it does not tell us anything about the timing (phase and delay) of the sound, and these are also important for the ear/brain system to determine a sound's direction of arrival. All three parameters—level, delay, and phase—are included in an Interaural Correlation Coefficient (IACC) measurement, which quantifies the difference between sound heard by the left and right ears. A greater IACC improves the sense of envelopment, as discussed by Yoichi Ando (see Resources).

If you have ever listened carefully to the BBC broadcast of the annual Festival of Nine Lessons and Carols from King's College Cambridge, you have likely been impressed by the sense of envelopment—of really being there— during congregational hymns.

A part of this magic results from the acoustics of the venue, but some of it also results from mic placements that capture the lateral level, delay, and phase differences (the IACC) of the venue. One way of achieving this envelopment is to use coincident cardioid mics mounted above the audience and directed at the two sidewalls (**Figure 3**). The axis angle α can be anywhere from 90°to 180°, and is best determined by experimentation and listening. Of course, each mic will pick up the reflections from the wall at which it is aimed; and the left-hand mic feeds the left channel, and the right-hand mic feeds the right channel. Thus, the person listening to the stream or recording will experience envelopment similar to that of a listener at the live event.

Resources

Y. Ando, *Architectural Acoustics*, Springer-Verlag, New York, 1998.

King's College Choir, Cambridge, Nine Lessons and Carols 1992, www.youtube.com/watch?v=hNg6Nv1Ey8Y

Thinking About DC Power Supplies (Part 1)

Circuits Used as Potential Power Sources

By
Frans de Wit

This article was written with the ambitious intent to discuss multiple circuit types for "power supplies" or better "power sources" for high-end audio systems. Resourcing to simulation software as a tool to evaluate the performance, Part 2 will detail the types of circuits most likely to be used as the power source for high-quality preamplifier or line-level amplifier.

I can talk a lot about "power supplies" or better "power sources," especially when intended for high-end audio systems. Mostly I do so in public at events such as the German "Frickelfest" where DIY enthusiasts can talk all day (and night) about power supplies. As there is, in my opinion, too little actual information available regarding this subject I felt compelled to put some of my ideas and insights on paper as a means to get the discussion going. I will, in this first part of my article, discuss the types of circuits most likely to be used as the power source for that next RIAA preamplifier or line-level amplifier that you might build. In Part 2, I will go into more detail about shunt regulators in a specific application.

If you are interested in high-quality audio circuits or you need a high-quality low power "power source" then this is an article for you. Nowadays, circuit simulations can give you a very accurate view of the actual circuit performance, and this is what has been used in this study.

So, there is a companion, downloadable, set of files with this article that contains all the source material needed to reproduce the schematic circuits and plotted diagrams used/shown in this article. The content can be found in the Supplementary Materials section of the *audioXpress* website.

Electric Power Conversion

The electronic "power source," as discussed in this article, is an electronic device/circuit that supplies electrical power to an electrical circuit/load. The primary function of a power supply is to convert electric power from a source (e.g., mains) to the correct voltage, current, and frequency to power the load. For me, there is only one problem with most sources—they all deliver their power in the form of AC.

Electronics is, or at the least can be, extremely sensitive to pollution, it needs pure unpolluted DC and the purer it is the better it is liked. This is especially true for audio electronics, the form of electronics that operates over a bandwidth of up to 5 octaves and a depth range of 60dB or more.

Note: One of the Resource links will bring you to the Wikipedia page about Power Supply Rejection Ratio (PSSR)—it is the ability of a circuit to reject (ignore) any noise coming from the power supply into the signal that is generated. Although a high PSSR is desired, not all circuits will, or can, deliver this, so it is important that the power supply that powers our circuit is free, at the least as free as possible, of any noise.

Note: About noise, all noise considered in these examples is the signal that will remain after removing the intended output voltage and current. That is to say, we are considering DC power supplies, these are power supplies that should create DC and nothing else. However this is not possible, as will be demonstrated in this article, all "extra" signals measured at the output of the circuit are considered to be noise. What we will not do in this article

is to consider Gaussian and thermal or Johnson–Nyquist noise, which is a different "kettle of fish" (see Resources).

The real question is: Why is that purity of the DC so important? First, the Volt was discovered, and only many centuries later did we discover the Ampere. It was Alessandro Volta who pioneered its properties, and we had to wait for André-Marie Ampère to figure out that Current is the useful ingredient (of the energy contained in electricity). Please remember, it is the Volt that will tickle you, it is the Ampere that will kill you (on a bad day). If the Volt value is less than 50 it will be very unlikely that the Ampere will harm you. Anyway, as Volts were discovered first, we tend to see current as the result of applied voltage and load. If Amperes were discovered first, we would probably see Voltage as the result of applied current and load.

And to answer the previous question, in a more practical way—all circuits that we build are imperfect and some of the noise currents (!) existing in a power line will find its way into our circuits, causing undesired noise and distortion, we really need pure DC power for our circuits.

What Is the Problem?

Pure DC independent of the load, under any condition, including varying mains voltages and changes in current demand, over a wide frequency range and across time—that is the problem. When power is needed, it must be created and that costs time. Time we do not have. We want instant power because we need that 120dB or more of dynamic range, always, now or we will not satisfy that bandwidth criteria (4 even 5 octaves). So, we see time is of the essence. To be able to supply an audio circuit, operating at frequencies up to 50kHz (giving us some leeway from the minimum of 20kHz needed), we need a power source capable of delivering clean power up to 500kHz (yes, my expectations are high). This sets the reaction time for the power supply to 2µs, a demanding task.

Power Factor

Power factor is one of these things you might have heard about, but what does it mean for your high-end top-of-the-line audio preamplifier being developed and built and how does it influence that 120dB (or more) of signal to noise you are planning to reach? You do not know?

Consider this: The better that transformer, the faster that rectifier; and the larger that ultra-low ESR bulk capacitor, the bigger the problem.

All these quality measures that would predict a better, faster, and more efficient power supply do lower the angle of conduction. A lower angle of conduction will enlarge the peak current flowing and generate high-frequency current noise. Small power supplies, let us say 100mA output current on average, can easily generate currents up to 1A with frequency components in the megahertz range. Excessive pulsed currents like that may easily transmit/radiate from the power supply into other parts of the circuit and ruin your day. It is important to consider potential accidents such as that and prevent them from happening, especially in low-power, high-quality circuits.

Before continuing, let us have a look at the "normal" power source that we use to power the electronics that we use (**Figure 1**).

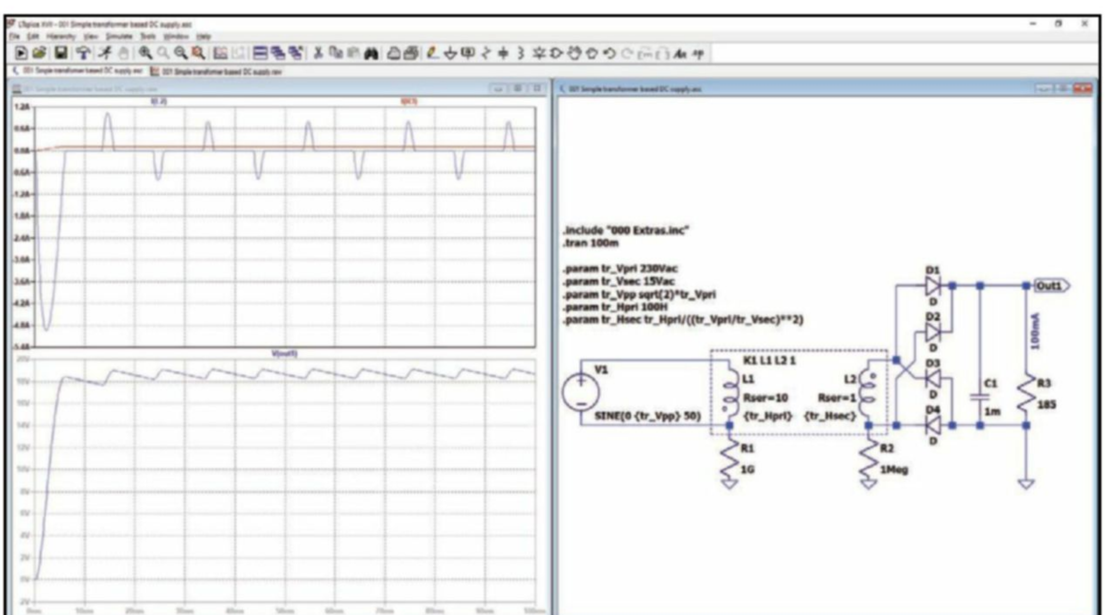


Figure 1: A simple transformer-based DC supply

Let me first briefly explain what can be seen in the schematic part of Figure 1. This is an LTspice simulation circuit with the following spice directives:

```
.include "000 Extras.inc"; Name of a file that contains a few transistor models and a few Spice directives.
.tran 100m; This sets the simulated run-time to 100ms for, in this case, a transient analysis.
.param tr_Vpri 230Vac; Vpri = The AC primary mains voltage used in this example calculation.
.param tr_Vsec 15Vac; Vsec = The AC secondary voltage of the transformer simulated.
.param tr_Vpp sqrt (2) * tr_Vpri; Vpp = The primary peak-to-peak voltage
.param tr_Hpri 100H; Hpri = The primary inductance of the transformer
.param tr_Hsec tr_Hpri / ((tr_Vpri / tr_Vsec) ** 2); Hsec = The secondary inductance of the transformer is
  calculated here.
```

Also, of interest is this Spice directive `K1 L1 L2 1`, here the K, coupling coefficient, for the transformer is set together with the inductors that make up the transformer.

Finally, in this short description, the Spice voltage source V1 has an associated voltage definition that reads: "SINE (0 {tr_Vpp} 50),"

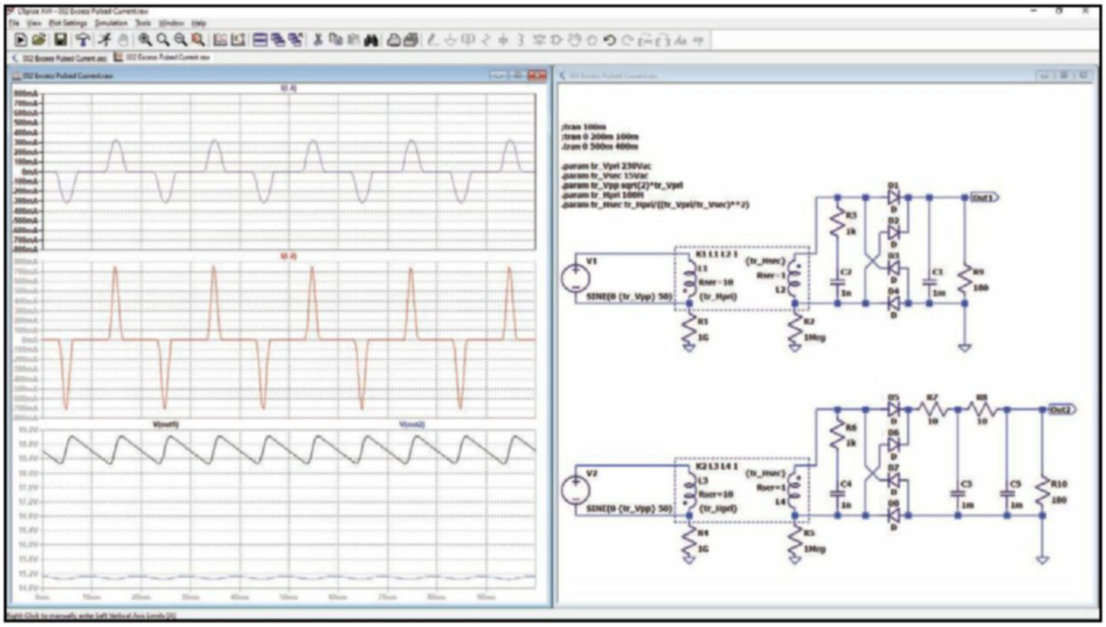


Figure 2a: The "Excess Pulsed Current" phenomenon can be seen by adding a second model to our simple circuit, spending two resistors and one more capacitor.

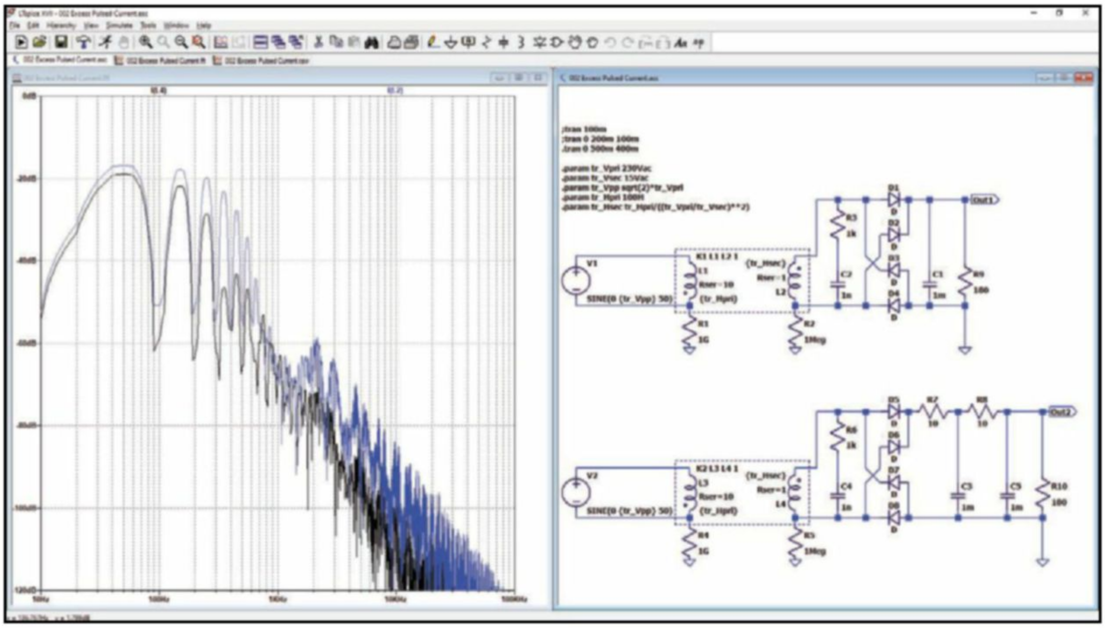


Figure 2b: This is the raw data in the form of a Fast Fourier Transform (FFT) plot.

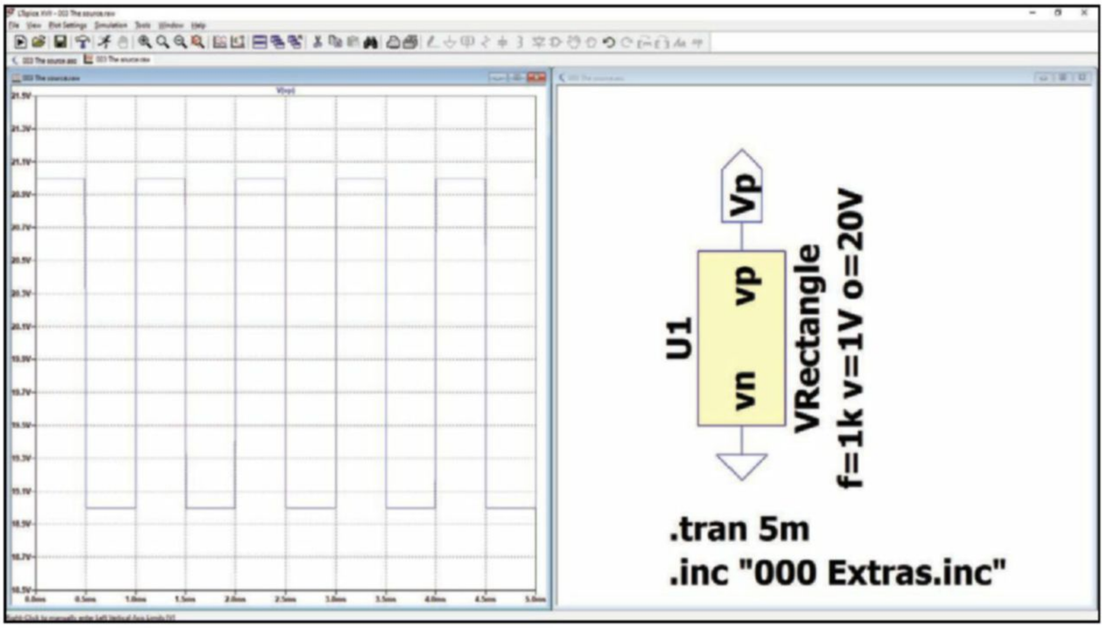


Figure 3: The power source is made to be imperfect; the average voltage is 20V and the deviation, or noise, part is a square wave of 1V with a frequency of 1kHz.

where: 0 is the Offset-Voltage; {tr_Vpp} is the Peak-Voltage; and 50 is the Frequency.

Note that "{tr_Vpp}" refers to the parameter statement that was previously shown.

Simple? Or maybe not so much, especially when supplying high-quality audio supplies, we want/need at the least 120dB (yes, I am a dreamer, 120dB is a lot and may or may not be reached but let us keep the dream alive) of headroom. Will this circuit cut the mustard? I do not think so. Why not? After presenting this circuit, I believe some points must be made.

Let us have a closer look at this "Excess Pulsed Current" phenomenon by adding a second model to this simple circuit, spending two resistors and one more capacitor, we can directly see this phenomenon at work. **Figure 2** basically shows the same power supply—for completeness I did add a small Zobel to remove most transformer resonances, but this Zobel will have little effect on the problem at hand.

Looking at the left part of the screen, top to bottom in Figure 2 we see:

- The transformer secondary current of the "improved" second schematic
- The transformer secondary current seen in the original simple schematic
- Output voltages of the original (Figure 2a) and the improved (Figure 2b) schematics

Voltage, as a source of stray signals, is not a problem, or better not the main problem. The real problem is current. Current is the "item" that generates stray signals, it will do so excessively and it will travel the ether as magnetic energy and will re-convert back into electrical energy at many points in the circuit that we were trying to supply with clean energy.

The right half of the Figure 2a schematic shows the previous simple schematic and the right half of Figure 2b shows an "improved" version. As we can see, this second circuit looks very strong, unlike these old power sources that were used by our grandparents, when they built their tube-electronics. Maybe that was not such a bad idea at all, maybe we should spend that "extra" capacitor and resistor.

The left half of Figure 2b shows the raw data in the form of a Fast Fourier Transform (FFT) plot.

Average RMS "Excess Pulsed Current" is down by about 20dB to 30dB at little cost. Maybe we need to rethink that DC source, drop a bit of efficiency in exchange for some gain in quality. We also see a large improvement in voltage ripple, the ripple is down from 680mV at "out1" to 90mV at "out2," all at the low cost of two small resistors and one capacitor.

The "Dirty" Power Source and the "Demanding" Load

Let us have a quick look at the power source and consumer that are being used in the continuance of the article. The power source is made to be imperfect; the average voltage is 20V and the deviation, or noise, part is a square wave of 1V with a frequency of 1kHz, as can be seen in **Figure 3**.

About the Author

Born and raised in the Netherlands, **Frans de Wit** went to a high school to become an electrical engineer. After this he found his first job in electronics retail, selling components over the counter and advising customers on component selection and application. Here he was spotted, by a headhunter, and invited to work for the Dutch importer for Motorola, Texas Instruments, Hewlett Packard, and many other great brands of the time. There he started his career as a Desk-Sales Engineer. During this time, he helped customers building trains, organs, milk-floats, and particle accelerators (yes, the one at CERN) among other things. Next, he co-founded an IT company and until 2016 he worked as a programmer and algorithms specialist building operating systems, compilers, and a large DAM system. In 2016 he founded a long-time ambition, an audio research and development company named Signature Origin.

The consumer or load that will be used for the following experiments is a current sink. The average current being sunk, or consumed, is 100mA with a deviation, or noise, part of 20mA and a frequency of 333.33...Hz (1kHz/3), as can been seen in **Figure 4**.

To easily create readable circuit diagrams, I opted to use a set of sub-circuits that represent the voltage source and the current sink in each circuit shown, the source material for these components are also supplied in the file-set that accompanies the article.

Back to the Problem

This might be surprising, but it is not as simple as is shown in the previous figures. The world in which electronics survives is not a friendly one—inductances, capacitance, and resistivity can be found at any place, they will "collaborate" to prevent you, the designer, from reaching your goals—those 120dB of dynamic range and 4 or maybe 5 octaves of frequency range.

Have a look at what happens when we take that nice, well-organized voltage source from Figure 3 and we connect a reactive environment, two resistors, two inductors, and one capacitor, followed by a passive load, and a constant current sink of 100mA. Suddenly it is clear that there are evil forces at work here, up to and over 60dB of range is shown in the FFT plot (**Figure 5**).

The current domain shows complexity beyond the simplicity indicated by the load as shown in

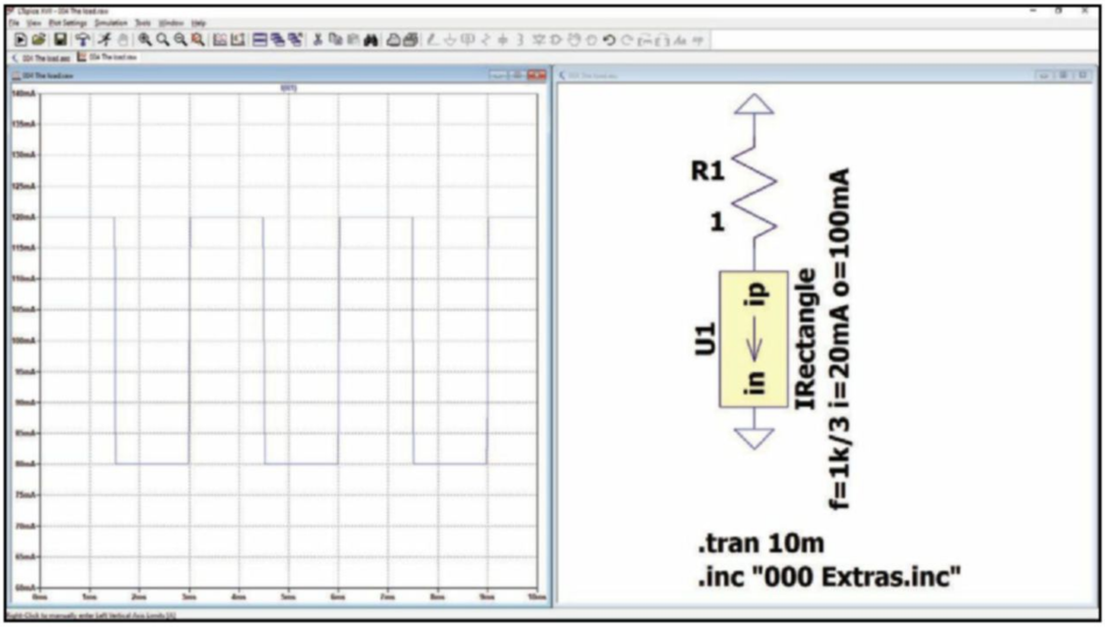


Figure 4: The consumer or load, that will be used for the following experiments is a current sink.

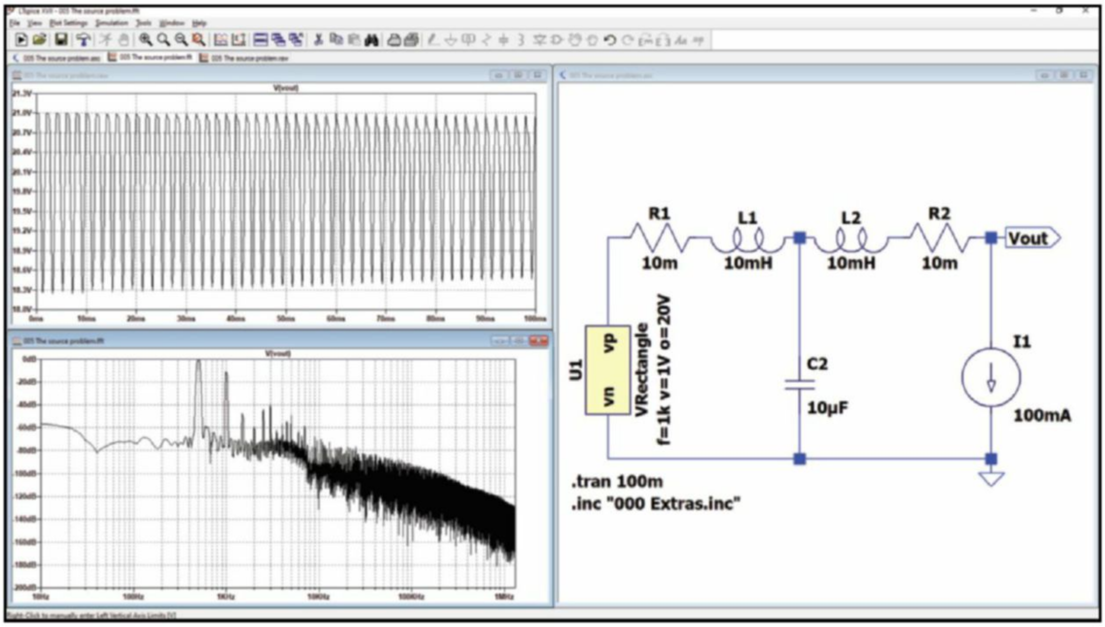


Figure 5: This is the outcome when we take that nice, well-organized, voltage source from Figure 3, and we connect a reactive environment, two resistors, two inductors, and one capacitor, followed by a passive load, and a constant current sink of 100mA.

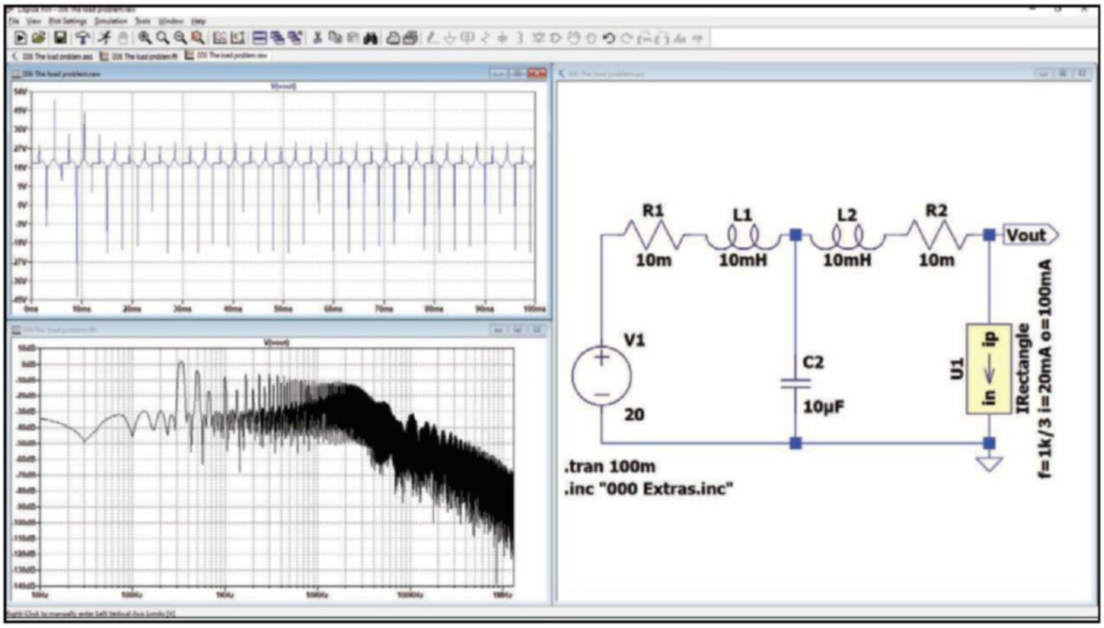


Figure 6: The current domain shows complexity beyond the simplicity indicated by the load.

Figure 6, which shows what happens when we connect the current sink to a simple constant voltage source.

Let's face it, we need to take strong measures to prevent these disruptive signals for ever entering the audio devices that we want to listen to. These artifacts will destroy the fine details in music. Some of these disruptive forces can be prevented by the DC power source, the only simple properties it needs to exhibit are, delivering a constant voltage and, at the same time, have a zero (0) output impedance.

Next Month

As we will see, it is not a simple task. The humble DC power source will be able to provide a reasonable facsimile of a DC voltage, and it will also do this at a reasonable low impedance. As we now will enter a more practical approach to the problems at hand, we will first address the geometries of the solutions available (not all, just seven). This will be in the form of the next seven circuits (yes, this article is finally about electronic DC power supplies). The seven circuits we tackle in Part 2 of this article will be:

- Just a Zener diode.
- Just a transistor.
- Series, out of the emitter.
- Series, out of the collector.
- Shunt, grounded emitter.
- Shunt, grounded collector.
- And finally, a reality check, using a LT1083 TO220 (3-pin) chip regulator.

Project Files

To download additional material and files, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html
To be able to fully make use of the files provided with the article, you need to install a copy of the LTspice circuit simulator software which is available from: www.analog.com/en/design-center/design-tools-and-calculators/ltspicesimulator.html

Together with the simulation files comes one included file with all simulations provided with the article. The file contains a few transistor models and a few Spice directives. Just open the file with a simple text editor (e.g., Notepad) to examine its contents:

.options numdgt = 7 Setup for high-resolution calculations

.options plotwinsize = 0 This will improve the drafting quality.

Resources

More detailed information about some of the terms used in this article series can be found on *Wikipedia*:

Alessandro Volta, https://en.wikipedia.org/wiki/Alessandro_Volta

André-Marie Ampère, https://en.wikipedia.org/wiki/Andr%C3%A9-Marie_Amp%C3%A8re

Complementary feedback pair, https://en.wikipedia.org/wiki/Complementary_feedback_pair

Direct Current, https://en.wikipedia.org/wiki/Direct_current

Gaussian noise, https://en.wikipedia.org/wiki/Gaussian_noise

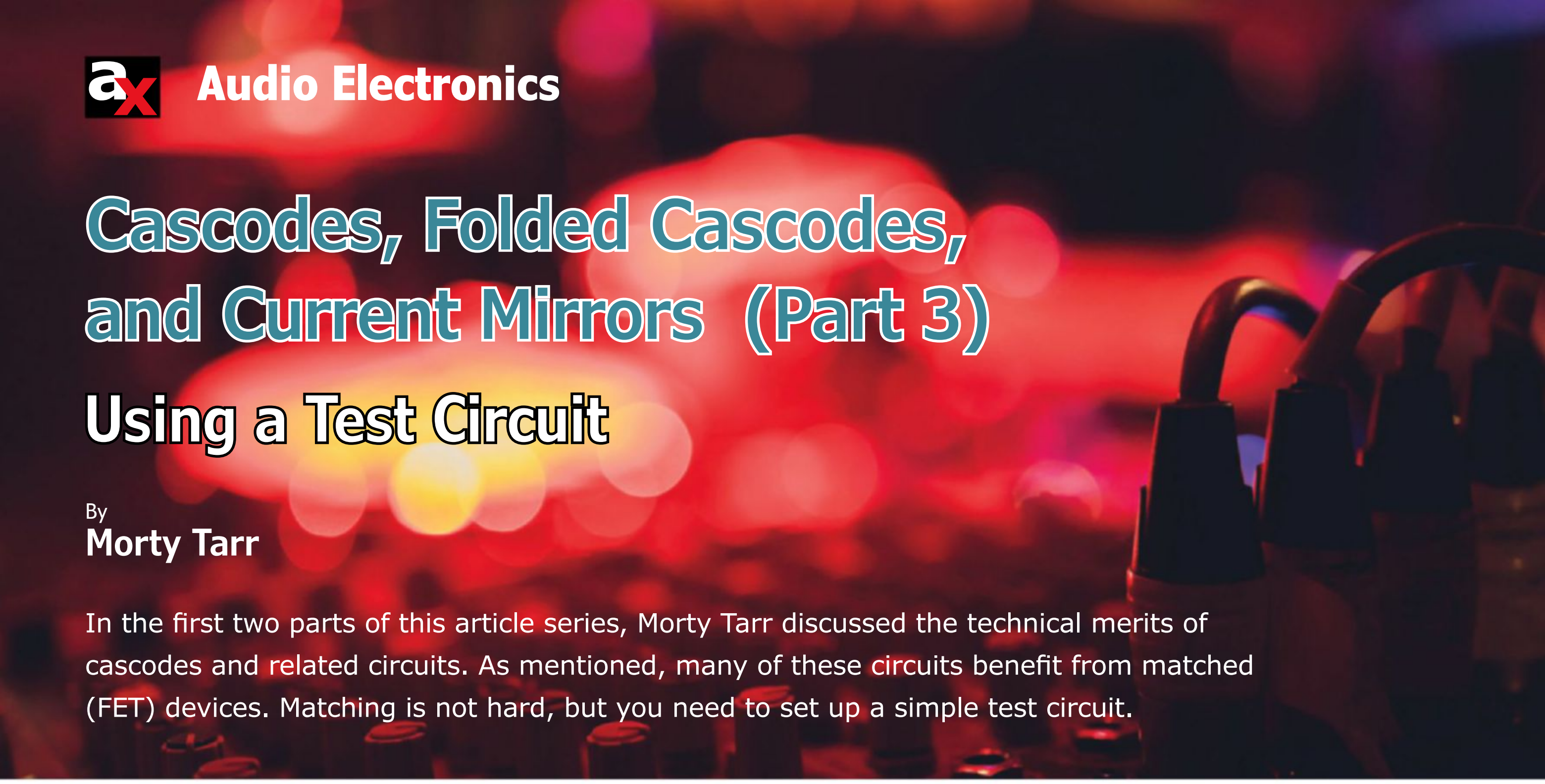
Johnson-Nyquist noise, https://en.wikipedia.org/wiki/Johnson%E2%80%93Nyquist_noise

Nickel-cadmium batter, https://en.wikipedia.org/wiki/Nickel%E2%80%93cadmium_battery

Nyquist frequency, https://en.wikipedia.org/wiki/Nyquist_frequency

Power factor, https://en.wikipedia.org/wiki/Power_factor

Power supply rejection ratio (PSRR), https://en.wikipedia.org/wiki/Power_supply_rejection_ratio



Cascodes, Folded Cascodes, and Current Mirrors (Part 3)

Using a Test Circuit

By
Morty Tarr

In the first two parts of this article series, Morty Tarr discussed the technical merits of cascodes and related circuits. As mentioned, many of these circuits benefit from matched (FET) devices. Matching is not hard, but you need to set up a simple test circuit.

Matching Components

Many circuits have improved performance when certain semiconductor components are matched. Matching involves controlling certain parameters while measuring another parameter and selecting device pairs, triples, or quads based on these measurements. For example, we can set up a MOSFET to have a V_{DS} of 10V and an I_D of 10mA; measuring V_{GS} under these conditions. This is how we could match the current mirror MOSFETs for the amplifiers.

The JFET pairs should be matched for I_{DSS} and V_{GS}. This will minimize both distortion and offset voltage. Fortunately, matching for V_{GS} at the operating point (I_D) is sufficient as V_{GS} and I_{DSS} are related for JFETs:

$$I_D = I_{DSS} \times \left[1 - \frac{V_{GS}}{V_P}\right]^2$$

When we match for V_{GS} at I_D, we are also matching for I_{DSS}. For a current mirror, it is advantageous to match the MOSFETs in the mirror. I use a simple matching fixture that only works for MOSFETs and BJTs. It is not for JFETs, as it has limited gate voltage range.

Matching Fixtures for MOSFETs and BJTs

In **Figure 20**, Q1 and Q2 are the devices to test. It is okay to just use one test socket at a time. Q3 is for the AD711, which had an output phase shift when starting up. C1–C4 prevent oscillations (this circuit has lots of feedback), reduce noise, and stabilize the measurement.

I use this fixture with a single 20V supply, which means there will be 10V across each MOSFET. With R1 and R10 = 1kΩ, there is 10mA I_D current. I measure V_{GS} for each MOSFET and tape the devices to a sheet of paper where I can write down the measured parameter.

Note that as the device is powered up, the die temperature is changing and the measurement will be changing as well. I wait 10 seconds after power up and take the measurement. The value is still changing at that point, but it has worked for me so far.

If we standardize on a 20V power supply, current through the device under test (DUT) can be changed by changing resistors R1 and R10. The voltage across the DUT is set by the divider R3 and R4. The ratio can be changed to change the voltage across the DUT. If you do this, you will need a separate divider for the N-Channel and P-Channel portions of the test jig. For more information, read Nelson Pass' article on device matching (see Resources).

There is a simple way to match JFETs for I_{DSS}. Referring to **Figure 21**, we measure across R1 (A to B), and multiply the voltage by 10. This number is I_{DSS} in milliamps. We can adjust V1 to get 10V across the JFET (or whatever voltage we need). I have found that running JFETs at about 10V V_{DS} puts them at a good operating point for the cascode and the current mirror amplifiers.

In general, a 3-1/2 or 4-1/2 digit digital voltmeter (DVM) will work to make these measurements. Either way, you will see the measurement drift as the die heats up.

We are looking to match devices, so any DVM can be used (calibration is not important), so long as its input capacitance does not make the circuit oscillate. Our goal is to use two (or three or four) devices that measure the same, or close to the same.

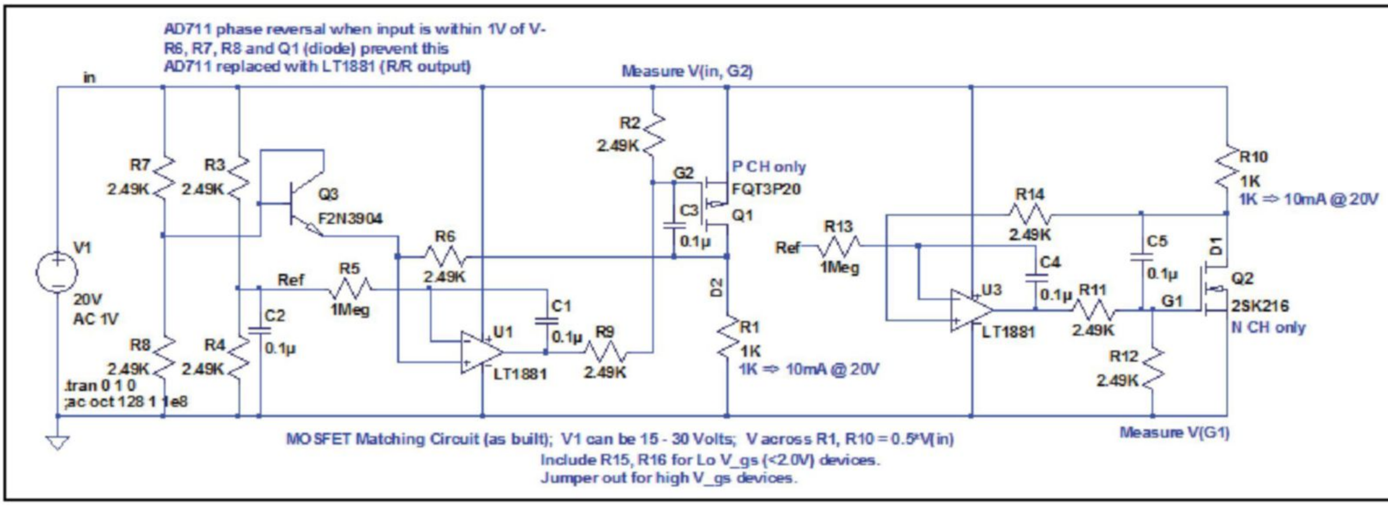


Figure 20: Q1 and Q2 are the devices to test.

I set the supply voltage so the voltage across the JFET is 10V. With a 10mA I_{DSS}, the voltage at point A will be 11V.

A Case History: the Blowtorch Line Stage

To provide a little background information, my research for the line stage designs discussed in Part 2 of this article started about 10 years ago. I discovered Dimitri Danyuk's compilation of John Curl's publications and blog posts, "Condemnation without Examination is Prejudice" online (see Resources).

After some reading, I became determined to hear a Blowtorch line stage. Since John Curl was no longer building them (and he wanted in excess of $10,000 to purchase one), my option was to make one myself. There was plenty of information in the document, and much more in the Blowtorch threads on the diyAudio website.

At some point, I had enough information to re-create the design. I also decided I wanted to meet John Curl. We actually met at CES about a year later. I was able to visit him in California and he gifted me several pairs of Toshiba 2SK389/2SJ109, which were matched to each other. I use those devices to compare the results obtained with the Linear Systems and other devices to insure that performance is not degraded.

Six years ago, I had PCBs fabricated and built my version of the Blowtorch line stage. I used IRF510 and IRF9510 MOSFETs as John did, and I was disappointed to get distortion in the -50dB range even at output levels of 2V to $4V_{PP}$, indicating a problem. A quick call to John resulted in the additional information that IRF P-Channel devices were not linear enough. John had used Fairchild MOSFETs, and I needed to purchase a curve tracer.

Fairchild (now OnSemi) no longer made IRF510s and IRF9510s. But the company had the FQP3N30 and the FQP3P20, which have similar specifications. That made a significant difference, as these devices were much more linear. Now I had distortion better than -80dB.

I thought I could improve the design further (my goal was distortion products better than -90dB, which is 0.0032%). This involved converting the resistor current sources to active current sources and experimenting with different MOSFETs. Using active current sources, I was able to reduce the current in the output stage to 10mA (John's design has about 40mA in the output stage). I suspect the reason for the higher current is to minimize the change in V_{GS} with signal level,

By trying different combinations of FQP3N30 and FQP3P20 devices, I could get distortion products as good as -88dB, but I was unable to determine which characteristics resulted in this performance. Exicon lateral MOSFETs allowed me to get to about -93dB distortion products (0.0022%), but it seemed excessive to use 200W lateral MOSFETs as devices dissipating only 250mW. This is the best distortion performance I have been able to achieve so far. I have also achieved -93dB with Microchip (formerly Supertex) VP2450 and VN2450 MOSFETs, and Diodes, Inc. (formerly Zetex) ZVP4424A and ZVN4424A MOSFETs.

My measurement system is not state of the art, consisting of a HP339A analog THD+N analyzer, which has the meter output connected to an HP 3580A spectrum analyzer. The spectrum analyzer shows the distortion products present, and I look for distortion to be limited to second and third harmonics until the signal gets close to clipping. Using an Audio Precision would likely yield better

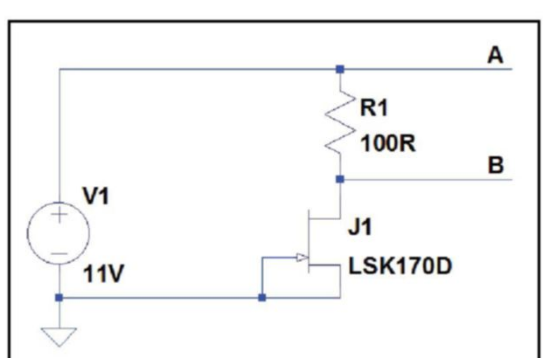
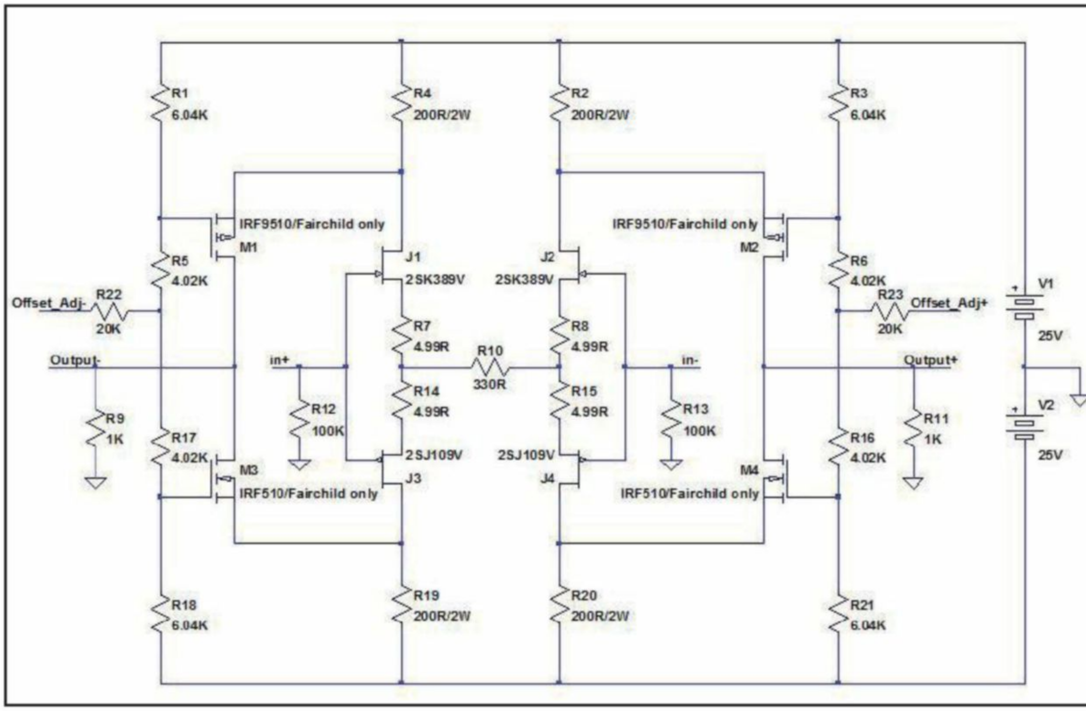


Figure 21: This is a simple way to match JFETs for I_{DSS}.

Figure 22: This is a schematic showing John Curl's circuit for the Blowtorch line stage.



results, as the fundamental could be fully removed and the noise could (optionally) be discounted. Also, it would make the measurement sequences much faster to acquire and record.

Starting with the Blowtorch

Figure 22 shows John Curl's circuit for the Blowtorch line stage. (It originally appeared in the diyAudio Blowtorch thread and on page 88 of the "Condemnation without Examination is Prejudice" article.)

John Curl invented this topology, and it is brilliant! JFETs J1 and J3 create a constant current source that is also the input buffer. Similarly, JFETs J2 and J4 create a constant current input buffer. The difference between in+ and in- is impressed across R10, the gain resistor.

The biasing resistors, with ±25V supplies, puts the sources of the MOSFETs at about ±15V, which sets the current in the 200Ω resistors to about 50mA. This equates to 10mA for the JFETs and 40mA for the MOSFETs. Note that the input resistors, R12 and R13, can be 100kΩ or higher. An unused input should be shorted to ground (for lowest noise and offset).

My circuit design for the Blowtorch line stage substitutes current sources for the 200Ω resistors. I found I could get better distortion with lower current, so my current sources are 20mA (10mA each for the JFETs and MOSFETs).

This design is the origin of the circuits presented in Part 2 of this article series (*audioXpress*, November 2021). As it is very expensive to get sufficient quantities of JFETs to match the I_{DSS} of the N-Channel and P-Channel JFETs to each other today, I modified the design so that matching P-Channel to N-Channel devices is less critical. My design has the additional benefit of creating twice the current in the gain resistors, as the current is created between the N-Channel JFETs and also between the P-Channel JFETs vs. once in John Curl's design.

Note that if you can tolerate some additional noise, MOSFETs can be used in place of the JFET input devices that are shown in Figure 17 and Figure 19 from Part 2 of the article series. The MOSFET N-Channel and P-Channel pairs should be matched for

About the Author

Morty Tarr has more than 30 years' experience in Electrical Engineering. Morty graduated from Cornell University in 1972 with a Bachelor of Arts in Physics, and a minor in Electrical Engineering. He continued his studies at Tufts University in Electrical Engineering.

Morty began his career at recording studios in New York, NY, and Boston, MA. Then he became interested in the design of electronic equipment and made the transition from audio to electronic design. He has extensive experience in the fields of video, audio, test and measurement, networking, and wireless (primarily Wi-Fi and Bluetooth). More recently, from 2006 to 2015, Morty led a team at Bose responsible for advanced development for a $2 billion business. He set the direction for the group and for technology strategy spanning wireless, digital, analog, and system architecture across multiple product lines. Prior to Bose, he worked for LTX developing GHz test systems; at Octave Communications developing conference call bridges; and at Avid Technology. Prior to Avid, at Data Translation, he created the first 24-bit resolution A/D for a PC for Chromatography applications, and then led the team that developed the first Media 100 video editing system.

Morty currently has 20 patents and has spent most of his professional career developing first-in-class products. Innovative ideas and new concepts come naturally to him.

V_{GS} at 10mA, but the N-Channel and P-Channel devices do not have to match each other. Distortion performance is about equal to the JFET input stage (**Photo 1**).

Inverting vs. Non-Inverting Amplifier Stages

Referring to the two line stage designs in Part 2 of the article series, the Cascode Line Stage is inverting as the signal connected to in+ produces out- directly. This means that if we use the amplifier as a single-ended stage (in and out), our output will come from the side of the amplifier whose input is grounded. While this may seem like a bad idea, it is actually good in the same way a cascode is good. There is no change in the gate-drain capacitance of the JFETs whose gates are grounded, which results in marginally better performance. The difference is on the order of 1dB to 2dB better distortion, which is not necessarily audible, but is measurable. The Current Mirror Line stage is non-inverting. The advantage here is that we can choose to have gain, and/or use simpler inverting integrators for DC servos.

There are several things I'd like to point out about building these circuits, should you decide to do so.

LED Strings: I have built a number of these circuits. While assembly techniques usually call for assembling the lowest profile devices first, and moving to higher profile devices in order of their height above the PCB, I have found that this may not be the best way to assemble this design.

I install the LED bias strings first. With just the LEDs and their biasing resistors installed, I apply power to the PCB. All the LEDs should illuminate! If some do not, there may be an LED inserted backward or a defective device. I have found defective devices to be very rare.

In order to find a backward LED, use a clip lead to connect a resistor to ground (the value is not critical, I use 4.99K). Starting at the LED closest to the power supply, touch the other lead of the resistor to each LED in turn until you find one that does not illuminate. That will be the backward device. Note that the LED strings should be bypassed with capacitors. This is not shown in the schematics detailed in Part 2 of the article. I used 1µF film caps for this.

If at any time during debug and bring-up there is an LED that does not illuminate, this indicates a problem. Once the amplifier is fully assembled, it is harder to find a backward LED.

Power Dissipation: When we design circuits without global feedback, it is important to be careful with regard to device power dissipation. It is best to limit the dissipation to one-third, or at most, one-half of the device rating. This helps each device maintain a constant temperature and minimizes drift from changing device temperature. We do not want device temperature to change once it is warmed up, and especially to not change with input signal.

The reason power dissipation is more important without global feedback is that many devices change value (even slightly) with changing temperature. In a feedback amplifier, the feedback will correct for minor changes with minimal impact on performance. Without feedback, the value is not "corrected." If we think of a simple single stage amplifier, imagine the collector resistor changing value with signal level due to power dissipation changes.

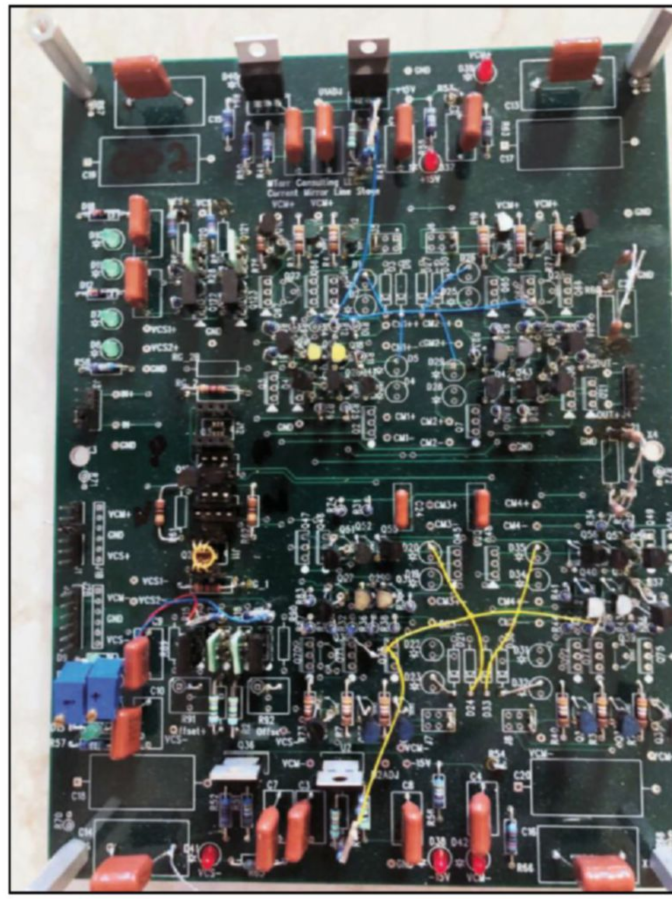


Photo 1: The current mirror line stage PCB. Three copies of this board were built, all worked very similarly. The first one has sockets for all the semiconductors; 12 sets of devices were run through this board to verify performance. The PCB shown only has sockets for the JFETs (J11 and J12, at left). Note that the MOSFETs are color coded based on matching data. These PCBs have wires and other modifications, as most prototype PCBs do. The boards were designed to accommodate three different versions of current mirror to determine the type that produces the lowest distortion without feedback.

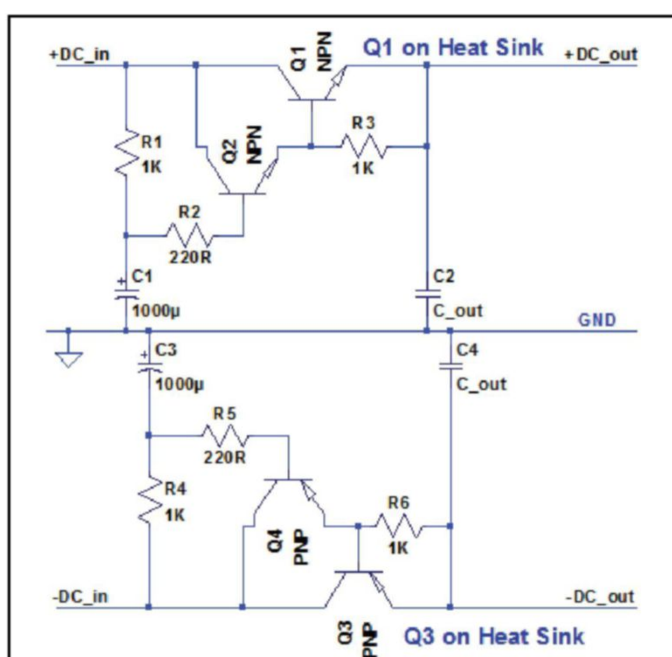


Figure 23: This schematic shows an example of a BJT capacitance multiplier.

This change can even follow the signal. As the resistor value changes, the gain changes, causing distortion. Without feedback, it is prudent to insure that device temperatures stabilize and do not change with input signal.

Power Supplies: In the article, we have not talked about power supplies. Amplifier performance can be affected by the power supplies to which it is connected.

The amplifiers presented in the article do not have a great Power Supply Rejection Ratio (PSRR) as many op-amps do. So the quality of the power supply is a significant concern with regard to amplifier performance.

Our goal for a power supply is to convert the AC line voltage to a DC voltage emulating an ideal battery; that is a DC voltage with virtually no ripple and no noise. Also, we want the power supply to be such that each amplifier is isolated from all the other amplifiers in the system.

There is not room here for a treatise on power supplies for audio, but a few recommendations are in order.

- Be mindful of the noise in the power supply output voltage (Zener diodes are noisy).
- If the output impedance of the power supply is significant, add a capacitance multiplier stage for each individual amplifier to isolate the amplifiers from each other.
- Use heatsinks for power transistors, if required.
- Use good quality capacitors to decouple the power supply at the PCB, and at points where the current draw can change with the input signal.

A capacitance multiplier can be effective for reducing power supply noise and preventing channel to channel crosstalk through the power supply. A capacitance multiplier can be made with BJTs or MOSFETs. Capacitance multipliers provide isolation when each circuit section has its own capacitance multiplier. **Figure 23** shows an example of a BJT capacitance multiplier.

A BJT-based capacitance multiplier has a voltage drop of about 1.5V. Q1 and Q3 are devices that handle the full supply current of the amplifier, and may need to be attached to heatsinks (depending on the current draw). Q2 and Q4 are small signal transistors, included to insure there is sufficient current gain.

R1 and C1 filter the input DC signal. The filtered signal is applied to the base of Q2 and then to the base of Q1 via Q2. R2 is present to ensure that the emitter followers do not oscillate. R2 should be located close to Q2. **Figure 24** shows an example of an MOS capacitance multiplier.

In the MOSFET version, we do not need Q2 and Q4 as the impedance at the MOSFET gate is very high. R2 and R5 should be located close to the gates of Q1 and Q3 to prevent oscillations. Since MOSFETs have a 3V to 5V differential from gate to source, we include voltage dividers R1/R3 and R4/R6 so the devices do not saturate. The voltage divider technique could be used with the BJT capacitance multipliers as well.

There are experts who advise that the output capacitors should not be electrolytic capacitors, but should be only high-quality film

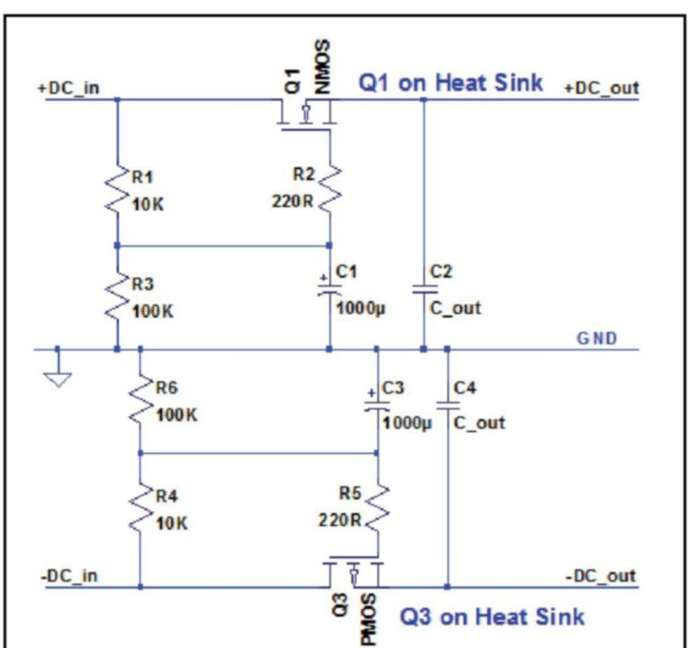


Figure 24: This schematic shows an example of an MOS capacitance multiplier.

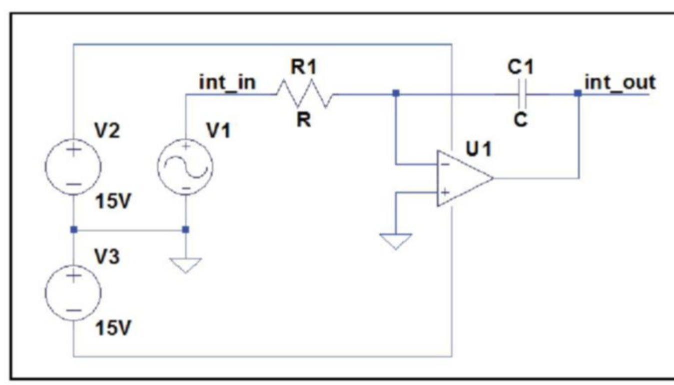


Figure 25: This circuit compares the input voltage to GND, and the op-amp output moves to keep node int$_{in}$ at GND.

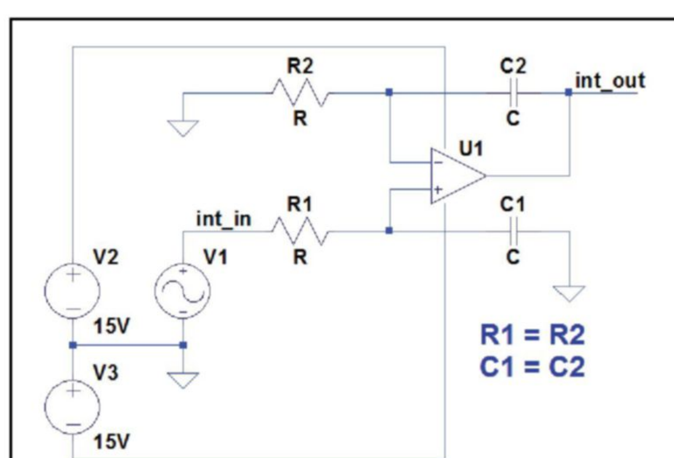


Figure 26: If our circuit is inverting, then we might need a non-inverting integrator (shown here).

capacitors. If you do this, the decoupling capacitors in the amplifier following the capacitance multiplier should be equally high-quality film capacitors.

Inverting and Non-Inverting Integrators

Integrators are important tools for finding average values. We use integrators as low-pass filters for DC servos in amplifiers. The basic idea is that the input of a servo is connected to the amplifier output. The output of the servo then represents the average or DC part of an amplifier output. By feeding this back to a suitable point in the amplifier, the servo can null the amplifier DC output. Amplifiers can be inverting or non-inverting so sometimes you need an inverting servo, at other times a non-inverting servo. Both types are discussed next. An inverting integrator is the simplest to make.

For the circuit shown in **Figure 25**, the time constant is R1 × C1 and a −3dB point of $F = 1/(2 \times \pi \times R1 \times C1)$. For R1 = 1 MΩ and C1 = 1µF, we have a time constant of 1 second and a -3dB point of 0.167Hz.

Note that the circuit shown in Figure 25 compares the input voltage to GND, and the op-amp output moves to keep the node int_{in} at GND. In this inverting integrator, int_{out} moves in the opposite direction of int_{in}.

If our circuit is inverting, then we might need a non-inverting integrator. For a non-inverting integrator (**Figure 26**), the time constants R1 × C1 and R2 × C2 must match. The time constant is equal to R1 × C1 as in the inverting case. The -3dB point is $F = 1/(2 \times \pi \times R1 \times C1)$, also the same as the inverting case. For the non-inverting integrator, int_{out} moves in the same direction as int_{in}.

In Conclusion

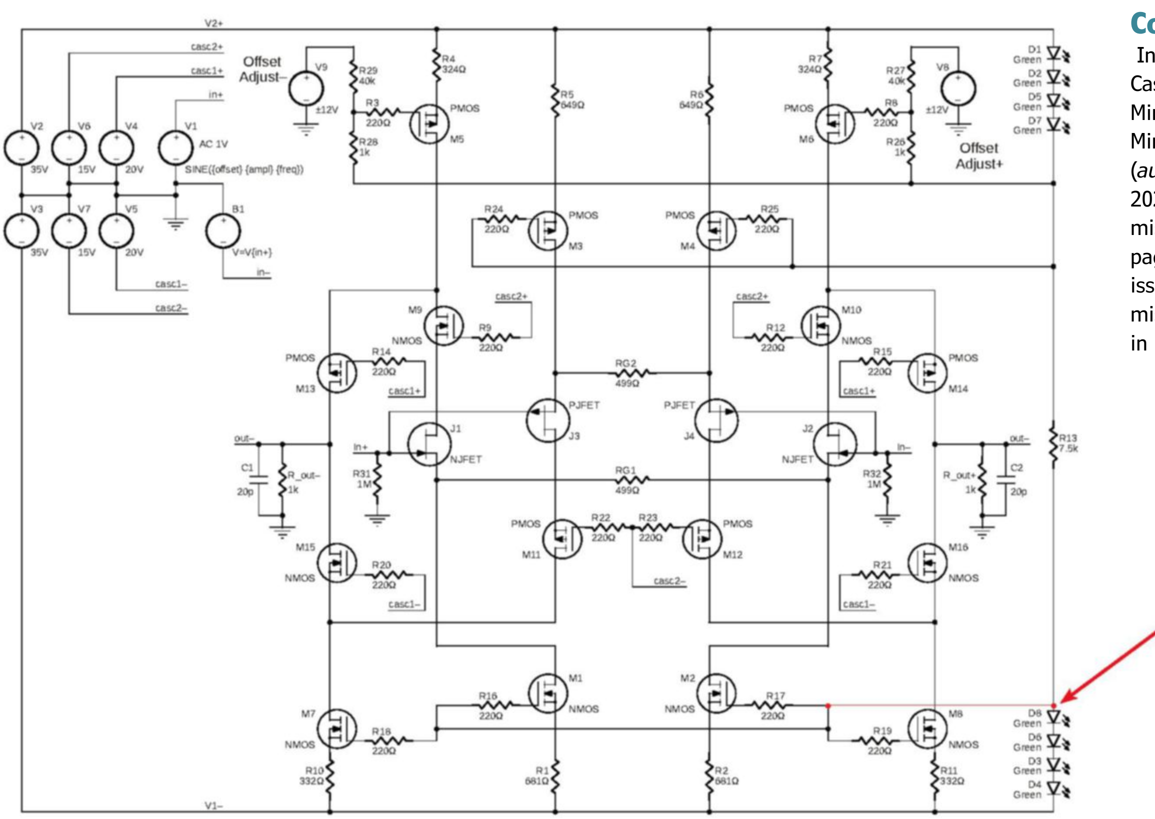
This three-part series is a brief description of some of the important aspects of amplifier design. It is by no means a comprehensive review of everything we need to know about amplifier design.

Resources

Blowtorch threads, diyAudio website:
Part 1: www.diyaudio.com/forums/solid-state/71189-john-curls-blowtorch-preamplifier.html
Part 2: www.diyaudio.com/forums/the-lounge/146693-john-curls-blowtorch-preamplifier-ii.html
Part 3: www.diyaudio.com/forums/the-lounge/318975-john-curls-blowtorch-preamplifier-iii.html

J. Curl, "Condemnation without Examination is Prejudice," May 2006, http://jockohomo.net/data/johncurl-v.0.1.pdf

N. Pass, "Matching Devices," Pass Laboratories, Inc., www.passdiy.com/project/articles/matching-devices



Correction:

In "Cascodes, Folded Cascodes, and Current Mirrors (Part 2): Current Mirrors and Line Stages (*audioXpress*, November 2021), there's a connection missing in Figure 17 on page 53 of the November issue of *audioXpress*. The missing connection is shown in red.

Simple Xtal-Based Motor Control for Turntables

The author proposes a special circuit for motor control using an Arduino microcontroller board that is likely to appeal to more technical DIYers and vinyl enthusiasts, as it offers a great way to deal with issues in mains frequency for critical record listening.

By
Michel Nieuwenhuizen

Photo 1: The finished prototype, built in an old external hybrid hard drive (HHD) enclosure.

Many turntables sold today are driven by a simple 115V/230V synchronous motor. Synchronous motors take their rotational speed from the frequency of the AC power that drives it. In simple (but by no means inferior) turntables, the power frequency is simply the mains frequency of 50Hz or 60Hz. Changing the speed from 33rpm to 45rpm is done by manually rearranging the drive belt to another pulley—examples include some turntables from Rega and Pro-ject [1].

Normally the mains frequency is quite stable, but not stable enough for critical record listening. Therefore, in more expensive units a special circuit derives the same 50Hz or 60Hz from an Xtal (crystal oscillator) reference. This also solves the speed change inconvenience, because the motor speed can be changed by changing the driving frequency.

Retrofits are available from some brands and types of turntables, but they tend to be expensive. Building yourself one from scratch is rather complex, but with some ingenuity and the use of existing boards, it is possible to assemble a perfectly working Xtal-based drive unit with very little effort. **Photo 1** shows the results of my efforts.

The Components

The Arduino microcontroller board is cheap, simple to program and, important for this project, there are many small daughter boards that extend the useability. My eye fell on a so-called DDS board based on an the AD9833 chip. This is a Xtal-based sine/triangle/square generator that can be accurately programmed to any frequency between a few Hertz and a few megahertz. The output level is approximately 200mV. The board's size is less than one square inch and it costs a few dollars (or Euros).

Control software for the Arduino is available at: https://github.com/Billwilliams1952/AD9833-Library-Arduino.

So for a few dollars/euros, we can have the stable 50Hz/60Hz reference we want. But how do we get from the 200mV output of the AD9833 to the 115V/230V needed? My solution is to use a small but powerful Class-D audio amplifier to amplify the signal to approximately 6VAC and then use a small mains transformer backward. So a 115V/230V input to 6V output becomes a 6V input to 115V/230V output. These amplifiers are available on the Internet for

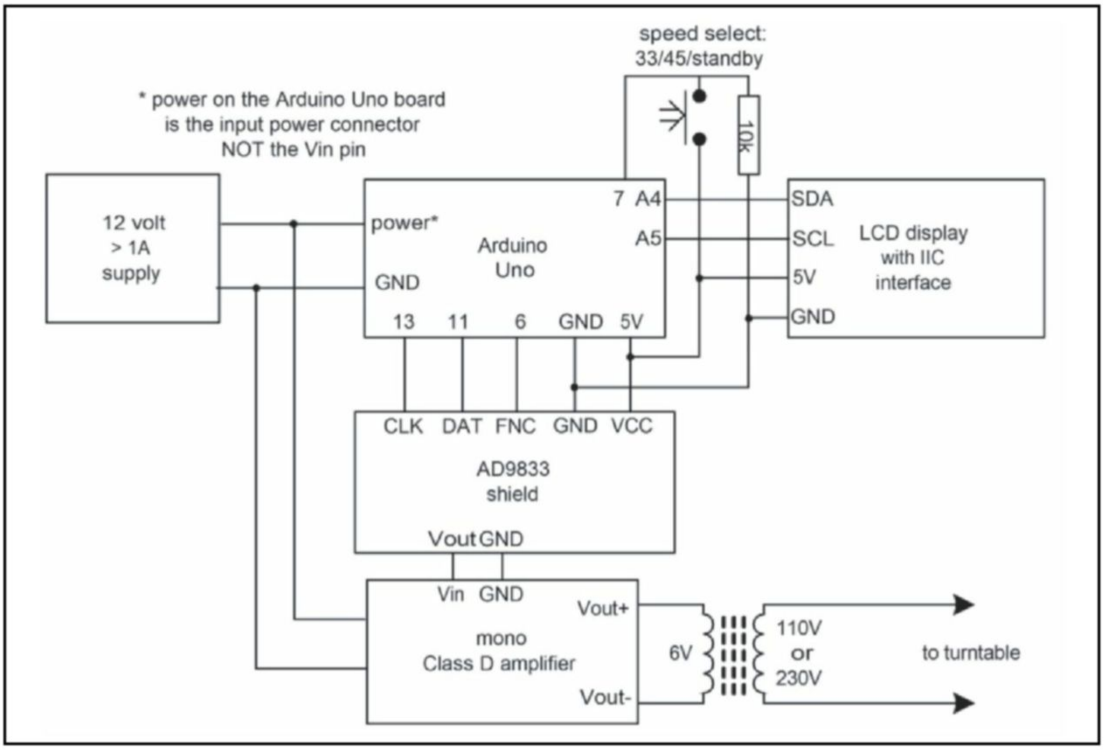


Figure 1: The complete circuit for the motor control

as little as $9. I chose a mono board based on the TPA3116. It runs on 12VDC and has more than enough power for my purpose.

Control is via a single pushbutton switch and a small LCD. A two-row 16-character LCD sells for about $3 online, including a sub-board for IIC control, which simplifies the connections to the Arduino board.

Software to control this board is available in the standard Arduino Integrated Development Environment (IDE). I wrote software that reads the pushbutton status and toggles through 0, 33.3rpm, and 45rpm, and it displays the status on the LCD.

The Arduino Uno board accepts 12V at its power input and a small built-in, step-down converter supplies 5V to the Arduino chip and has enough power to also feed the AD9833 and the LCD. Since the amplifier also needs 12V, we only need a 12V supply with an approximate 2A rating for the whole circuit. I used an old external hardrive supply that I had lying around.

The Circuit

This means that the complete circuit becomes extremely simple as shown in **Figure 1**. The circuit has only four discrete components and is extremely simple to build. The C++ source code of the software is only 108 lines long and can be downloaded from the Supplementary Materials section of the *audioXpress* website (see Project Files for the link). Uploading the code to the Arduino is also simple: Download and install the Arduino IDE, add the AD9833 library, and you can compile and upload the software with a few clicks.

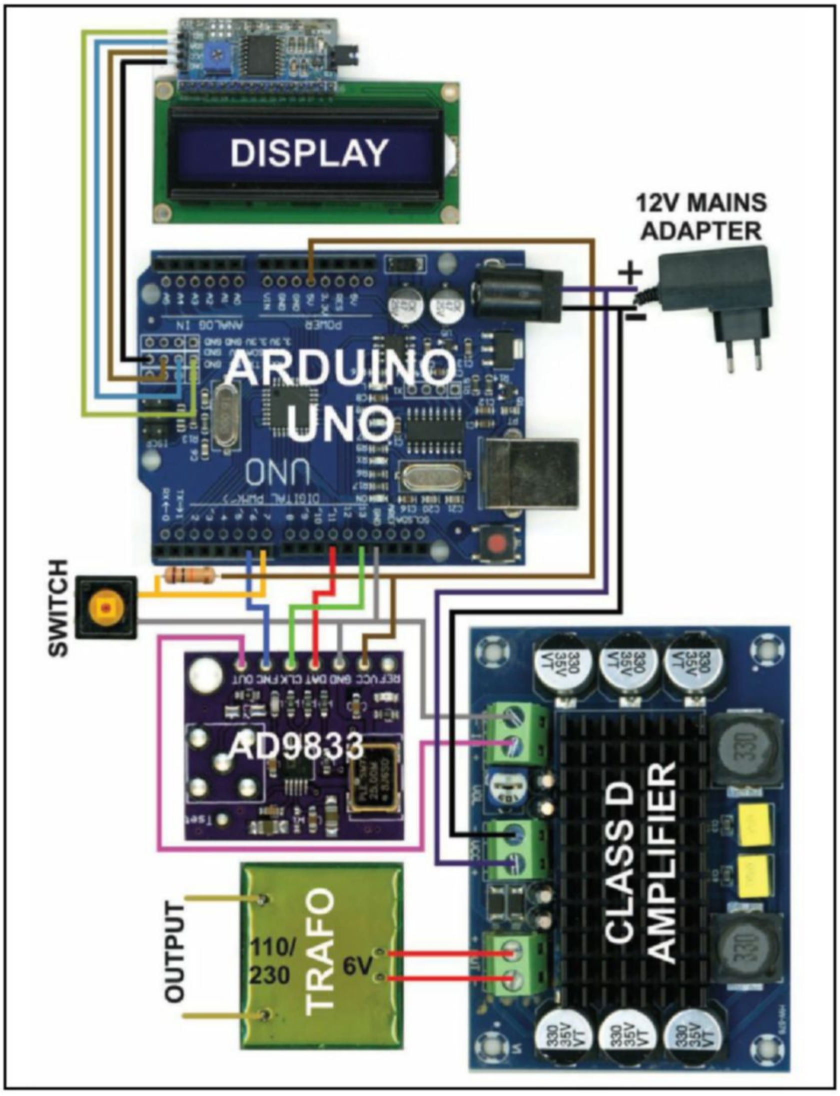


Figure 2: This is how simple the system is in practice.

Alignment is simple, but it turns out to be quite critical. After this you only have to set the 115V/230V appropriate output voltage by means of a potentiometer at the input of the amplifier. The output is not critical.

In my case, the output voltage is about 195V (instead of the ideal 230V), but this has no discernible impact on the performance of the turntable apart from taking a little more time to spin up. Be careful that the amplifier is not overdriven because (at least with my amplifier) at that point the amplifier behaves strangely.

The potentiometer was, in my case, integrated on the amplifier board. If it is not on your board, add a 10kΩ trim pot between the AD9833 and the amplifier yourself.

If you have a stroboscope disc, you might check and, if necessary, fine-tune the turntable speed. I found that my Rega turntable had a very slight speed error, which I corrected by changing the frequency setting in the software.

In **Figure 2** I tried to illustrate the simplicity of the construction. Note that the boards on the figure are not to scale (e.g., the AD9833 board is much smaller). Remember that a synchronous motor either runs at the input frequency determined revolutions per minute (rpm) or not at all!

About the Author

Michel Nieuwenhuizen studied electronics at the Eindhoven Technical University in The Netherlands. He then worked for 27 years with Philips Electronics on audio and video processing, resulting in a number of patents on video image improvement. After his retirement he concentrated on audio projects such as a software package for automation of local radio stations called Caban and microphone techniques.

Project Files

To download the code, visit: http://audioxpress.com/page/audioXpress-Supplementary-Material.html

Footnote

[1] Some recent models have a 24V motor driven from a 115V to 230V to 24V transformer. However, the principle is the same.

Appendix: The code

```
/*
  SynchroDrive.ino
  2021 M Nieuwenhuizen
  Library code of the AD9833 driver found at: https://github.com/Billwilliams1952/AD9833-Library-Arduino
  other libraries are standard components
*/

#include “AD9833.h”     // Include the library
#include “LiquidCrystal_I2C.h”

#define FNC_PIN 6

const int buttonPin   = 7;
int buttonState   = LOW;
int buttonState1  = LOW;
int buttonState2  = LOW;
int first         = true;
int freq33        = 50;    //change to 60 for 60 Hz countries
int freq45        = 67.6;  //change to 81.1 for 60 Hz countries
int speed    = 0;

LiquidCrystal_I2C lcd(0x27,20,4);
AD9833 gen(FNC_PIN);

void setup() {
    gen.Begin();

    lcd.begin();                      // initialize the lcd
    lcd.noBacklight();

    // no output yet, first all voltages must stabilize
    gen.ApplySignal( SINE_WAVE,REG0,50 );
    gen.EnableOutput(false);    // Turn ON the output - it defaults to OFF
    speed = 0;
```

```
        pinMode(buttonPin, INPUT);
    lcd.setCursor(0,0);
    lcd.print(" CABAN  digital ");
    lcd.setCursor(1,1);
    lcd.print(" initialising  ");
    lcd.backlight();

    //wait for stabilized voltages
    delay(4000);

    //now we can start to 33 rpm
    gen.ApplySignal( SINE_WAVE, REG0, freq33 );
    gen.EnableOutput(false);   // Turn OFF the output for now
    lcd.setCursor(1,1);
    lcd.print("   standby    ");
    speed = 0;

    // intialisation is ready
 }

void loop() {
  //debounce
  buttonState1 = digitalRead( buttonPin );
  delay(50);
  buttonState2 = digitalRead( buttonPin );
  // change status if both recent measurements are the same AND they are not the same
  // as the old button position
  if ( (buttonState1 == buttonState2 ) && ( buttonState!= buttonState2 ) )  {
    buttonState = buttonState2;
    // toggle the frequency
    if( buttonState == HIGH )  {
      switch( speed )   {
        case 0:   {
          // go to 33
          gen.ApplySignal( SINE_WAVE, REG0, freq33 );
          gen.EnableOutput(true);   // Turn ON the output - it defaults to OFF
          lcd.setCursor(1,1);
          lcd.print("speed = 33 rpm");
          speed = 33;
          break;
        }
        case 33:   {
          // go to 45
          speed = 45;
          gen.ApplySignal( SINE_WAVE, REG0, freq45 );
          gen.EnableOutput(true);   // Turn ON the output - it defaults to OFF
          lcd.setCursor(1,1);
          lcd.print("speed = 45 rpm");
          break;
        }
        case 45:
        default:  {
          // go to off
          speed = 33;
          gen.ApplySignal( SINE_WAVE, REG0, freq33 );
          gen.EnableOutput(false);   // Turn ON the output - it defaults to OFF
          lcd.setCursor(1,1);
          lcd.print("   standby    ");
          speed = 0;
          break;
        }
      }
    }
  }
}
```

Subminiature Tubes and Effectrode Pedals

Subminiature vacuum tubes have an interesting history that is directly linked with portable equipment requirements. Richard Honeycutt talks to Phil Taylor, the founder of Effectrode Thermionic, a company creatively exploring miniature and subminiature vacuum tubes for guitar effect pedals.

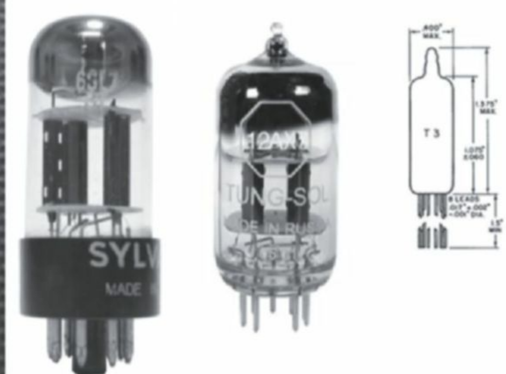



Photo 1: Vacuum tube sizes have decreased through the years: (shown left to right) 0A1 triode—outline ST-14, 6SL7GT dual triode—outline 14C, 12AX7A—outline 9A, and 6021—outline T3.

By
Richard Honeycutt

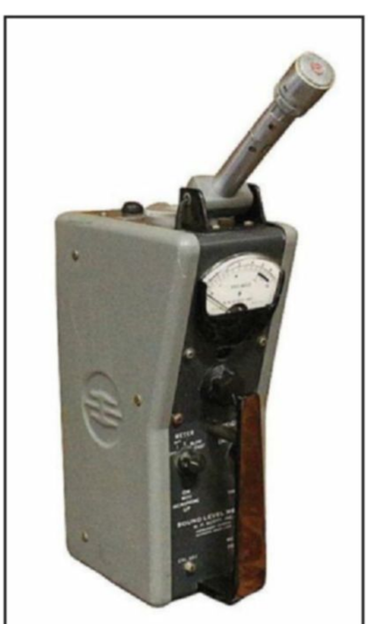


Photo 2: This 1950s-vintage H. H. Scott sound level meter used subminiature triodes.

Everyone knows that the history of electronics is one of continuous miniaturization. **Photo 1** shows four vacuum tubes: the 01A introduced in the early 1920s, the 6SL7GT introduced in 1941, the 12AX7 introduced in 1947, and the subminiature tubes (e.g., the 6021) introduced in the early 1940s. So why was the miniature tube (with a 9A outline) introduced after the 6021? Herein lies a tale!

The Reason Behind Miniaturization

Much of the impetus for miniaturization of tubes came from the consumer market. This has been true for a long time. Table radios with limited bass response replaced floor-standing models because of the reduced size. The cassette tape player replaced the reel-to-reel in consumer applications because of size, weight, and cost. Transistorized hi-fi equipment began replacing tube units even before the reproduction qualities were anywhere near comparable. Even today, many people choose to put up with limited bass and sound level so that they can listen to music on their cell phones or laptop computers rather than use a musically superior, but larger, stereo system.

However, World War II brought a more urgent consideration: military aircraft became increasingly dependent upon avionics, and size, weight, and reliability were crucial! Raytheon responded to this need by developing a line of subminiature (submini) tubes. Being about the diameter of a pencil, they were also known as "pencil tubes." These tubes were designed to meet the MIL-E-1 specification for reliability. They were designed for use at high temperatures and high altitudes, and for very low microphonics. They could withstand up to 20,000G shocks (a 3′ to 4′ drop onto a concrete floor).

In addition to avionics, submini tubes were used in missile guidance systems. The small size and tight spacing of the elements suited them for use up to 400MHz. Since they were primarily sold for military use and extensive (and expensive) testing was required to document reliability and performance, these tubes never made much impact on the consumer market.

The First Submini Tubes

The first submini tubes I ever saw were used in a 1950s-vintage H. H. Scott sound level meter (**Photo 2**). This unit was powered by a 90V "B" battery and a lower-voltage filament battery. Probably the most common consumer product using submini tubes was the hearing aid (**Photo 3**) althoough they were also used in the 1953 Emerson 747 coat pocket radio. One result of these tubes' military heritage is extreme reliability: in the words of a Raytheon datasheet, "...failures became very rare."

Photo 3: This hearing aid used three submini tubes as amplifiers.

Photo 4 shows a few Sylvania subminiature tubes. You probably noticed from Photo 3 that these tubes are not socket-mounted as are most other tubes: instead, they have flexible leads like those of transistors, and were soldered to the circuit board.

While this prevented the vibration-failure mode of tubes jiggling out of their sockets, it also kept them from being user-replaceable for most consumers. This may be one reason for the rarity of their use in consumer products. Although replacement was very seldom needed, consumers were accustomed to expecting tube failures in electronic equipment, so marketing departments may have considered the lack of pluggability a disadvantage.

Table 1 shows a few of the more common submini tubes along with some important characteristics. The 6021 is a medium-mu dual triode (μ=25) roughly comparable to the 12AU7A (μ=19); the 6112 is a high-mu (μ=70) dual triode comparable to the 12AT7 (μ=60) but with a lower gain than the most common high-mu dual triode, the 12AX7A (μ=100). **Figure 1** shows a typical circuit using a 6021 tube. The same circuit could be used with a 6112 tube.

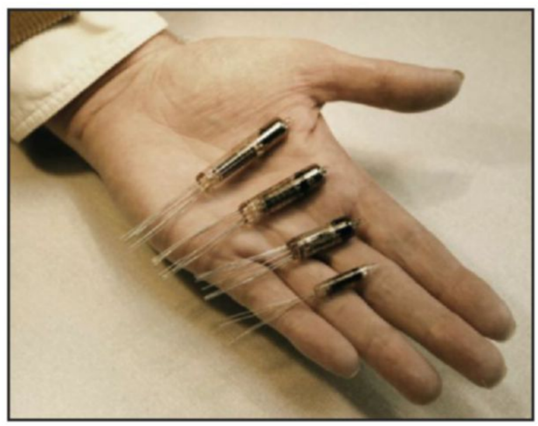


Photo 4: This photo illustrates the size of the subminiature tubes.

Effectrode

In 1996, a new player entered the audio world! British engineer Phil Taylor (**Photo 5**) founded Effectrode Thermionic with the express aim "to build effects that might have been made if the transistor had never been invented. The transistor first became commercially available in the late 1950s. At this time, audio equipment had a certain appearance with simple clean lines, was built to last, could be repaired, and sounded beautiful.

I had the opportunity to interview Phil Taylor for *audioXpress*.

audioXpress: Can you tell me when and how you became interested in tubes and tube audio?

Phil Taylor: I vividly recall when my interest in vacuum tubes was first set alight. It was during the early 1990s when a friend asked me if I'd take a look at his old Rogers "Junior" EL84 monoblock amplifier—the Junior was a lovely little tube push-pull hi-fi amp. Being much younger in those days and always seeking the newest thing in guitar gear,

Tube	Type	Ih (A)	Gm	mu	RaP(W)	Pdiss(W)	eBay	Datasheet
6021	Dual triode	0.3	5.4K	35	6.5K	1.1	6021	6021
6111	Dual triode	0.3	5K	20	4K	–	6111	6111
6112	Dual triode	0.3	1.8K	70	38K	0.3	6112	6112
6814	Triode	0.15	6K	29	4.8K	2.2	6814	6814
5719	Triode	0.15	1.7K	70	41K	1.3	5719	5719
5744	Triode	0.2	4K	70	17K	1.3	5744	–
5840	Sharp cut-off pentode	0.15	5K	1300	260K	1.1	5840	5840
5899	Remote cut-off pentode	0.15	2.5K	650	260K	–	5899	5899
5902	Beam power pentode	0.45	4.2K	15K (Zout 3K)	1	3.7	5902	5902

Table 1: These "high-voltage" subminis (filaments 6.3V and maximum plate voltage >100V) illustrate some of the available types.

	Ebb = 100 Volts							
Rb (megohms)	.047		0.10		0.27		.47	
Rcf (megohms)	.10	.27	.10	.47	.27	.47	.47	1.0
Rk (ohms)	1500	1500	2200	3300	6800	8200	12000	16000
Ib (ma)	.85	.86	.51	.44	.19	.18	.115	.10
Ec (volts)	−1.27	−1.29	−1.12	−1.45	−1.29	−1.48	−1.38	−1.60
Eb (volts)	58.8	58.2	47.9	54.5	47.4	49.8	44.6	51.4
Esig (volts, rms)	.1	.1	.1	.1	.1	.1	.1	.1
Eout (volts, rms)	2.01	2.14	2.04	2.18	2.05	2.1	2.04	2.14
Gain	20.1	21.4	20.4	21.8	20.5	21.0	20.4	21.4
% Distortion	1.4	1.4	1.4	2.4	1.4	2.3	2.4	1.5
Esig* (volts, rms)	.47	.57	.42	.63	.55	.62	.58	.78
Eout (volts, rms)	9.4	12.1	8.5	13.5	11.0	12.8	11.7	16.0
Gain	20.0	21.2	20.2	21.4	20.0	20.5	20.2	20.5
% Distortion	5.0	5.0	4.7	5.0	5.0	5.0	5.0	5.0

	Ebb = 200 Volts							
Rb (megohms)	.047		.10		.27		.47	
Rcf (megohms)	.10	.27	.10	.47	.27	.47	.47	1.0
Rk (ohms)	1000	1200	1500	2700	4700	5600	9100	12000
Ib (ma)	2.05	1.95	1.20	.98	.45	.43	.26	.24
Ec (volts)	−2.05	−2.34	−1.80	−2.65	−2.12	−2.40	−2.36	−2.88
Eb (volts)	101.6	106.2	78.2	99.4	76.9	81.6	75.6	84.1
Esig (volts, rms)	.1	.1	.1	.1	.1	.1	.1	.1
Eout (volts, rms)	2.34	2.42	2.36	2.4	2.3	2.36	2.28	2.29
Gain	23.4	24.2	23.6	24.0	23.0	23.6	22.8	22.9
% Distortion	1.20	1.3	1.3	1.4	1.3	1.3	1.3	1.4
Esig* (volts, rms)	1.15	1.35	.97	1.41	1.15	1.35	1.2	1.6
Eout (volts, rms)	26.8	32.1	22.8	33.5	26.5	31.5	27.2	36.0
Gain	23.3	23.8	23.5	23.8	23.0	23.3	22.6	22.6
% Distortion	5.0	5.0	5.0	5.0	5.0	4.9	4.8	5.0

*Maximum signal for 5% distortion or 1/8 microampere grid current.

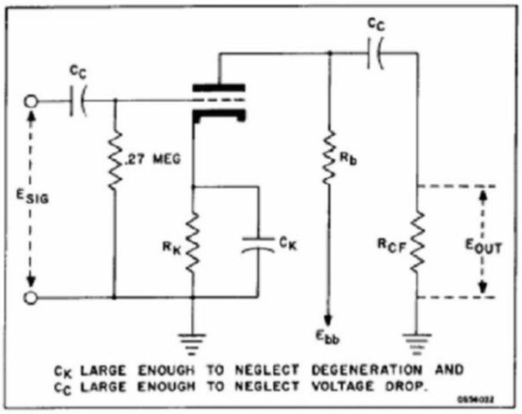


Figure 1: This circuit from the Sylvania datasheet shows a typical small-signal amplifier circuit using one triode of a 6021 dual triode submini tube.

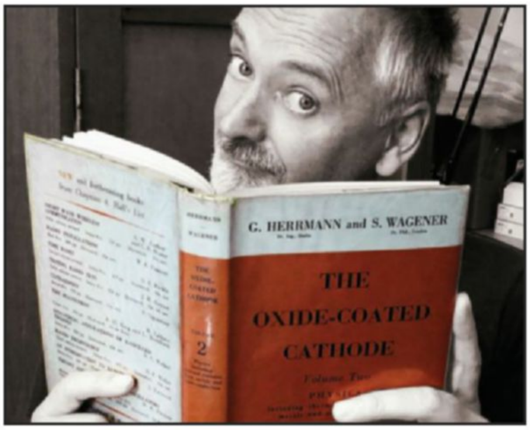


Photo 5: Effectrode's Phil Taylor engages in a little light reading.

Photo 6: The Effectrode Tube Vibe pedal uses miniature tubes.

I remember being terribly skeptical about vacuum tubes. How could this ridiculously under-powered little amp built with such archaic devices possibly sound better than a powerful solid-state MOSFET amplifier housed in a sleek 19″ black anodized aluminum enclosure? But it did—this little amp had a beautifully rich and sweet musical tone full of presence and warmth. Maybe solid-state amps measure well in the lab, but they sounded flat and clinical to my ear.

audioXpress: Can you summarize your career history?

PT: During the early 1990s, I worked as a production engineer at BSS Audio, an innovative maker of professional audio signal processing gear for studios and live venues all over the world. Three years later I moved on to work in acoustics advising on reverberation control and sound insulation in theatres, halls, studios, and derelict office buildings that were being converted into housing. Much of the work was based in central London, and it was an extraordinary opportunity to see the underside of London—the dark and dismal places that had been long forgotten.

After six years I moved on again, this time to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work as a research associate at Cardiff University designing the electronics for marine instruments for collecting water samples and precise measurements of temperatures from the deepest parts of the ocean—a “proper” hardcore engineering job. These instruments were deployed on the mid-Atlantic ridge and in the Arctic Ocean—as you can imagine the electronics had to be super-reliable as any failures would be very costly. I feel fortunate to have been involved in such fascinating work and met some nice people along the way. I left Cardiff with a doctorate in Electrical Engineering and Communication in 2008.

Photo 7: The Effectrode Fire Bottle uses a subminiature dual triode.

Photo 8: This photo of the circuitry of the Effectrode Fire Bottle shows the subminiature dual triode mounted on the circuit board.

audioXpress: How do you decide which products will use miniature tubes and which will employ subminiature ones?

PT: The main unique feature of our pedals is that they’re based on a genuine all-tube signal path operating at real amp plate voltages (300VDC)... in a compact effect pedal. In 2009, we made a jump to using subminiature tubes in one of our pedals. We’d designed and constructed a prototype LA-2A style optical tube compressor pedal based on a 12AU7, but realized we could make our compressor pedal even smaller by utilizing a Sylvania subminiature 6111 tube instead. As long as the circuitry wasn’t too complex and required no more than one tube, we could cram all the electronics inside standard-sized effects pedal enclosure. Other subminiature tube pedals soon followed, including a booster and fuzz pedal.

audioXpress: We published an article in June 2016 about Electro-Harmonix, which now owns New Sensor in Russia. I don’t believe the company makes submini tubes. Are your subminiature tubes custom-made, or is there a factory that produces them?

PT: The subminiature tubes in our pedals are new-old stock (NOS) components manufactured in Sylvania’s Emporium, PA, plant, up until the late 1980s. Additional info on the factory can be found at: www.effectrode.com/knowledge-base/thanksgiving-is-sylvanias-lucky-date and information about the subminiature tubes at: www.effectrode.com/news/fragile-guided-missile-parts-inside.

audioXpress: If the owner of an Effectrode pedal needs to replace a subminiature tube, would (s)he order it from your factory, or are there other sources besides NOS vendors?

PT: We do hold stock of the 6111, the 6112, and the 6021 Sylvania subminiature tubes used in our pedals. There are also a few vendors that carry them.

Photos 6–8 show a couple of Effectrode effects pedals, including the innards of one using a 6112 subminiature dual triode.

A wealth of information about tubes, in general, and subminiature tubes in particular, and Effectrode products can be found at: www.effectrode.com. Also included are sound clips that allow a prospective buyer to hear what each effect pedal will do for the sound of his or her instrument.

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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