

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Market Update

Speakers and Transducers Optimization of the Species

By J. Martins

Electrostatic Pure Silicon Microspeakers for Wireless Earbuds

By Hermann Schenk and Lutz Ehrig
(Arioso Systems)

Bass Shakers and Tactile Sound

By Mike Klasco

Fresh From the Bench

Four Digital Stylus Force Gauges

By Gary Galo

You Can DIY!

The Live-Edge Dipoles

By Perry Marshall

You Can DIY!

Vanderveen Trans-SE10 Secrets

By Menno van der Veen

Getting Started with A2B
Designing and Building Systems
By Brewster LaMacchia

Sound Control
Loudspeaker Radiation
and Directionality
By Richard Honeycutt

Audio Scope
Simple Improvements
to a Recent Audio
MFB Filter Lowers
Harmonic Distortion
By Michael Steffes

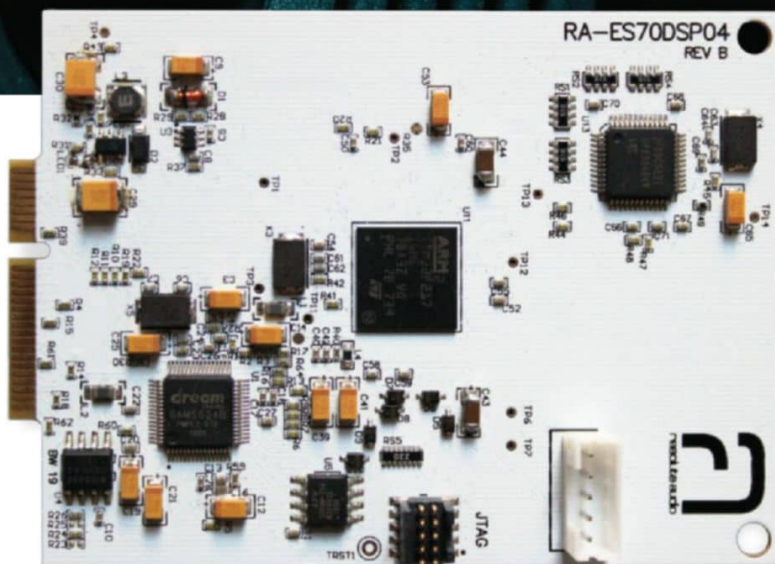
50
Years



JANUARY 2021

AES70

network-controlled
DSP module



AES70 lays the foundation for a multitude of new control apps for monitoring, automation and workflow improvements, and expands the reach of current apps, thereby adding value to AES70 compliant hardware products.

Resolute's **RA-ES70DSP04 DSP module** is easily programmable, so you can add AES70 networked control and monitoring to your next generation product, be that a powered speaker, microphone pre-amp, power amp or small mixer.

The algorithms available for the RA-ES70DSP04 include multiband EQ, Crossover Filters, Vocal Reverb, Compressor, Noise Gate, Spring Reverb and many more. An onboard web server provides rich controls for each algorithm.



You'll find all Resolute products and other audio parts at profusionplc.com

DSP Design Tools

The RA-ES70DSP04 comes with all the software tools to help you program the DSP and implement AES70. In addition, there is an evaluation board that includes 4 channels of analog audio in and out, and 2 channels of AES67 audio in and out, ideal for developing networked audio products.



1140 Series

Studio-Grade Impedance Matching Transformers

Rooted In Solid ENGINEERING

- Microphone Input
- Microphone Splitter
- Line Input
- Line Output
- Direct Box

- Designed with Nickel alloy cores for high performance and Wide Bandwidth
- Frequency Response: 20 Hz – 20 kHz ± 0.35 dB, (ref. @ 1 kHz)
- Low distortion at levels up to +27 dBu @ 20 Hz
- Excellent Shielding – Mu Metal enclosures
- Line Output Transformers are open style (not Mu Metal)





January 2021 ISSN 1548-0628

www.audioxpress.com

audioXpress (US ISSN 1548-0628) is published monthly, at \$50 per year for the US, at \$65 per year for Canada, and at \$75 per year Foreign/ROW, by KCK Media Corp., P.O. Box 417, Chase City, VA 23924, US Periodical Postage paid at Chase City, VA, and additional offices.

Head Office:
KCK Media Corp.
P.O. Box 417
Chase City, VA 23924, US
Phone: 434-533-0246
Fax: 888-980-1303

Subscription Management:
audioXpress
P.O. Box 417
Chase City, VA 23924, US
Phone: 434-533-0246
E-mail: custserv@audioxpress.com
Internet: www.audioxpress.com

Postmaster: Send address changes to:
audioXpress
P.O. Box 417
Chase City, VA 23924, US

Advertising:
Strategic Media Marketing, Inc.
2 Main Street
Gloucester, MA 01930, US
Phone: 978-281-7708
Fax: 978-281-7706
E-mail: audioxpress@smmarketing.us

Advertising rates and terms available on request.

Editorial Inquiries:
Send editorial correspondence and manuscripts to:
audioXpress, Editorial Department
P.O. Box 417
Chase City, VA 23924, US
E-mail: editor@audioxpress.com

Legal Notice:
Each design published in audioXpress is the intellectual property of its author and is offered to readers for their personal use only.
Any commercial use of such ideas or designs without prior written permission is an infringement of the copyright protection of the work of each author.

© KCK Media Corp. 2020
Printed in the US

Celebrating a Double Anniversary

This edition of *audioXpress* magazine should be covered in glitter and gold to properly signal and celebrate our double anniversary, but unfortunately that's not an appropriate mood for the times in which we live. However, it is worth noting that this edition marks 20 years since *audioXpress* was launched. And this is also the 50th anniversary since the publication's founder's company was started!

Many of our readers are extensively familiar with the foundational history of The Audio Amateur company by founder Edward T. Dell. This publishing effort started in 1970 and was the beginning for *Audio Electronics*, *Speaker Builder*, *Glass Audio*, *Multimedia Manufacturer*, and other audio-focused periodicals. Eventually, in January 2001, Ed Dell decided to create a monthly magazine, merging those quarterly publications into *audioXpress*. This means that, in January 2021, we celebrate 50 years from the company's origins and 20 years of *audioXpress*.

Since its foundation, *audioXpress* inherited a long-standing tradition of serving up the best in audio DIY, electronics and innovations—the reason why *audioXpress* magazine adopted the slogan “Advancing the Evolution of Audio Technology Since 1970.”

Fifty years is a remarkable landmark for any company, and particularly for a publishing venture. The publication's team of award-winning and highly recognized authors from around the world are the reason why *audioXpress* has gained such wide recognition from global audio electronics, engineering, and DIY enthusiasts communities like no other publication, and is today recognized by R&D teams and audio product developers and designers worldwide.

Of course, *audioXpress* evolved, and is today much more focused on Audio Development and Design, particularly with its globally recognized website serving daily product and industry updates. Also, *audioXpress'* audience has significantly expanded since 2014, when the magazine embraced its audio development focus and expanded its coverage on audio innovations across all market segments to embrace headphones, wireless audio, voice interfaces, communications, and automotive.

Now reaching many more readers all over the world with its print and digital combined circulation, the majority of the magazine's readers are engineers working in manufacturers' R&D departments, engineers involved in product design and development, manufacturing, systems integration, and consultancy.

For 2021, we have planned an exciting editorial calendar that will also help our readers during these troubled times. With the global pandemic, we've seen that more people are now open to the idea of getting magazines digitally, but we are also planning a tighter convergence among our other media platforms, including our website and e-newsletter. The largest challenge in 2020 and throughout 2021 will continue to be the need to adapt the process of accessing relevant information. With the industry reorganizing without the traditional conventions and trade shows, this has not been easy, but we recognize our readers require even more up to date information—and they rely on us to find it.

In a way, the pandemic did level the field in terms of access to information and it revealed the absurdity of having hundreds or thousands of media sources that are limited to simply replicate information, most of the time with no ability to fact-check it or add value to it. And that's where our direct connection to the industry and our expert network enables us to maintain a firm course.

In these 50 years since The Audio Amateur began, our publications have remained relevant because of the quality of its content, reflecting the interests of a unique community. Today, that community—a strong network of professionals, industry connections, experts and enthusiasts—makes *audioXpress*, and we are confident we are able to offer unique content and a valuable perspective.

J. Martins
Editor-in-Chief

The Team

President:	KC Prescott	Associate Editor:	Shannon Becker
Controller:	Chuck Fellows	Graphics:	Grace Chen
Editor-in-Chief:	João Martins	Advertising Coordinator:	Nathaniel Black

Technical Editor:	Jan Didden
Regular Contributors:	Bruce Brown, Bill Christie, Bob Cordell, Joseph D'Appolito, Vance Dickason, Jan Didden, Scott Dorsey, Gary Galo, Chuck Hansen, Richard Honeycutt, Charlie Hughes, Mike Klasco/Nora Wong, David Logvin, Oliver Masciarotte, Nelson Pass, Christopher Paul, Bill Reeve, Fernando Rodrigues, Michael Steffes, Stuart Yaniger



OUR NETWORK



SUPPORTING COMPANIES

ACO Pacific, Inc.	57	Menlo Scientific, Ltd.	77
Acoustic Sciences Corp.	37	NTi Audio, Inc.	35
All Electronics Corp.	79	Parts Express International, Inc.	84
Audio Precision, Inc.	33	Powersoft S.p.A.	11
Beyma Professional Loudspeaker	75	Profusion plc	2
Clark Synthesis	29	Purifi Audio	81
COMSOL, Inc.	9	QuantAsylum USA, LLC	49
Eastech	25	Redco Audio	53
GRAS Sound & Vibration	45	SB Acoustics	57
Hammond Manufacturing, Ltd.	3	SB Audience	29
Hypex Electronics B.V.	67	Scan-Speak A/S	19
Jensen Loudspeakers	39	Triad Magnetics	83
KAB Electro-Acoustics	79	Warkwyn	37
KLIPPEL GmbH	43	Wavecor	61
Lavoce Co., Ltd.	55		

NOT A SUPPORTING COMPANY YET?

Contact Peter Wostrel (audioxpress@smmarketing.us, Phone: 978-281-7708, Fax: 978-281-7706) to reserve your own ad space in the next issue of the magazine.

The Audio Voice

Sign up now for our **FREE** weekly e-newsletter



Weekly Newsletter for the Audio Industry, Professional Designers and Audio Enthusiasts
Product Design | Audio Electronics | Acoustics | DIY | Innovations in Audio

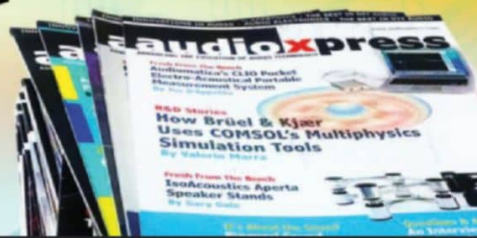


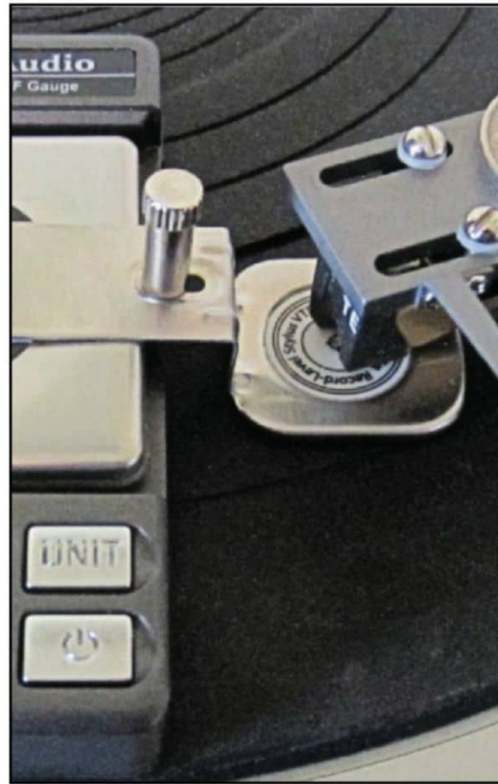


ax Cache

Every issue of audioXpress published since 2001, on USB.

Order at: **cc-webshop.com**





● Features

50 Four Digital Stylus Force Gauges

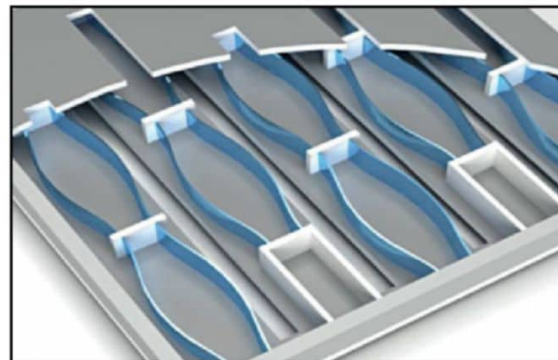
By Gary Galo

If you want to get the most from your turntable, tonearm, and phono cartridge, an accurate stylus tracking force gauge is essential. Many manufacturers refer to “stylus pressure” as “stylus force”—the two terms are interchangeable. Gary Galo tests a few of the gauges currently available.

58 Vanderveen Trans-SE10 Secrets

By Menno van der Veen

Recently, a new tube-amplifier DIY kit was brought to market by Vanderveen (design) and Tentlabs (kit manufacturing). The net result is remarkably detailed sound reproduction where the tubes’ second harmonic and output transformer magnetic distortions are minimized. In this article, the “why and how” of the unusual design are explained by its designer.



68 The Live-Edge Dipoles

By Perry Marshall

Perry Marshall has built almost every kind of speaker you can imagine. One day during a visit to Big Red Sawmill in Palmyra, NE, he had the desire to build the Live-Edge Dipoles. He bought two slabs of birch and brought them home to Chicago, IL. This dipole system achieves Constant Directivity across the entire audio spectrum.

76 Getting Started with Automotive Audio Bus (Part 4) Designing and Building Systems

By Brewster LaMacchia

The first three parts of this article series examined the basics of A2B technology, how data moves between nodes with fixed latency, A2B phantom power, and the chip’s LVDS interface. This article takes a step back and looks at creating systems and the tools that are available to speed up that process. One of the nice things about A2B development is that it is license/royalty free, and ADI offers the software libraries and SigmaStudio design tool at no charge.



● Market Update—Speakers and Transducers

MARKET UPDATE

8 Drivers, Speakers, and Transducers

Optimization of the Species

By J. Martins

In this Market Update, we selected some examples of innovations in speaker technology and highlighted a few of the most recent announcements.

26 Electrostatic Pure Silicon Microspeakers for Wireless Earbuds

By Hermann Schenk and Lutz Ehrig

This article discusses how innovations in microspeaker technology, and specifically a new MEMS topology from Arioso Systems, will contribute to next-generation true wireless earbuds.

30 Bass Shakers and Tactile Sound

By Mike Klasco

From powerful bass shakers that move floors and chairs in movie theaters to subtle bone-conduction transducers and haptic feedback technologies, there is a world of transducer solutions that continue to improve. This article reviews the history and working principles of those shakers and details some notable examples now available in the market.

● Departments

4 From the Editor’s Desk

5 Client Index

● Websites

www.audioxpress.com
www.voicecoilmagazine.com
www.cc-webshop.com
www.loudspeakerindustrysourcebook.com
www.circuitcellar.com
www.linuxgizmos.com

● Columns

SOUND CONTROL

40 Cone Loudspeaker Radiation and Directionality

By Richard Honeycutt

AUDIO SCOPE

46 Simple Improvements to a Recent Audio MFB Filter Lowers Harmonic Distortion

By Michael Steffes



@audioxp_editor



audioxpresscommunity



linkedin.com/company/audioxpress

Drivers, Speakers, and Transducers

Optimization of the Species

By
J. Martins
(Editor-in-Chief)



There's a lot happening in speaker technology. Drivers and transducers are constantly evolving, and innovations are happening not only in design, construction, and materials, but also in integrated auxiliary technologies, able to optimize loudspeakers to their maximum potential performance. In this Market Update, we selected some examples of those innovations and highlighted a few of the most recent announcements.

From the tiniest available drivers to the largest and most powerful subwoofers, not forgetting the rising category of intelligent designs and technologies required by smart speakers, this is a product category in overdrive. Even with the disruptions in the global market caused by the pandemic, manufacturers continue to invest in new developments and are announcing new designs. The home audio and high-end categories are showing clear signs of growth while people spend more time in their homes, but there's also increasing R&D activity in the pro audio sectors, leveraging this partial pause to prepare for a return to market, as soon as installation and live performance activities are back in full force.

There's innovative materials, new topologies, completely new designs and form factors for transducers and drivers of all sizes. And speakers are being supported by optimizations taking place in the circuits directly associated with the signal chain, including feedback loops, and speaker diaphragm position sensors. And then there's increasingly integrated amplifiers and DSP.

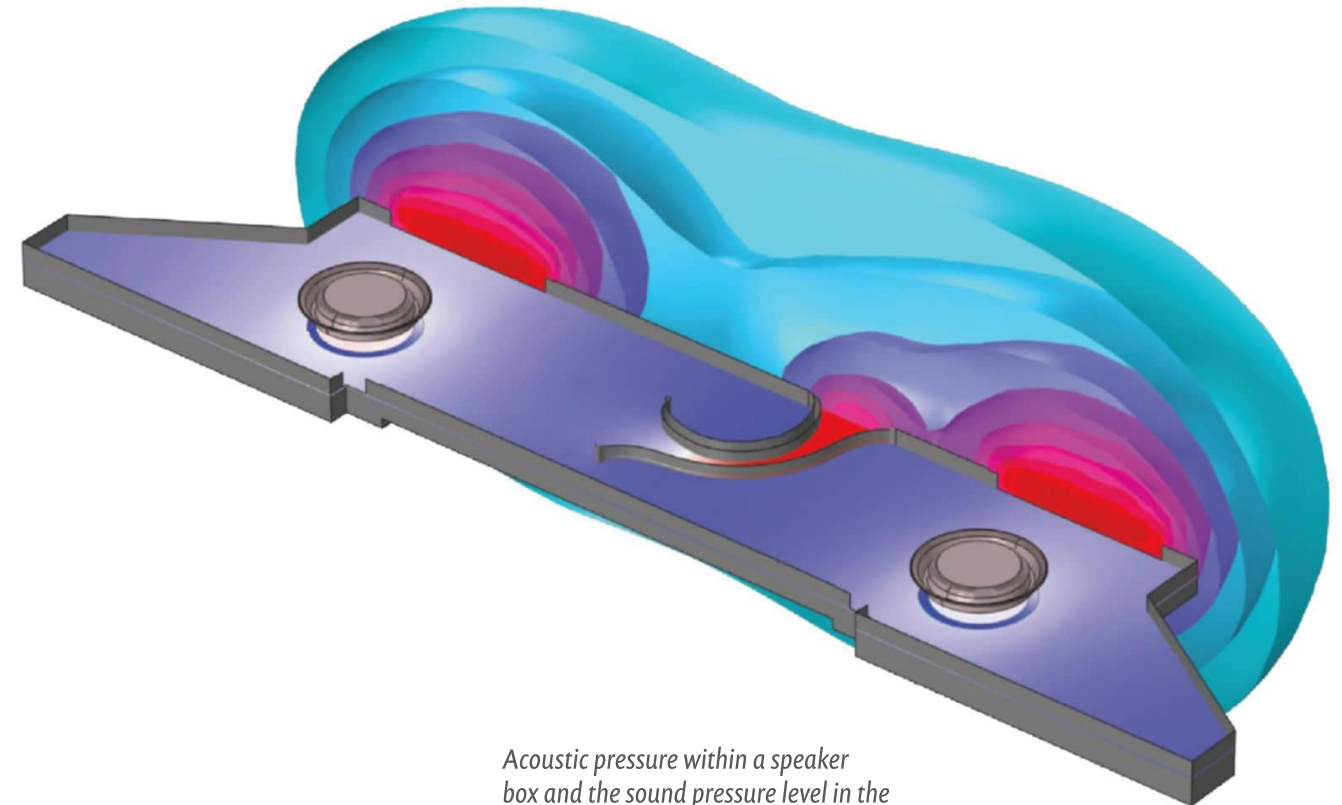
Smart Amplifiers are self-contained audio systems that combine a power-boosting amplifier with integrated DSP that runs cutting-edge speaker-

protection and signal-boosting algorithms. This is critical to boosting the audio quality and efficiency in the smallest drivers and also the largest bass drivers. In theory, all this is evolutionary. In practice, the results can be revolutionary, particularly at the extremes—small-size transducers and the largest loudspeakers.

As speaker designers know all too well, the main limitation is not about the research and the ability for innovation, but in the actual implementation of a complete product design. Sometimes a project is scheduled to go to market with price restrictions. Sometimes, there simply isn't time enough to experiment with the practical benefits of something new, or the added risk simply doesn't match the perceived benefits.

There's also a lot of innovation put into a single product, where the inventor or manufacturer restricts access with patents and a wall of protections to the work. As a result, products that are highly praised and recognized at a certain point, are completely forgotten over time. We've seen that happen countless times, and unfortunately loudspeaker history contains a large graveyard of great but abandoned products and ideas.

*Simulation + testing =
optimized loudspeaker designs*

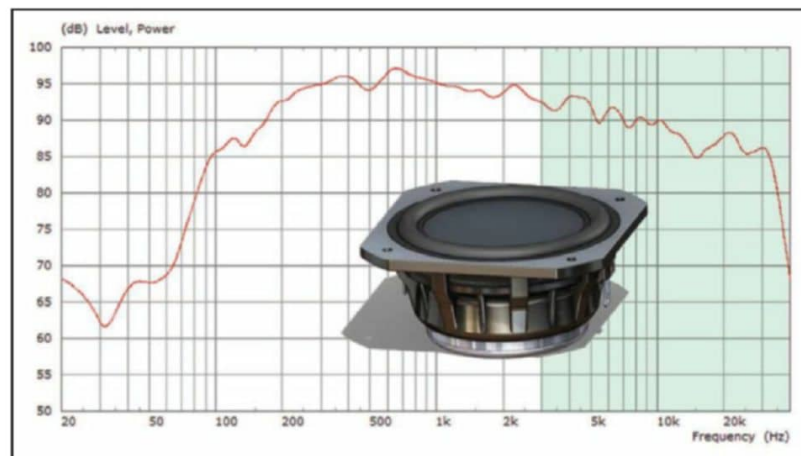


Acoustic pressure within a speaker box and the sound pressure level in the surrounding domain.

A global leader in electronics rose to the top of the audio industry by adding multiphysics simulation to their design workflow. COMSOL Multiphysics® enables audio engineers to couple acoustics analyses and other physical phenomena to address design challenges inherent to loudspeaker and soundbar designs.

The COMSOL Multiphysics® software is used for simulating designs, devices, and processes in all fields of engineering, manufacturing, and scientific research. See how you can apply it to your loudspeaker designs.

comsol.blog/loudspeaker-design



Frequency response of the new Tectonic BMR-TEBM54C20-8 driver, which is even more compact and able to generate an extended response.

New Flat Radiators

Tectonic Audio Labs is heir to a long tradition of speaker innovation, technologies and patents. Tectonic Audio Labs manages all associated NXT-patented bending-wave OEM audio assets and now focuses mainly on its latest Distributed Mode Loudspeaker (DML) designs and Balanced Mode Radiator (BMR) products—the company's next-generation effort to replace moving coil speakers and traditional cone diaphragms.

BMRs combine bending wave mode technology with pistonic motion to create full audio range, wide directivity speaker drivers that are now ideal for many consumer designs—from portable Bluetooth speakers and soundbars to automotive applications—and new unusual product categories, such as smart connected devices (e.g., smoke alarms) or even virtual reality headsets. BMRs are available in square and

round formats and include models with an extended frequency response and wide directivity compared to a conventional drive unit of similar size, even in very small form factors.

Tectonic recently announced its new TEBM54C20-8 BMR compact model that displays extended frequency response and extremely wide directivity characteristics, even into the ultrasonic frequencies. The TEBM54C20-8 BMR is ideal for two-way systems with a crossover as low as 200Hz, ceiling speakers, soundbars, and the growing category of conferencing systems, fitting also the demanding requirements of voice-based speakers that need to offer full-range premium quality and big sound in a small size.

Speaker designers at the beginning of a new smart speaker project can greatly benefit from engaging early with the experienced Tectonic engineering team, to optimize acoustical, mechanical, and industrial design aspects, while benefiting from manufacturing support.

A more recent company determined to replace traditional moving coil speakers is Resonado, a technology startup founded at the University of Notre Dame. The company developed a reconfigured speaker with a planar voice coil with bar magnets mounted perpendicularly to its flat "race track" diaphragm, which Resonado calls Flat Core Speaker (FCS) technology. The ingenious architecture enables slightly thinner, lighter, and more efficient speakers than conventional equivalents, ideal for soundbars, home theater speakers, and automotive applications.

The first Resonado FCS design was a high aspect ratio configuration, which, when oriented vertically, yields wide line source horizontal coverage with reduced midrange ceiling and wall reflections. For soundbars, the form factor enables a shallower and lower profile resulting in a more elegant design. For automotive applications, implementing FCS



Exploded view of the Resonado Flat Core Speaker (FCS) technology, detailing the parallel bar magnet and planar voice coil array mounted perpendicularly to a flat diaphragm.



M-Force 301P02

Unbridled power harnessed for everyone

We have teamed up with industry professionals to deliver a set of **reference designs** that will make the implementation of the mighty **M-Force** finally accessible to anyone. These designs are available for free to all sound system owners, speaker builders, and install professionals that want to take their sound to new depths.

The new iteration of the M-Force 301P02 features **advanced integration options** and several upgrades that enable anyone to add **infra-bass capabilities** to their system.



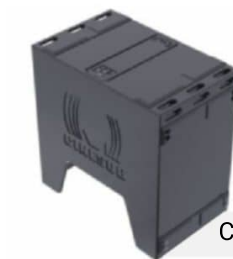
SCAN FOR MORE



EDM SUB



DV SUB



CINE SUB



M SUB

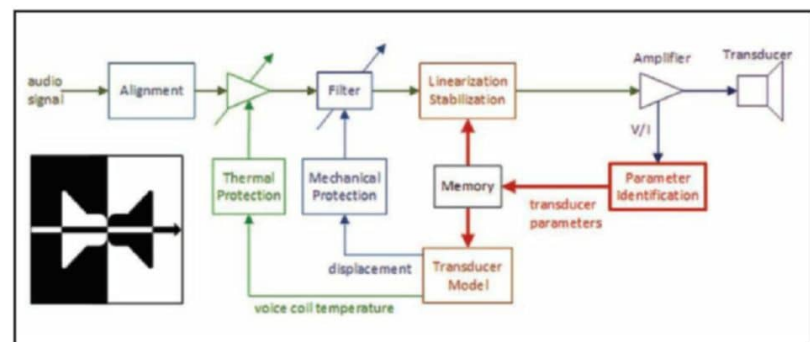


The ESS Crescendo VII is a multicore audio processing SoC with speaker optimization algorithms and features.

technology means a far lighter sound system and far more flexible speaker placement. While FCS technology has a somewhat different internal structure from conventional speakers' workings, the material supply chain remains essentially unchanged. For that reason, Resonado has been able to source all of the components, albeit with new tooling, from all the same vendors used for conventional speakers.

For subwoofers' higher power and lower aspect ratio (more square than rectangular) applications, FCS lends itself to very high BL yet low-cost voice coil magnetic structure enhancement.

Resonado focuses on licensing its speaker technology to brands and manufacturers, and is now ready to bring to market its patented FCS technology. The company has taken a pivotal step toward introducing its speaker technology to the automotive and consumer electronics industries by signing licensing agreements with major manufacturing and design players Soundlab and Zylux.



Klippel Control Sound (KCS) flow chart and Nuvoton NAU83G20 diagram

Integrated Audio Processing

California-based ESS Technology recently launched its seventh generation of audio processors for soundbars and wireless speakers. Better known for its A/D and D/A converters, ESS has been leading in audio processing chips for more than 20 years. The Crescendo VII SoCs are highly integrated solutions for consumer audio products that provide a multicore platform for system control and signal processing with a range of inputs and interfaces. Combining audio processing and signal routing with speaker optimization, the Crescendo VII enables compact, power-efficient, and cost-effective designs for soundbars, wireless speakers, and active subwoofers.

The Crescendo VII is a multicore audio processing chip with two 32-bit RISC and three custom audio DSP processors, and a 64-bit media processor dedicated for audio decoding. The solution is certified for DTS Virtual-X as well as Dolby Digital, MP3, FLAC, and many more. ESS even offers a suite of common audio algorithms to make system design easy including virtual surround, bass enhancement, lip-synchronization, a software parametric equalizer (PEQ) for speaker equalization, as well as custom algorithms that can run on the user-programmable 32-bit Audio Coprocessor.

Designed specifically for audio processing applications in speakers, the Crescendo VII offers the lowest system bill-of-material cost. By utilizing multicores and focusing on power requirements, it offers a reduction in both operating and standby power consumption from the previous generations, enabling systems to meet the demands of green product approvals. The lower operating power also eliminates the need for a heatsink in standard applications, and together with the small 128-pin Quad Flat Pack (QFP) package, enables compact designs for end systems. The new generation adds improved USB and UART speeds that allow for support of improved Bluetooth connectivity, higher bit rate audio, and an increased number of supported channels. The hardware features include stereo ADC for connection to line input sources, an eight-channel I²S input, a four-input SPDIF/ARC receiver for connection to coaxial, optical, HDMI and ARC input sources, a four-channel system ADC for analog status measurement, a three-channel key ADC for multi-key support, and a high-speed USB host controller for firmware upgrade or music playback.

Another exciting integrated solution recently launched with a focus on speaker optimization is the result of a collaboration between Klippel and IC-maker Nuvoton Technology from Taiwan. Together, the two companies developed a new type of smart amplifier with integrated Klippel Controlled Sound

(KCS) adaptive technology that compensates for nonlinear speaker responses. As a result of the project, Nuvoton developed the highly efficient mono Class-D audio amplifier devices, the NAU83G10 and the NAU83G20.

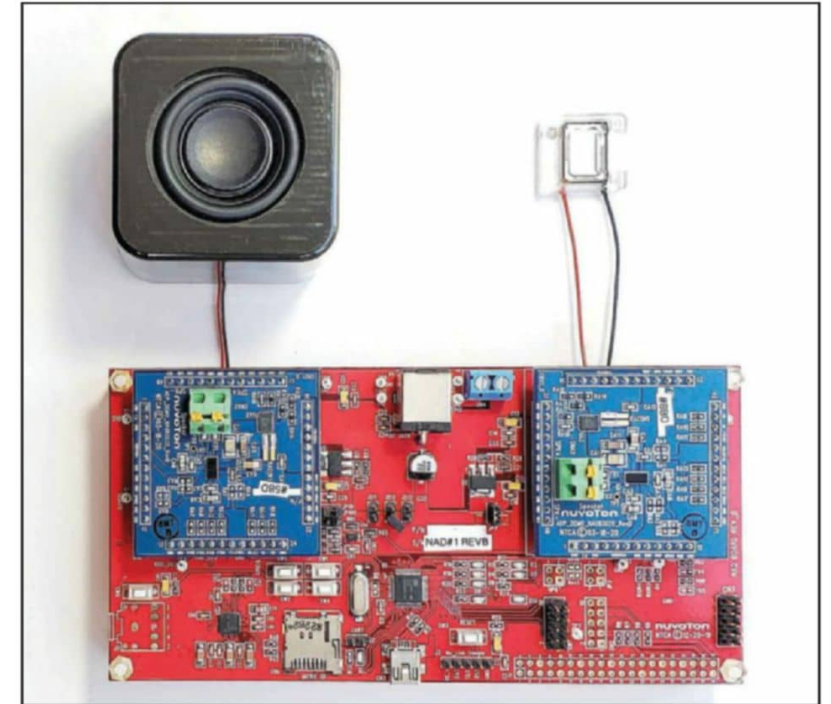
Both the Nuvoton NAU83G10 and the NAU83G20 devices combine a low-latency Tensilica HiFi Audio DSP, high-quality 24-bit audio ADCs for current and voltage sense, and a battery tracking and brownout prevention ALC, all integrated on-chip and optimized for KCS algorithms. This basically enables manufacturers to cope with the undesired effects of signal distortion, heat, and external influences that limit the maximum level and the quality of the reproduced sound.

The solution fully explores the potential of Klippel's adaptive and nonlinear speaker model implemented in the KCS algorithm, to improve speaker performance and sound quality. This nonlinear speaker model describes speaker states (e.g., voice coil excursion and velocity) not only at low levels (small signal domain), but also in the large signal domain where the voice coil excursion is high. This allows the mechanical protection system to work very precisely over the full operating range, which is not possible with linear models. In addition, a nonlinear model can identify and actively shift the voice coil to its optimum position using a small DC voltage. This is achieved by exploiting nonlinear distortion found in the current signal, which is caused by nonlinear transducer parameters.

These features allow the transducer to operate over a wider working range without damage, and improve the sound quality and acoustic echo cancellation (AEC) barge-in performance. Also, the information about the speaker's behavior, including resonant frequency, damping, and other time-variant parameters, allows the use of transducers with more nonlinear motor topologies to focus on maximizing efficiency. KCS even allows to compensate time-variant behaviors and ensuring that the performance stays constant over time. By doing an automatic system alignment to a user-specified target response, KCS automatically calculates the necessary filter to achieve this target.

Other integrated KCS software features include predictive speaker protection against thermal/mechanical overload, automatic speaker equalization to a desired bass response, and online speaker diagnostics to improve product reliability.

The new Nuvoton NAU83G10 and the NAU83G20 smart amplifiers will contribute to smaller speakers that need to safely operate at the physical limits, generate more acoustical output and offer overall higher sound quality. Offering speaker linearization



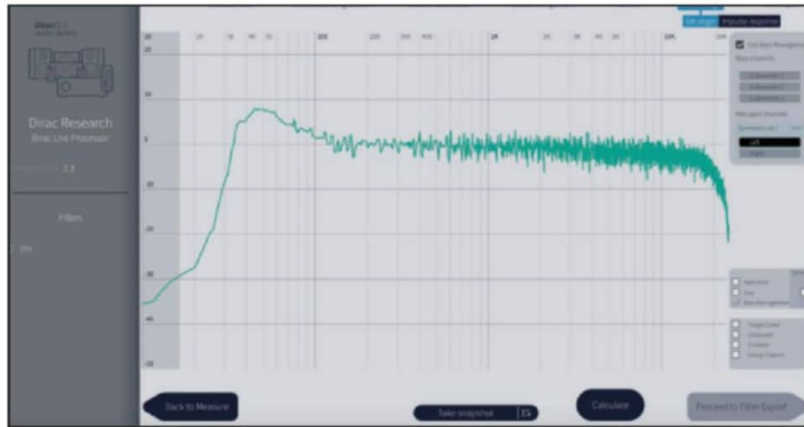
Evaluation system for the Nuvoton NAU83G10 and the NAU83G20 smart Class-D amplifier devices, including all the necessary hardware and software to test KCS data sets and equalizer settings. This enables developers to connect and tune their own designs for a target application. The initial identification of KCS speaker parameters is done almost automatically using a Klippel Analyzer 3 measurement system and only takes a few minutes. The maximum working range of the device under test is identified automatically, providing comprehensive information about the suitability and potential quality of a speaker system.

in real time, a small speaker can be driven to its limits before distortion and it drastically improves ANC, echo cancelers and many systems with mic-speaker feedback.

Speakers Are Getting Even Smarter

There are also speaker optimization solutions that can be directly applied in the end-user environment. Until now, those EQ-correction and room optimization solutions have been largely enabled by software embedded in amplifiers and AV receivers. An industry leader in this area is Dirac, the Swedish audio processing specialist, which continues to license its technologies to an increasing number of integrated platforms and device categories, from mobile to automotive. In the home theater category, Dirac currently offers its Dirac Live room correction solution, which was expanded with Dirac Live Bass Control software to tame specifically the undesired effects of uncontrolled low frequencies in a room.

Available as software for AV receivers, integrated amplifiers, and home theater processors, Dirac Live Bass Control allows for a more immersive home theater experience in any space, increasing overall



Dirac Live Bass Management aggregates measurement and location data from each subwoofer to determine how a system's bass is distributed throughout the room. It then identifies gaps in the sound waves and distributes bass evenly across the room. The solution also corrects the low-frequency sound waves produced by the speaker pair, so the bass produced by each speaker is in sync with the system.

bass and speaker performance by controlling the effects of sound waves reflecting off of surfaces in rooms not designed for optimal home speaker performance. This feature is available in three tiers: Enabled, for manual single subwoofer management; Limited, for automatic single subwoofer management; and Complete, for AI multi-subwoofer management. Dirac Live Bass Control is now featured on systems from Bryston, Focal, StormAudio, Arcam, AudioControl, JBL Synthesis, Bluesound, NAD, Dali, Bluesound, and many others.

This software solution can be licensed by manufacturers and ported to virtually any modern platform that uses machine learning and artificial intelligence to co-optimize the components of a sound

system to deliver accurate, consistent bass throughout a room. Dirac Live Bass Control basically takes the set of measurements that are performed by the user in its location, and sends back the complex calculations needed by the system. Together with the latest Dirac Live updates, this includes phase-correction of speakers and subwoofers across all frequencies.

Dirac is currently adapting its software to support cutting-edge processing platforms that are able to perform these very advanced and complex calculations directly on device, as required by the neural engines that support AI and machine learning implementations, running on "the edge"—without needing to connect to a remote server. This is essential on wearable and mobile devices, which today represents a very large and profitable market for Dirac. But very soon, these same platforms and processing engines will start powering most integrated speaker systems. They already show up in "smart speakers" that are able to perform automatic acoustic compensation every time the user moves the speaker to a different location. Very soon, sophisticated room correction, fully automatic or requested by the user, will be available in most home audio systems.

For more integrated "smarts" that can be directly adapted to speakers, there are multiple vendors with ready-to-use platforms that also offer the added advantage of integrating music streaming service integrations (e.g., Spotify Connect), multi-room wireless protocols (e.g., AirPlay 2), and support for all required wireless connectivity including the latest Wi-Fi and Bluetooth technologies. But speaker designers these days also need to consider voice recognition integration, and that's precisely what companies such as MediaTek, Qualcomm,

Using Libre Wireless' MAVID platform, the Canton Smart Soundbox 3 is a powerful 120W smart speaker from the German brand, and can be paired with a second speaker or as part of a larger multi-room setup.



StreamUnlimited, Linkplay, or Libre Wireless can also offer to the speaker industry.

Libre Wireless Technologies is an extremely innovative company from India that is leading the way with its MAVID low power platform, which is powering new generations of Wi-Fi/Bluetooth connected speakers, enabling a no-compromise far-field voice interface certified for Google or Amazon digital assistants.

The MAVID (Multiprotocol, Audio, Voice, IoT Device) from Libre Wireless is a single-chip mic-to-cloud and edge-computing solution, launched in 2017 and continuously expanded with support for multiple voice implementations. MAVID continues to expand with new SDKs for connected devices, leveraging its responsive Free RTOS operating system, enabling also the support of unique connectivity and libraries for manufacturers and ODMs.

New High-End Drivers

As stated, there's no shortage of new innovative speaker designs for high-end audio applications, from both the most recognized brands and new manufacturers. And that covers all sizes as proven by SEAS with its new Excel Graphene line, initiated with the launch of the W16NX003 mid-bass driver in February 2019, and followed by the new 8" woofer, the W22NX001, which is now the fourth product in this new line from the Norwegian brand.

The new W22NX001 (E0077-08) also represents a successor to the famous W22EX001 (E0022-08) from SEAS, and it's been upgraded like other products with the new graphene cone technology, as well as a new longer voice coil with a titanium former, magnet system with copper rings and caps, and a thicker ferrite magnet, designed and optimized for lower distortion and high power handling.

As with other designs in the Excel Graphene range, the precision-cast, machined, and Graphene-treated magnesium cone acts as a piston through the working frequency band without showing any sign of midrange resonance. High purity Graphene, which is a nano-material with special properties, is added to the surface treatment. The SEAS Excel magnesium cone itself is known for its low moving mass and its high stiffness and damping, and is capable of transmitting even the finest details from mechanical motion into the air, resulting in the cleanest and purest sound. The Graphene-coating technology used by SEAS was developed in cooperation with CIC nanoGUNE (www.nanogune.eu) in Spain.

And speaking of innovative new material formulas for cones, many amateur speaker builders and some of the most prestigious hi-fi brands are currently experimenting with SB Acoustics latest Satori



The SEAS W22NX001 is a state-of-the-art high-fidelity woofer, intended for bass reflex and closed-box designs.

TW29TXN-B-4 tweeter, using a 29mm TeXtreme diaphragm.

This new thin-ply carbon material was introduced by Oxeon of Sweden, an advanced composites fabrication specialist, and has generated so much attention in the audio industry that TeXtreme has now created a separated business unit to support the numerous ongoing speaker projects, and is soon to launch a more advanced, second-generation diaphragm formulation with increased stiffness. In fact, TeXtreme has already generated a new category of ultra-lightweight speakers for multiple applications, with the latest Eminence N314X pro audio compression driver being the first to reach the market with a TeXtreme diaphragm.



The Satori TW29TXN-B-4 is the first home audio tweeter in the market that uses a TeXtreme diaphragm, and will soon be followed with a range of high-end Satori TeXtreme woofers using second-generation Advanced TeXtreme cones.

Likewise, the Satori TW29TXN-B-4 is the first home audio tweeter in the market that uses a TeXtreme diaphragm, and SB Acoustics is also promising a whole new range of high-end Satori TeXtreme woofers, in sizes ranging from the 5" MW13TX-4 model to the 6.5" MW16TX-4 and 7.5" MW19TX-4 models. These woofers combine the second-generation Advanced TeXtreme material for the cones, as well as a soft low damping rubber surround, advanced Bimax spider and a neodymium motor system.

The already-available Satori TW29TXN-4 tweeter compares very favorably with beryllium alternatives at a much more competitive price point. By optimizing the fiber architecture, the symmetric breakup modes are replaced with more but smaller and local breakup modes. This creates a distributed breakup with smaller peaks in the frequency response and ultimately a smoother and more natural sound without harshness.

And SB Acoustics has no shortage of other exciting product launches underway, as a direct result of the intensive work by its Danish engineering design team at Danesian Audio, and significant investments in R&D at Sinar Baja Electric in Indonesia. Exciting products

to look for include the SB Acoustics MD60N-6, a new 2.5" soft dome tweeter using a new coated fabric and powerful neodymium motor system, fitted with a closed-end transmission line chamber, dynamic dome stabilizer and flow resistor.

In its prestigious Satori sub-brand, SB Acoustics introduced the new SB29NRX2-00 10" Norex woofer, using a new exclusive hard paper cone formula for improved piston operation with a low damping rubber surround in a cast-aluminum chassis, creating a versatile solution with easily adjustable moving mass.

Another new Satori product is the AT60NC-4 Kapton ribbon tweeter, using a unique foil damping technique with an optimized rear chamber, which SB Acoustics says generates a smooth controlled dispersion with a response that will fit demanding high-end applications.

And with regard to ribbon drivers, California-based Radian Audio has been teasing the market with an expansion of its LT Series of FEA-optimized planar ribbon drivers, which is now being manufactured with some improvements that make it suitable for hi-fi and high-end audio applications. Proven for its exceptional sound quality in the mid- and high-frequency range, the Radian LT Series drivers have been conquering new manufacturers in the pro audio and cinema markets, where thin-film diaphragms have long been preferred to replace compression drivers. But with its smaller models, the LT2, the LT3 and even the LT6, Radian is now paving the way for demanding applications, which is also appealing because of its reasonable costs and reliability. Radian Audio's LT Series planar ribbon drivers offer a unique combination of wide horizontal and vertical dispersion, low distortion, and the highest output of its class.

The Danish Heritage

And from the expert hands of high-end driver manufacturer Wavecor, there are also more exciting product introductions, as the company continues its routed traditions from the best in Danish loudspeakers. Wavecor drivers are frequently praised for being natural sounding and transparent, and even more praised for being the choice of some of the most demanding and exclusive high-end speaker brands. And that's the reason why most of its products are seldom mentioned, because they are kept within a closed relationship for OEM/ODM agreements. So, when the company's catalog is expanded, not with one new product, but multiple products, it's time to pay attention.

The results of Wavecor's latest development efforts allowed for three new drivers to be added to its catalog: the WF275BD01 (4Ω) and the WF275BD02

(8Ω) 10" mid/woofers in die cast frame; the TW022WA07/09 and the TW022WA08/10 high-end tweeters; and the FR070WA05 70mm aluminum-cone full-range driver with extended bass response. All the drivers offer innovative features and are certain to attract attention from both manufacturers and DIY speaker-building enthusiasts.

First in the new range comes the 10.75" WF275BD01 (4Ω) and WF275BD02 (8Ω) transducers, designed by Wavecor as high-performance bass/midrange units for monitors and high-end hi-fi applications. The 10" mid/woofers are built with die cast frames and are usable even for two-way speakers, offering extended bass performance and overall dynamic and detailed reproduction. As Wavecor details, the design is ideal for multi-way loudspeakers, offering builders the rare opportunity of working with 10" two-way solutions if paired with a suitable tweeter, like for instance some of the Wavecor 30mm units.

The new WF275BD01/02 drivers are paper/fiber-cone units, fitted in a very rigid die cast frame, with the brand's FEA-optimized own Balanced Drive motor structure, for optimal drive force symmetry resulting in largely reduced even order harmonic distortion.

And to reinforce its high-frequency category, Wavecor announced a new series of 22mm chambered neodymium tweeters with increased sensitivity. The new TW022WA07/09 and the TW022WA08/10 are true high-end textile tweeters designed for the most demanding applications, featuring an array of performance-improving details that contribute to obtaining low resonance frequency, low distortion, and very high-frequency extension.

The precision textile dome ensures extended response and consistency, while the 22mm voice coil design offers higher power handling and low resonance frequency, with a copper-clad aluminum voice coil wire that offers lower moving mass, combined with built-in cavities under the dome/edge to equalize pressure. The Wavecor catalog was also expanded with the new FR070WA05 70mm full-range driver (4Ω) with extended bass response. This new beefed up sibling of Wavecor's existing FR070 full-range family received an increased voice coil diameter and increased linear travel and suspension travel. The result, Wavecor explains, is "lower resonance and almost real bass in a very small package."

The new aluminum cone design is recommended for small systems, where frequency range, neutrality, and low distortion are the highest priorities. According to Wavecor, the new compact full-range is also suitable for high definition line source designs, with or without subwoofer assistance. The usable frequency response range is specified from 100Hz to 20kHz.



The Wavecor WF275BD01/02 10" mid/woofers, the TW022WA07/09 and the TW022WA08/10 high-end tweeters, and the FR070WA05 70mm aluminum-cone full-range driver

Scan-Speak Vanguard

One of the world's renowned driver manufacturers, with an impressive number of OEM/ODM clients including well-recognized top-tier brands, Scan-Speak operates as an independent company in the Eastech Group and continues its tradition of Danish development and manufacturing at the company's facility in Videbaek, Denmark.

This year, celebrating its 50th anniversary, Scan-Speak has been extremely active with several product launches expanding its catalog. First, the company announced its latest Revelator 19mm Dome Tweeter, the D2104/712000, featuring the company's patented Symmetric Drive (SD-2) motor, with a neodymium ring magnet system with double vent and coated textile diaphragm.

Deriving directly from the larger sibling D2904/7100, the Revelator 19mm dome tweeter D2104/712000 features a large neodymium ring symmetrical drive magnet system, non-resonant



The new Ellipticor 38WE woofer combines the powerful Symmetric Drive (SD) AirCirc magnet system pioneered by Scan-Speak engineers, achieving high sensitivity, low distortion, and fast response to transients—in common with other Ellipticor drivers.



The Radian LT2 and LT3 planar ribbon drivers are excellent alternatives to other high-frequency solutions in demanding speaker designs.



Lavoce DN14.300T neodymium 1.4" throat compression driver

rear chamber, large roll-surround and sleek design with machined anodized aluminum faceplate. The idea, as Scan-Speak explains, was to offer a smaller tweeter to obtain improved directivity. And yet, the D2104/712000 maintains the relatively high compliance despite the smaller size and frequency response to above 40 kHz.

The large roll surround technology of the Revelator family of tweeters represents a breakthrough in overall performance, with outstanding off-axis response, high output capability, and low-resonance frequency. Additional enhancements have been made to reduce distortion and dynamic compression, such as a careful design to optimize airflow in the chambers.

The other driver is a new 15cm midwoofer, the 15W/8534T00, expanding the established Classic line, originally introduced at CES 2020. This new 5.5" (15cm) midwoofer features the well-known

Symmetric Drive design with copper in the magnet system and low loss linear suspension, also featuring a new Phenomax cone, which Scan-Speak says has shown excellent properties in stiffness and damping.

The new 15W/8534T00 fills a long-lasting gap in the Classic family, where there never was a 5.5" driver. Rated for 60W (100W long-term max power) at 8Ω, with a sensitivity of 85.5dB (2.83V/1m), the design is based on the principles of the praised Scan-Speak 18W/8545 midwoofer, continuing the same tradition with a slim-frame aluminum chassis, symmetrical drive motor system, low loss linear rubber suspension, and updated in the new model with a new cone material called Phenomax. The Phenomax cone is made using five layers of the material with different properties. This multilayer structure incorporates alternating hard and damping thin films (a Kevlar/Carbon-Fiber mix combined with Resin layers, etc.).

More recently, Scan-Speak expanded its Ellipticor driver family, with the 38WE/8582T00, a large-size 15" (38cm) woofer. In combination with the powerful Symmetric Drive (SD) AirCirc magnet system pioneered by Scan-Speak engineers, the 38WE displays high sensitivity, very low distortion, and has an extremely fast response to transients—in common with other Ellipticor drivers.

The new Ellipticor 38WE/8582T00 woofer is built on the same principles as its smaller siblings 18WE and 21WE, sharing the same design principles of an elliptically shaped voice coil and magnet gap. Key features for the new woofer are the effectiveness of the motor structure in this particular range of frequencies, minimizing circular cone break-up modes. This Ellipticor woofer, with its light rigid paper cone, low-loss-linear-suspension, and a very powerful magnet system, allows for a very unusual combination of sensitivity (SPL), low distortion, and dynamic transient response.



B&C Speakers' ME464 1.4" horn was built especially for the DCX family of coaxial drivers, which includes the DCX464 coaxial ring radiator compression driver.

The Pro Audio Side

As already stated, just because the pandemic brought the previously flourishing installation and live sound pro audio segments to a halt doesn't mean that the manufacturers focused on these markets have stopped their development activities. And the results are starting to show, with brands shipping concepts announced earlier in 2020, or completely new products that didn't even have the chance of featuring in a trade show—except maybe in China.

Italian loudspeaker company, Lavoce is one of the fastest-growing driver manufacturers in the pro audio industry, benefiting from a strong R&D base in Rome, Italy, and its own vertically-integrated factory in Jiashan, China.

Following its schedule of new product launches, in this case unveiled at NAMM 2020, Lavoce launched three new "next-generation" neodymium magnet 1" exit compression drivers. The Lavoce DN10.142, the DN10.172M, and the DN10.172K, all offer compact and lightweight solutions for system designers who are looking for low distortion, no-compromise acoustic performance to 20kHz and that critical price-performance ratio, which is so important in

today's competitive pro audio market.

The DN10.142 is a 50W AES neodymium magnet 1" exit compression driver with a 1.4" voice coil, 107dB sensitivity, a high-temperature polymer diaphragm and surround. It also has a useful 60mm diameter, making it perfect for high performance compact two-way systems and line arrays.

The two 1.75" voice coil neodymium designs are the DN10.172M using a vented Polyester diaphragm and surround, rated at 50W AES, and 109dB sensitivity; and the DN10.172K using a vented Polyimide diaphragm and surround, rated at 55W AES, 110dB sensitivity. Both have an 85mm diameter and deliver premium performance in two- or three-way systems or line arrays.

More recently, Lavoce launched its DN14.300T neodymium 1.4" throat compression driver. Following the same "next-generation" design approach and incorporating the company's patented Integral Input Surface phase plug technology, the new DN14.300T has been created to compete at the premium-end of the 1.4 exit compression driver category, offering manufacturers a high flux density neodymium driver with low distortion and a very linear frequency trend to 20kHz.

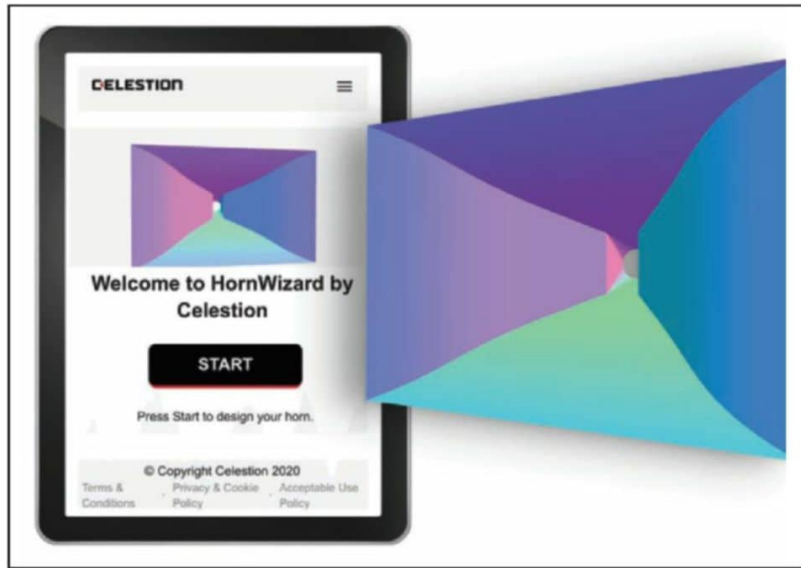
With the Ellipticor family a long tradition of circular motor structures and their inherent breakup behavior has been broken since one of its key features is an elliptical voice coil and magnet gap.

In combination with the powerful SD AirCirc magnet system the Ellipticor's has high sensitivity, very low distortion, and has an extremely fast response to transients.

Above all a TRUE TO LIVE sound, which makes it among the very best speakers on the market!

ELLIPTICOR

SCANSPEAK
www.scan-speak.dk – info@scan-speak.dk



HornWizard is a web app designed with a simplified parameter set to access Celestion's full design software that processes the results.

Featuring a one-piece titanium diaphragm and surround with 3" edge-wound copper-clad aluminum voice coil, the DN14.300T is rated for 110W AES power and 109dB sensitivity. With a conservative 1200Hz recommended crossover point, net weight of 2.1kg, diameter of 130mm and a shallow depth of just 51mm, the DN14.300T is described by Lavoce as "a truly versatile device for a wide range of premium high performance install or touring applications."

For its 1" range of affordable ferrite magnet compression drivers, Lavoce has announced the new DF10-101LS models, a cost-effective screw-on fitting solution for high-performance audio systems using threaded horns. This new 1" compression driver was designed to have enhanced sensitivity across a very linear frequency trend to 18kHz, with an impressive 20W AES power handling supported by a 1" high temperature polymer diaphragm and FEM-optimized ferrite motor structure.

Another Italian stalwart brand in the pro audio market, B&C Speakers has released the ME464 1.4" horn built especially to extract the full potential of its DCX family of coaxial drivers and leverage innovation breakthroughs made possible by wide-bandwidth ring radiator drivers.

As reported previously by *audioXpress*, B&C engineers have been working for more than five years on the development of its next generation of high-frequency devices, including a new family of state-of-the-art, time coherent coaxial ring radiator designs. Given the massive amount of power that needs to be delivered in today's sound reinforcement applications, compression drivers are the linchpin of a PA system. And yet, operating at wavelengths too small to readily couple with other drivers, these compact drivers alone have to fight distance and

atmospheric losses to deliver concert sound pressure levels to ever larger audiences.

The largest horn so far introduced by the Italian company and commercially available, the ME464 horn offers 70° x 55° nominal coverage with constant directivity and excellent loading down to 300Hz. It was designed as an immediately available solution for those demanding professional audio applications. It features a 22" (575mm) wide mouth loading an attached driver to 300Hz, while the 1.4" throat stabilizes the polar pattern to the upper limit of human hearing.

Made of durable and non-resonant polyurethane, the ME464 can stand up to road use as well as shine in high output arrays in fixed installation. As B&C Speakers details, the reduced inter-box interaction and off-axis energy from the tight pattern brings immediate benefits to sound quality and clarity in in high-power configurations. Plus the ME464 is the only horn that can support the full frequency response of the brand's DCX464 coaxial ring radiator compression driver, which delivers lower distortion at higher SPLs than has ever before been possible.

This development follows the introduction of the B&C DCM414 midrange compression driver, building on the same DNA of the DCX464 HF coaxial design. The result of new materials and thousands of hours of modeling and testing, the DCX464's midrange diaphragm covers 300Hz to 5.5kHz with 110dB sensitivity, and its 100mm voice coil handles 220W. The 64mm coil high frequency diaphragm covers 3kHz to 18kHz with 112.5dB sensitivity and handles 160W. The patent-pending midrange integrator allows both diaphragms to work in harmony over a wide bandwidth, for greater combined output and crossover flexibility. All this energy arrives at a 1.4" throat, from the most compact package that can be designed today.

And now B&C is also offering a new option to manufacturers working with dedicated midrange drivers. B&C's new DCM414 features a high-temperature polymer ring radiator diaphragm that covers 300Hz to 6kHz with 113.7dB sensitivity. Its 100mm voice coil handles 220W thanks to an aluminum body optimized to remove heat. All this energy emerges from a 1.4" exit (or 2" exit in the DCM420 model) in the most compact form factor available.

And since we mentioned horns, many speaker designers and even enthusiasts will be happy to know that Celestion just introduced its HornWizard app for high-frequency horn design. Celestion has a long history in this domain, and the Celestion engineering team programmed its own tools for horn simulation, essentially to support its customers. Now, HornWizard is a new desktop and mobile application aiming to

introduce systems engineers to Celestion's in-house horn design capabilities. The new app enables the user to test a basic concept design, using just a few parameters, to determine whether a particular horn specification could work in practice. In the app, users can reliably verify designs, quickly adjust them with different parameters and then view the results in a matter of seconds.

HornWizard has been created as a Web Progressive App (WPA), usable on both desktop and mobile devices to access Celestion's full design software using a simplified parameter set. The app doesn't contain any

horn design software itself: It simply acts as a portal. The user can input parameters that are processed by the in-house software with results returned to the app interface.

HornWizard is not a complete horn simulator, but uses a simplified method to produce a close estimate of directivity and frequency response. First and foremost, it's a "reality check" to see if particular specifications can be achieved within specified constraints. It also enables the user to see how different drivers work on the horn, aiding in the driver selection process.

Bold North Audio

New High-Quality Speakers Designed and Assembled In America

American audio solutions manufacturer, MISCO, has expanded its line of premium and performance-based drivers, designed and manufactured under the Bold North Audio brand. These new models add to a growing line of premium and performance-based drivers, all designed by transducer engineers with decades of experience and manufactured in MISCO's own facility in St. Paul, MN, avoiding issues with tariffs and ongoing trade disputes.

The Bold North Audio BDT-2901 (P/N 78036) tweeter was designed to meet requests for a precision 29mm dome tweeter with a low crossover point, ideal for use in a two-way system paired with a 6.5" or 8" woofer. It can be easily applied in products targeting high-end audio, home theater and monitor applications, offering both excellent performance and repeatable quality.

As MISCO details, the BDT-2901 is currently the only soft dome tweeter produced in the US, and is fully Klippel verified using state-of-the-art Klippel testing equipment to optimize the design. This ultra-low distortion tweeter also combines a unique inductance modulated neodymium magnetic circuit with a high-quality silk dome manufactured by Dr. Kurt Müller in Germany, along with a large back-chamber to allow for a crossover below 1kHz.

The BDT-2901 dome tweeter combines this large rear chamber and a damped center core with inductance reducing copper caps to provide one of the broadest frequency ranges available in a soft dome design: 1000Hz to 25,000Hz. The unique T-Pole design is coupled with a tapered front plate to concentrate gap flux and increase BL.

Two High Excursion 6.5" Woofers

Following the introduction of the high performance BDT-2901 soft dome tweeter, MISCO unveiled more products to its Bold North Audio line, creating a cohesive line of complementary drivers. Two new Bold North Audio 6.5" high excursion woofers, BWX-6501 and the BWX-6502, explore the XBL2 motor design, licensed

from inventor Dan Wiggins, where the lightweight, copper-clad aluminum voice coil travels in a double magnetic gap, centered within inner and outer inductance modulating elements with copper shorting rings in the motor to reduce distortion. By using a high energy neodymium ring magnet, a high BL is maintained while the magnet structure size and weight is minimized.

The BWX-6501 (P/N 82109) and the BWX-6502 (P/N 82141) both allow for a wide range of crossover points, where the BWX-6502 model responds from 40Hz to 1kHz, and the BWX-6501 offers a 40Hz to 4kHz frequency response for mid-bass applications. The BWX-6501 woofer employs a highly damped abaca fiber cone with a rubber edge, as well as an aluminum phase plug to further reduce acoustic-wave energy interference—a design compatible with the new Bold North Audio 29mm dome tweeter. The BWX-6502 woofer uses the same cone material with a highly damped, oversized rubber edge, and a right-sized dust cap ideal for very compact subwoofer enclosures.

The Bold North Audio range also includes a new 4" (102mm) carbon fiber bass/midrange woofer, the BMW-4001, with a wide frequency range, between 60Hz to 12kHz, appropriate for a



The new Bold North Audio BDT-2901 is currently the only soft dome tweeter produced in the US.



MISCO has taken the XBL2 technology and designed its new Bold North Audio 82141 neodymium magnet 6.5" woofer with a Klippel-verified, minimum linear Xmax of 9.5 mm!

wide range of high-end audio systems—whether it's a two-way bookshelf speaker, a multi-driver floor standing speaker or a professional line array. The rigid, but lightweight carbon fiber cone along with an edge-wound copper voice coil (1") on a Kapton former, combine for excellent transient response and an extended and controlled top end roll-off. The frame is heavy duty, aluminum die cast with venting below the flat spider and through the center of the magnet structure to avoid air turbulence and noise.

Finally, the range was extended with the BWF-8001, an 8" stamped frame woofer/subwoofer to add low-end to any speaker design, including high-end home theaters. With an EBP of 94, this woofer is ideal for ported systems and is driven by a heavy-duty dual ferrite magnet and an overhung four-layer coil. The cone is a Klippel-optimized Abaca fiber cone with rubber surround. The stamped steel basket has additional depth below the spider for extra mechanical travel.

www.miscospeakers.com

Acústica Beyma

Raising Force Factor with New QLEX Woofer Series

While many OEM pro audio driver manufacturers have temporarily retreated and quarantined during the coronavirus pandemic, others leveraged the additional available time to refine designs. Beyma continued its search to break the limits of acoustic technology and in May 2020 unveiled the latest QLEX woofer series, developed to offer the extra energy and performance that the latest generation of sound reinforcement systems demand.

Joining the concept and design of Beyma's successful 18LEX1600Nd, the company introduced three new high-performance subwoofers in 2019: the 12LEX1300Nd, the 18LEX1000Nd, and the 15LEX1000Nd. The new woofers feature long excursion capabilities, very high output and very low distortion and have been designed for performing better in vented cabinets and loaded horn designs. All of the models are top competitive weight/performance units thanks to an optimized FEA neodymium motor structure, which features Beyma's exclusive Maltcross technology. As with the other LEX drivers, the new models feature waterproof cones with treatment for both sides, double silicone spiders, and triple surround rings.

Right on the heels of these innovative LEX woofers and subwoofers, the Spanish professional loudspeaker manufacturer announced the new QLEX woofer series of low-frequency drivers with very impressive specs and longer excursion capabilities, featuring three unique ceramic drivers, available in 21", 18", and 15" sizes, all powered by Beyma's exclusive four-layer voice coil technology "QUATTRO". The series FEA-optimized motor structure raises the drivers' Force Factor (BL) almost 30%, which means more energy in the gap and improved control from the amplifier.

This BL increase also makes any of the drivers in the QLEX

series a perfect ally for acoustic systems that require higher SPL speaker levels, excellent linearity, and ultra-low distortion. The Beyma QLEX 21", 18", and 15" woofers, all feature unique ferrite drivers that are able to compete with similar-sized neodymium drivers on the market. And once again the heat dissipation of the entire QLEX family is improved thanks to Beyma's Maltcross technology, allowing higher power handling, with a superior ratio of total SPL vs. weight, which make them lighter than similar neodymium speakers for a given coil diameter.

The larger 21QLEX1600Fe woofer, optimized for direct radiation and band-pass subwoofer applications, is rated for 3200W program power and 1600W AES power at 8Ω with 98dB (1W at 1m) sensitivity. This large size subwoofer offers extended controlled displacement



The new Beyma QLEX Series features unique ferrite drivers that are able to compete with similar-sized neodymium drivers on the market.



Figure 1: The Maltcross technology is a forced convection circuit at the core of the driver, helping to lower the working temperature of the coil and also acting as a demodulation ring, linearizing the inductance of the transducer.

with ±13mm Xmax. It uses a waterproof cone with treatment on both sides and a double silicone spider.

The intermediate 18QLEX1600Fe woofer model, shows identical specifications (3200W program power and 1600W AES power at 8Ω with 97dB (1W at 1m) sensitivity. Construction details add an exclusive Neck Coupling Reinforcement (NCR) membrane. The high power handling and low distortion 15" subwoofer, the 15QLEX1600Fe, makes the QLEX series design even more impressive, showing similar specifications and similar sensitivity, including Xmax displacement. According to Beyma, all these drivers are better performing in vented cabinet, band pass, and horn-loaded designs.

Throughout 2020, Beyma also introduced the new MCB Series and Lex Ferrite series. The MCB series are intended for low- and mid-bass applications and are labelled as the most competitive high SPL 3" voice coil woofers (10", 12", and 15") in the market. The LEX Ferrite features five speakers from 12" to 18," in 1600W and 1000W AES and are the ferrite brothers of the original LEX neodymium subwoofers launched in the past 3 years. All feature long excursion, linearity, and ultra-low distortion in very light formats.

Maltcross Technology

All these products feature Beyma's patent-pending Maltcross technology. Maltcross was developed during the last great neodymium price crisis (2010) to enable a ferrite alternative to neodymium-magnet speakers. Basically, Maltcross enabled Beyma to offer much lighter ferrite speakers and the company has since then kept the momentum going, as it is able to offer smaller and more compact coil loudspeakers and mechanical assemblies than its competitors. Going against trends in the transducer industry to go bigger, thanks to Maltcross technology, Beyma achieves the same performance of larger voice coil loudspeakers with smaller coils.

Maltcross is based on Beyma's advanced cooling system (patent pending), which dramatically lowers the temperature of the voice coil, extracting heat, and consequently increases power handling and reduces the power compression losses (Figure 1). The concept of the design lies in creating an air flow when the speaker is working, reducing the temperature through forced convection. This cools down the voice coil's entire surface from the instant at which the transducer comes into operation, preventing excessive temperature increments.

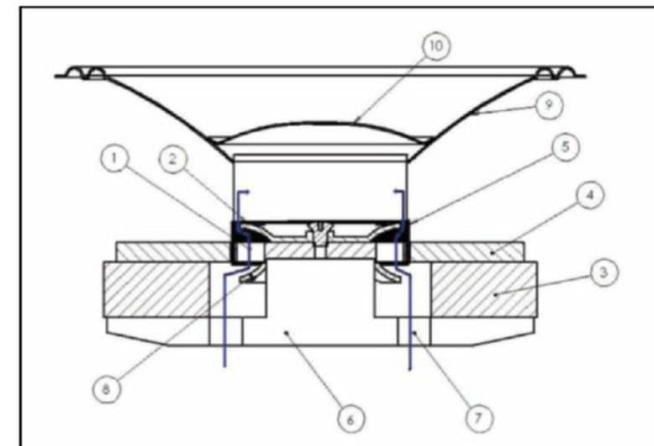


Figure 2: Section view of the Maltcross concept, showing metal parts with specific air exits (1 and 7) and special diffusers (2 and 8), placed in order to direct the air flow to the voice coil.

Figure 2 shows it is the self-oscillatory movement of the coil (5) which, transmitted to the dust cup (10) of the membrane (9), acts as pump impeller (in both directions) of forced air cooling. When moving in the upward direction, that forced air flow enters through the holes (7) located at the bottom of the back plate (6) and circulates through the holes (1) located in the top plate in the direction to the upper diffuser (2), so that, the air will be blocked and forced to move towards its end.

It is precisely this lateral air displacement that is responsible for cooling the upper part of the coil (5), which protrudes from the air gap. In the same manner, when the displacement of the moving assembly takes place in the downward direction, air passes again through the holes (1) located on the top plate, but in the opposite direction toward the lower diffuser (8), which causes again air displacement toward the lower area of the coil (5), thus allowing it to cool. Finally, the hot air is expelled to the outside through holes (7) of the back plate (6). This process achieves a more efficient cooling since the air flow is in direct contact with the entire surface of the coil outside of the air gap, not just a part of it, keeping the coil in a constant and relatively low temperature.

As previously explained, one of the main benefits of Maltcross is the increase in power handling capacity for a given coil size. As a useful reference, a speaker with a 2.5" voice coil, rated between 250W and 350W AES, will be able to handle more than 700W AES without burning the coil with this system. Similarly, Beyma is able to offer a 3" voice coil woofer that performs more efficiently than a similar rated 4" voice coil model, with all the advantages in reduced weight and lower cost.

Another important benefit of the Maltcross technology is reducing power compression, which occurs in real applications, once the system is working and the speaker reaches thermal stability. This well-known effect occurs when temperature increases, increasing also the resistance of the coil and consequently reducing dissipated power. This means less SPL after some minutes of work. By considerably lowering the coil temperature, Beyma's Maltcross consequently reduces the losses due to power compression in around 2dB, which is more than significant in real-world applications.

www.beyma.com

SB Audience

A New Pro Sound Brand Enters the Market

As reported by *audioXpress* in 2019, Sinar Baja Electric announced a new strategy and new product segment organization for its various brands. SB Acoustics will continue to be its hi-fi driver product range, including Satori for the high-end segment, while the new brand SB Audience will represent its pro sound product line available for the international market, and SB Automotive will identify its new car audio product line worldwide.

The company has more than 35 years' experience in pro audio loudspeakers, since SB has been producing a rather extensive line of pro sound OEM products including 6.5" to 18" PA woofers, midranges, and compression drivers, restricted to Indonesia and the Asia-Pacific region. With SB Audience, these products will now reach a global audience, expanding distribution to all continents.

SB Audience drivers will be labelled with three product lines that include Rosso, for high performance at an attractive price; Bianco for "good performance at the most reasonable cost"; and Nero, the "highest performance" line, still achieving exceptional price performance ratio.

One of the very first SB Audience products to reach the market was the Bianco 44CD-K compression driver. Being a low-cost design, the market could quickly understand how the brand would impact the market, since this 1" throat exit compression driver is already an impressive design with excellent performance. The Bianco 44CD-K features a 1.75" diameter voice coil with Kapton former, driving a single piece Kapton diaphragm and surround, plus a FEA-optimized ferrite magnet motor structure, allowing for a continuous power handling of 100W with 101dB sensitivity.



The SB Audience Rosso ROSSO-65CDN-T and the 65CD-T titanium diaphragm compression drivers with a 1.4" throat exit are available with neodymium and ferrite magnets.

Next, SB Audience introduced its Nero 6MRN150D midrange woofer, another impressive model, showing the company's ability to deliver an extremely well-specified driver, starting with a proprietary cast-aluminum frame, paper/graphite composite cone with a flat foam surround, FEA-designed neodymium ring magnet, and a 1.95" voice coil with round CCAW wound in a glass fiber former.

More recently, SB Audience expanded its range of compression drivers with two new Rosso high-performance designs, both showing a distinctive looking cast-aluminum rear chamber that visually defines the new products and was optimized to minimize resonances and absorb reflections.

The SB Audience ROSSO-65CD-T compression driver combines an advanced ferrite motor assembly with a titanium diaphragm, voice coil with flat copper clad aluminum wire and a copper cap for reduced distortion and extended high-frequency response. It offers high sensitivity and linear frequency response, slightly accentuating the upper frequency response.

The SB Audience ROSSO-65CDN-T compression driver is a neodymium motor assembly version, also using a titanium diaphragm and shares the remaining construction and designs features with the ferrite version, offering a smooth yet detailed and dynamic character with high sensitivity and linear frequency response.

Worldwide distribution and local stock for SB Audience products is being established in the United States, Europe, and Asia.

www.sbaudience.com



The SB Audience Nero 6MRN150D midrange woofer uses a proprietary cone paper material with graphite.



The Jensen N12K speaker from Fender's Tone Master Series amps is now available as a standalone purchase.

CE Distribution

New Jensen N12K Speaker

The Jensen N12K, a vintage neodymium 100W 12" speaker, was previously only available in Fender's own Tone Master Deluxe Reverb and Twin Reverb amps and has now been made available for purchase as a standalone driver. The Jensen N12K delivers warm, full-bodied tone with remarkable headroom and just the right amount of brilliance and sparkle. It produces true original American tone in a modern and lightweight 4.41 lb. speaker.

Since the late 1990s, Jensen speakers have been manufactured by SICA Altoparlanti, an Italian speaker manufacturer, under a licensing agreement. Jensen worked in conjunction with Fender to develop the N12K. The process involved extremely accurate analysis and optimization of the overall sound, starting from the same seamed-cone membrane used in the C12K. The frequency response curve of the two speakers is strikingly close, as well as most of the key elements and parameters.

Pre-registered wholesale businesses can order the Jensen N12K directly from CE Distribution, a global supplier, shipping around the world, based out of Tempe, AZ, and serving the new and vintage musical equipment parts requirements for the OEM and wholesale markets.

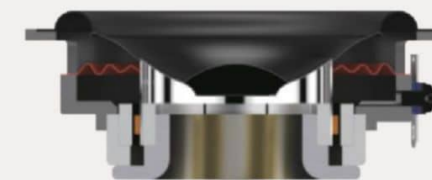
CE has been supplying music stores, repair shops and OEMs for more than 20 years, offering the most extensive variety of products for guitar and guitar amps available. Retail sales are handled by Amplified Parts at amplifiedparts.com.

www.jensentone.com | www.cedist.com



Radial Magnet Transducer Series

From 2.25 to 3.5 inch. This series using our patented dual suspension design with a dual-ring radial structure on a magnetic motor system for better BL(x) and excellent Kms(x) symmetry performance.



The key feature on this design delivers more even magnet field lines around the voice coil, contributed to low magnetic field leakages. Another key attribute on this design is when during high excursions, the BL changes are very low and contributed to very low distortion comparing to conventional magnetic motor designs.

This slim design is perfect for soundbar and compact portable applications, targeting on premium audio qualities.

Proudly Produced by **EASTECH**

marketing@punktkilde.com

www.punktkilde.com

Electrostatic Pure Silicon Microspeakers for Wireless Earbuds

This article discusses how innovations in microspeaker technology, and specifically a new MEMS topology from Arioso Systems, will contribute to next-generation true wireless earbuds.

By **Hermann Schenk**, CEO, Arioso Systems
and **Lutz Ehrig**, Head of Technical Marketing
& New Business Development, Arioso Systems

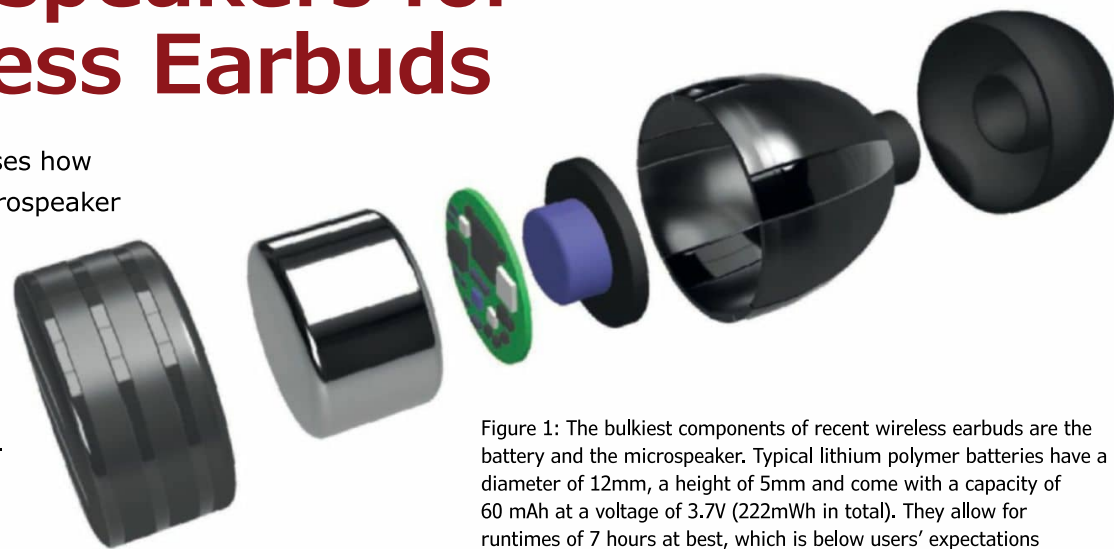


Figure 1: The bulkiest components of recent wireless earbuds are the battery and the microspeaker. Typical lithium polymer batteries have a diameter of 12mm, a height of 5mm and come with a capacity of 60 mAh at a voltage of 3.7V (222mWh in total). They allow for runtimes of 7 hours at best, which is below users' expectations of 8 hours per day minimum (2020 ams survey). Typical earbud transducers have diaphragm diameters of up to 12mm. With a height of approximately 3mm they account for approximately 340mm³ space.

For most people, wearing glasses or contact lenses all day long is a normal thing to do. These aids are lightweight, they do not need to be recharged, and they do not give the user the feel of being cut-off from the surroundings.

For earbuds this is certainly different. They are often more bulky, heavier, and less comfortable to wear, so they are not conducive to being worn all day. Even worse, their current technology quickly drains the battery. This limits how much smart functionality, Internet connectivity, and privacy earbuds can offer. A glance at a typical teardown (**Figure 1**) reveals that it is mostly the battery and the speaker consuming the space inside these designs.

In any case, the strive for better earbud technologies has recently gained substantial momentum. Improved batteries are on the agenda. Luckily, battery technology advances are currently driven by the megatrends of electromobility and green energy storage. Batteries for mobile communication will continue to become smaller, accordingly. However, there are tight limits for the battery capacity when it comes to earbuds: Mounted inside the ear canal, the total energy stored in a suitable battery should not be large

enough to represent a safety hazard. For this reason, more power efficient wireless solutions and less power consuming speech recognition platforms are key items on the R&D roadmap. In fact, all next-generation earbud components must improve: They must shrink in size, become much more power efficient, and comply with tough cost targets. This clearly calls for innovations on microspeaker technology.

Pure Silicon Microelectromechanical Systems (MEMS) Transducers

Silicon technology is the modern world's epitome of miniaturization and price competitiveness. Silicon derives its commercial success from replacing complex mechanical assembly processes by lithography—a process conceptually as simple as taking a photographic picture. Thousands of devices are made on a single silicon wafer, a kind of silverish disc, in one go. An unprecedented product quality, manufacturing productivity, and decline in cost per unit results. Is this also an option for microspeakers?

The mantra here is MEMS devices, transducers made purely from silicon. MEMS devices are ubiquitous in modern life, they tell our smartphones to rotate the picture from portrait to landscape, if

needed, and trigger the airbag in our cars. Last but not least, they have become today's dominant microphone technology. The idea of a MEMS microspeaker has attracted many talents around the globe for over a decade now. But making this dream come true is not easy.

Let us look for example, at the size of a MEMS microspeaker chip. A headset designer probably wants a sound pressure level (SPL) of say 120dB—to have enough systems budget for active noise control. The silicon version of a speaker using a diaphragm needs to have a large enough silicon chip surface to generate this SPL. But the cost of processing one silicon wafer is roughly the same, be it a wafer full of microphone chips or full of speaker chips. Larger chip size means less chips per wafer and thus higher costs per chip. A typical MEMS microphone chip has 1mm². So, a MEMS microspeaker of say 50mm² is likely to be 50 times as expensive as a MEMS microphone! Commercially attractive MEMS microspeakers should, therefore, be much smaller, which is hardly compatible with the use of a silicon membrane.

There potentially are several ingenious ways to cut through the surface area dilemma. A very attractive strategy is depicted in **Figure 2**. The idea is to use the chip volume, rather than the chip surface! An earbud only needs to displace a certain volume of air to generate the desired sound pressure level in the ear canal. No need to be concerned about the physics of sound radiation.

An air displacement as small as 0.5mm³ is enough for 120dB and a typical wafer is 0.75mm thick. That makes the idea of using the wafer volume so attractive. The novel design, therefore, is to virtually cut the speaker diaphragm into small strips and mount them upright into the chip volume for lateral movement. When two of these lamellas move toward each other, they will squeeze the air enclosed between them through air vents, suitably positioned in a top and a bottom silicon layer enclosing the device. That makes a perfect acoustic dipole as required for a tiny earbud!

Lamella Movement

Now what makes the lamellas move? This is a challenge on its own! Employing electrostatic forces certainly provides scope for excellent audio quality without the need for any audio signal preprocessing. High-end headsets are designed this way. However, existing electrostatic headsets require hundreds of volts to drive them. That is not an option for earbuds! Neither would it be acceptable from a product safety point of view, nor would it be compatible with the performance of a small battery.

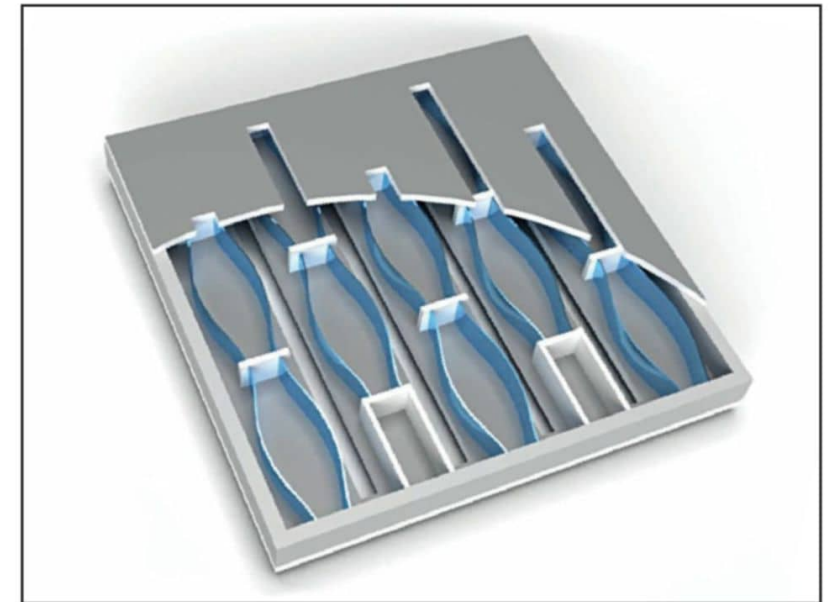


Figure 2: A new pure silicon MEMS design principle uses the volume of the silicon chip rather than its surface for sound generation. This is achieved by arranging hundreds of lamellas for in-plane movement between a top and a bottom silicon layer. Like in an air motion transformer, the air between two adjacent lamellas is squeezed through suitably positioned vents, by the lamella movement. This allows for a small transducer form factor and very competitive production costs, which are dominated by the silicon chip size.

The root cause of the voltage dilemma is the difficulty to design an elastic suspension strong enough to tame electrostatic forces. Think of two electrically charged and elastically suspended plates, used to generate sound pressure, that are facing each other at a certain distance. The electrostatic force between them quickly gains strength with the surfaces approaching each other. This happens so rapidly that after about one-third of the initial distance, the balance of the elastic and the electrostatic forces collapses. An elastic suspension cannot stop this. A large initial electrode distance is usually required for this reason—to generate enough sound output. But a large distance between the charged surfaces means that a large drive voltage of hundreds of volts is inevitable.

A design solving the voltage dilemma is depicted in **Figure 3**. The lamellas shown in Figure 2 are indeed electrically insulated stacks of three electrodes each. The tiny roughly 2.5µm gaps between the electrodes, reduce the drive voltage requirements to only a few volts. Still it remains true that the maximum stroke of the electrode movement is restricted to about one-third of the gap. That is about 800nm only!

The curvy geometric shape of the lamellas depicted in Figure 3 is capable of mechanically converting the minute electrode movement into the required large stroke of the lamellas. That is the secret of this design. Essentially it is a smart mechanical lever mechanism.

The design principles presented so far enable a small form factor and a low voltage drive with a technology that exclusively uses pure silicon technology, requires no specialty materials or manufacturing processes, and is compatible with wafer-level assembly. The remaining topic to be addressed is power consumption.

Power Consumption

Conceptually, electrostatic speakers do not dissipate any energy at all. Any electrical energy required to compress the air in the ear canal, is released from the device back into the drive

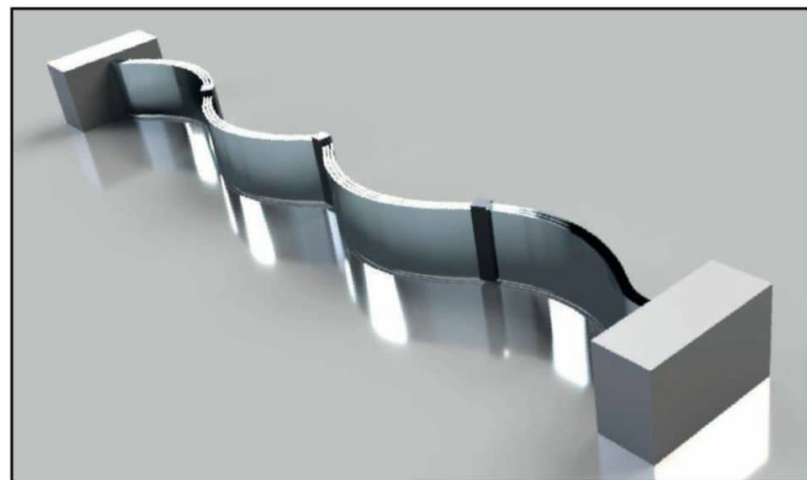


Figure 3: The lamellas are moved by electrostatic forces. They consist of a stack of three differently charged electrodes closely arranged together. The wavy shape of the lamella mechanically translates every minute relative movement of the electrodes within one stack into a sizable stroke of the entire lamella. This is the secret enabling small drive voltages in an electrostatic MEMS microspeaker.

About the Authors

Hermann Schenk holds a PhD in theoretical physics from Bonn University. He was founder and managing director of Covion Organic Semiconductors GmbH, commercializing OLED materials. After the sales of Covion to Merck KGaA, he served as managing director of Freiburger Compound Materials GmbH, a leading supplier of III/V- Semiconductors. Currently he is co-founder and managing director of Arioso Systems GmbH, dedicated to the development, manufacturing and sales of MEMS μ Speaker technology. His recent research contributions are centered around novel modelling techniques for MEMS systems.



Lutz Ehrig studied Mechatronics in Dresden, Germany, and Acoustics in Aalborg, Denmark. In 2009 he joined the Fraunhofer IDMT in Ilmenau, Germany, as a research assistant in electroacoustics. In 2015, he joined a consulting engineering office as a project manager and was working for two years on room acoustics and environmental noise. Afterward, he worked as a research assistant at the Fraunhofer IPMS in Dresden, where he participated in the development of a CMOS compatible MEMS microspeaker. In 2019, he co-founded Arioso Systems and joined the company in 2020. He is responsible for Technical Marketing and Business Development.



electronics once the audio signal commands the speaker to expand the air again: Effectively, the electrostatic microspeaker is all based on reactive power (i.e., power that is not dissipated). Power of course means energy utilization per time. Unfortunately, a major amount, if not a multiple of the reactive power, will be dissipated by the audio amplifier into heat. And the series resistance of the battery worsens this power dissipation, even when the average power take-up is low. This is the challenge.

Basically, all MEMS designs aspiring to microspeaker markets are, from an electrical engineering point of view, capacitors. This means that the reactive power doubles when the frequency doubles, it quadruples when the voltage doubles. There is little that can be done regarding the frequency and the voltage range. The frequency range is defined by the required audio performance, and even if 120mV are enough to generate 80dB SPL, it takes a perfectly linear electrostatic transducer 12V for a 120dB SPL output.

Yet, for a miniaturized MEMS audio system this still is manageable, provided the electrical currents required to load and unload the electrical capacitance of the speaker are small. An attractive earbud audio reproduction system should have a total power dissipation of well below 3mW for a typical use case. Essentially, this puts a cap on the total electrical capacitance of the speaker chip. It should be significantly below 1nF. Note, the basic physics of sound generation (i.e., the adiabatic air compression in the ear canal), in principle allows for capacitances of a factor 30 below.

The additional capacitance is required to provide the energy needed to move the silicon parts inside the MEMS microspeaker. Fortunately, there are various apt MEMS design principles (e.g., reducing the total mass of elastically deformed silicon parts), avoiding fast moving silicon valves and using electrical surface charges instead of electrical space charges.

MEMS Speaker Development at Arioso Systems

Arioso Systems GmbH is developing high-fidelity microspeaker technology designs for mobile audio reproduction, far superior in size and power efficiency, and based on standard silicon-technology with the potential to claim cost-leadership in mass markets.

The idea is to only use CMOS-compatible materials and manufacturing processes, enabling a swift technology adoption by the market. Meeting tight cost targets calls for small microspeaker chip

sizes. This is achieved by Arioso's approach using the chip volume rather than surface mounted silicon diaphragms. An artist's view of an earbud design using Arioso's technology is shown in **Figure 4**.

Arioso's technology aims at a wafer-level microspeaker assembly (**Figure 5**), largely avoiding costly mechanical assembly steps on the level of the individual speaker chips. The targeted electrical capacity of a microspeaker that produces a sound pressure level of 120dB from a 10mm² chip, is below 1nF. This restricts the reactive electrical currents to the level compatible with using small charge pumps without external passive components and enables the use of a novel, proprietary and very power efficient audio amplifier technology for driving. The targeted battery drain of the entire audio reproduction system is well below 3mW, for typical use cases.

Arioso Systems is a spin-off of the Fraunhofer Institute for Photonic Microsystems, Dresden. The company seeks to reach its key technological targets mid-2021, followed by a Series-A investment round. Currently, Arioso Systems is working with selected market participants to perfect its technology and identify market opportunities.



Figure 4: A commercially attractive electrostatic MEMS microspeaker should have a size of not much more than about 10mm² and be able to produce a sound pressure level of 120dB in the ear canal. The battery drain of the entire audio reproduction system, including the amplifier and the charge pumps providing the necessary MEMS drive voltages from a lithium polymer battery, should not exceed 3mW for typical use cases to be competitive.

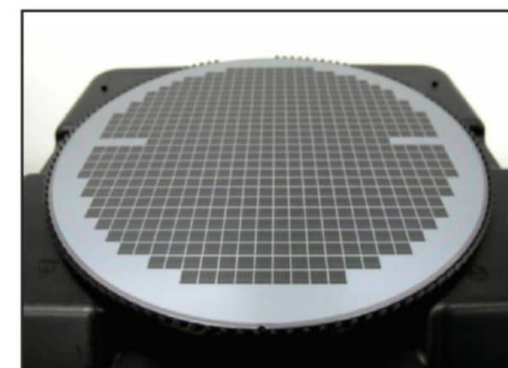


Figure 5: A 8" (200mm) wafer containing 585 microspeaker chips with a chip area of 36mm². Shrinking down the area per chip to 10mm² results in a total number of around 2000 devices per wafer. Wafer-level assembly leads to fully functional devices at the end of the processing that can be tested on the wafer-level as well.

BIANCO 120B150 / 150B350

Open Baffle Applications

- Optimized proprietary cone paper material
- Minimum damping fiber glass voice coil former
- Large voice coil
- Long life silver lead wires
- Designed for open baffle applications

www.sbaudience.com | info@sbaudience.com

ENHANCE THE EXPERIENCE

CLARKSYNTHESIS Professional Transducer TST429 Platinum

www.clarksynthesis.com Patent # 5,473,110

303-797-7500 • info@clarksynthesis.com • www.clarksynthesis.com

Bass Shakers and Tactile Sound



(Image source:
Flexound Systems)

From powerful bass shakers that move floors and chairs in movie theaters to subtle bone-conduction transducers and haptic feedback technologies, there is a world of transducer solutions that continue to improve. This article reviews the history and working principles of those shakers and details some notable examples now available in the market.

By
Mike Klasco
(Menlo Scientific, Ltd.)

At a concert you see the performers, hear the music, and feel the bass. Sound is energy made by vibrations. When an object vibrates, it causes movement in the surrounding air molecules. These molecules bump into the molecules close to them, causing them to vibrate as well. This makes them progressively bump into more nearby air molecules. If this compression and rarefaction of molecules reaches your ear's tympanic membrane, you will perceive these vibrations as sound.

This seems simplistic but you can spend your life studying this stuff and most of us who are reading this do just that. But our perception of music and the auditory world in general includes not only airborne sound energy, but also structure-borne energy, which is perceived by our haptic senses. The ability to perceive and interact with the world through vibration and touch is central to all of our lives, and now increasingly so. In experiencing live sound, our haptic perception contributes to the realism, and conversely haptics can strengthen the illusion and simulation of reality.

Tactile and Haptic Perception

Haptics have the power to intensify everything—from allowing mobile game players to actually feel what they're seeing—to improving the sensory interaction with a wide variety of devices. Haptic perception can be uniquely a bilateral interaction—you press a touchscreen and it shutters back to confirm your action.

On the other extreme—for home theaters, gaming, movie theaters, discotheques, and theme

park rides—this “tactile” component is able to add an extra dimension (sometimes referred to as 4-D) to reproduce the impact of explosions, engine sounds, thunder, and other bone-rattling low-frequency content. This creative impact can be powerful, yet theatrically subtle—as the anticipation created by the vibration effects in the original T-Rex appearance in the *Jurassic Park* movie

Paraphrasing from a whitepaper from Clark Synthesis, Steven Spielberg was trying to give the dinosaur's footfall as waves in a puddle, then hear the first tremor, and then the thundering sound of the big lizard. Because Spielberg was unable to shake the theater floor, he used the next best thing—a visual effect of water rippling. But with bass shakers, the world would shake the way a stomping T-Rex would.

This “feeling” is your haptic perception, and is divided into two different classes: Tactile and Kinesthetic. The things and texture you feel with your fingers, feet, and so forth are because of sensors on the surface tissues embedded in the skin and right underneath it, enabling your brain to feel vibration, pressure, touch, and texture.

The things you feel from the sensors in your muscles, joints, and tendons are classified as Kinesthetic. Weight, stretch, and joint angles of your arm, hand, wrist, and fingers, but also the bass drum thump from your feet to your gut. Our tactile perception of live sound (or even recorded music with powerful subwoofers) actually shows up first, just before the airborne sound component as the

speed of sound through materials is faster than through the air.

The world of tactile and haptic perception has been on the periphery of audio for quite a while. Reproduction of this structural component of live sound can be achieved through the use of a tactile transducer popularly known as a “bass shaker,” physically coupled to various surfaces (e.g., floors, chairs, and a whole load of other stuff).

These transducers pass low-frequency vibrations into them so that they can be felt by the listener. Tactile devices may augment a subwoofer, but they cannot be substituted for a subwoofer.

Tactile transducers are used in home theaters, commercial movie theaters, arcade games for special effects, amusement park rides, aerobic studios, and so forth. Some innovative applications have also been developed for the hearing impaired, bypassing the hearing system in a manner similar to a cochlear implant.

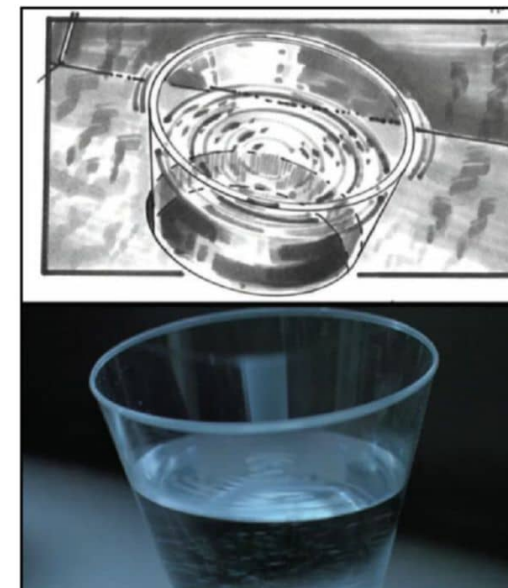
A Bit of History

Bill Phillips of Clark Synthesis, one of the first and a leading tactile device manufacturer, provided some valuable insights to tactile device history.

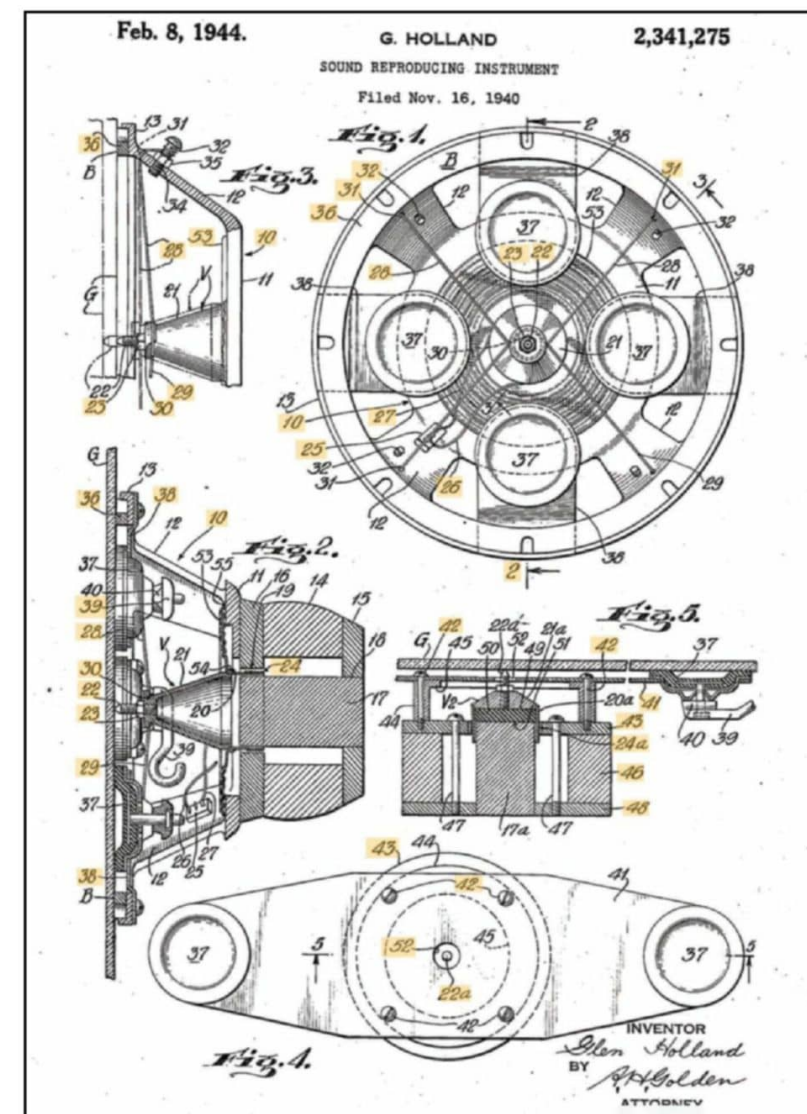
In 1944, Glen Holland received US Patent #2,341,275 for a new “Sound Reproducing Instrument.” This is the earliest patent found for a device designed to transfer audio vibrations into materials (e.g., wood and glass). His design used a stylus attached to a voice coil to transfer the vibrations. I've never actually seen one of the devices, so I don't know if he ever manufactured any or not.

In 1978, a tinkerer named Tom Fenner wanted to see if he could inject sound into a waterbed. He was playing with the little Rolan Star transducer. It worked, but the vibrations were weak. He started experimenting with creating a large tactile sound transducer. Many years later, Fenner created a transducer 14” in diameter and weighing 12lbs. Around this time, Curtis Clark, an oil investor, hired Fenner to install a home theater. Fenner demonstrated his device to Clark, who liked it, but declared it too big. Fenner reworked his product to an 8” device and went back to Clark, who liked what he saw, and in 1993 established Clark Synthesis to manufacture and sell the device, creating a new genre—large-area haptic feedback devices.

I also had some involvement with the tactile side of things. In 1976, Nozomu Matsumoto founded Bodysonic. Matsumoto is best known as the founder of Pioneer Electronics in Japan. Bodysonic was the first “bass shaker,” which Matsumoto developed in his hobby shop. Around 1980, I represented the company for commercial applications and we had



The water shaking on the cup effect on *Jurassic Park*. Script illustration and movie frame (Copyright: Universal)



This drawing from Glen Holland's original US 2341275A patent for a “Sound Reproducing Instrument,” was filed in 1940 and granted in 1944.



ButtKicker Concert and ButtKicker Gamer2 Playseat Edition. The first system is designed specifically for musician monitoring, stage, and studio use, and is a patented low-frequency transducer that features a 3.75lbs (1.48kg) magnetically suspended piston. The second adds instant immersion to any game by mass producing a powerful haptic effect.

design wins at Disney’s Epcot theme park, Chicago’s Science Museum, discotheque dance floors, and other neat projects. Later, I worked with Sammi Sound on its Tremors shakers (also marketed by Philips), followed by Aurasound, whose shakers were also marketed by Alpine and RBH. Some of Aura Systems’ linear actuators were developed for experimental aircraft landing gear, active vibration canceling for helicopters, and igniting hydrogen bombs (talking about serious boom!).

Discussions about some of the early inventions in this field—from sub-harmony synthesizers to Dolby effects channel and the Sensurround experiments—can be found in my first two articles on this topic (see Resources).

High-Definition Tactile

In comparing bass shaker specs and attempting to measure output, we want to offer some caveats. As with most electroacoustic transducers, there is a gain vs. bandwidth trade-off with just about everything. It is easy to design an efficient bass shaker with huge output. Think of a bass shaker as having a Q, so a

high-Q design would have a lot of output over a very narrow band of bass frequencies—sort of a pendulum or a one-note subwoofer. Alternatively, we can design for a wider Q and more linear response over the operating range, but at the expense of efficiency. Products that are reviewed by unsophisticated hobbyists and found to be super-efficient may just be a “one-note” design that wants to vibrate at their favorite but narrow frequency range.

Many shakers are moving magnets, where the casing is fixed and the magnetic structure is compliantly suspended and driven by a stationary voice coil. The voice coil exerts force on the magnetic structure and a reaction force on the bass shaker’s body, with the latter forces transmitted into the mounting surface and similar to the NXT “exciter” momentum drive. The lower-cost shakers typically used ferrite magnetic structures with the electromagnetic control and bandwidth limited—but that is a generalization.

Alternatively, with high magnetic density neodymium, practical and cost-effective direct-drive tactile transducers essentially use a “linear motor” (a conventional speaker topology, but the voice coil drives a piston rather than a speaker cone). Let me say, there are high-performing shakers of both types.

When it comes to seating design for bass shakers, it is important to fool the senses. If the bass shaker only drives a chair, your senses won’t be fooled, particularly if it is apparent that only the chair is vibrating. But if the chair, ottoman, and floor have shakers, then the entire room will appear to vibrate rather than just the chair. There is also a preferred balance of vibration distribution between the back, waist, hips, and buttocks. Also both the overlap and the response of the shaker and the subs must be artfully contoured to reach the most credible effect.

In the current market, there are a number of bass shakers and new transducers for haptic and tactile applications. In this report, we will take a look at some interesting offerings currently available.

Guitammer Cinema and ButtKicker

For more than 20 years, the Guitammer Co. has created compelling entertainment experiences with haptics for gamers, musicians, cinema patrons, and more with its ButtKicker brand. ButtKicker’s low-frequency audio transducers are musically accurate, powerful, virtually indestructible, and patented linear transducers, which recreate amplified audio signals in the feeling range.

The ButtKicker Cinema System (BKCS) has an install base of more than 45,000 seats worldwide offering solutions for new and existing auditoriums

The TST429 Platinum Professional Transducer from Clark Synthesis is the most powerful audio transducer on the market. It outputs full-frequency sound from 5Hz to 17kHz to meet the demands of audio applications, such as high-end home theater furniture, theater risers and platforms, floors (including dance floors), simulators, and gaming systems.



Acoustic Production Test Redefined

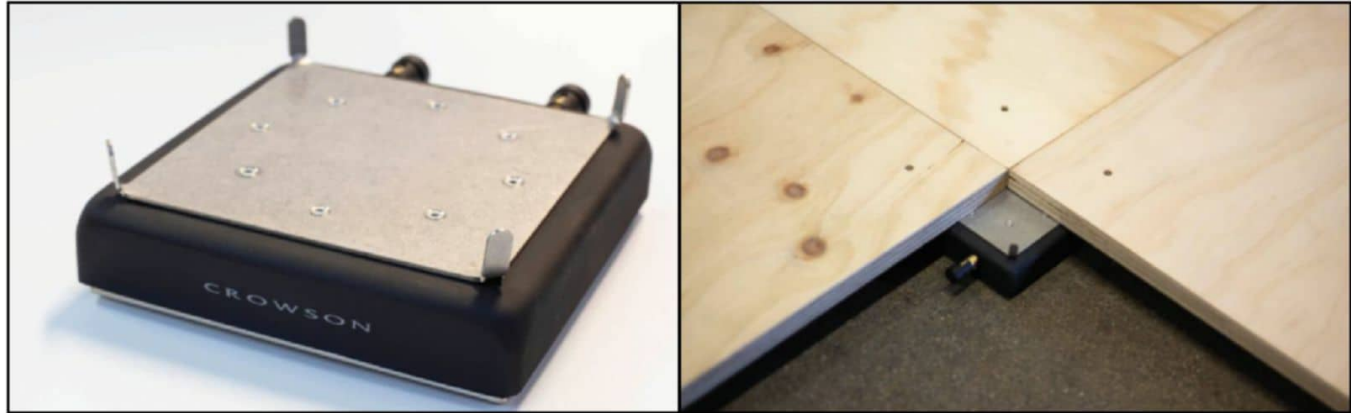
The APx517B Acoustic Analyzer



An integrated system for the production test of speakers, microphones, headphones & headsets



www.ap.com

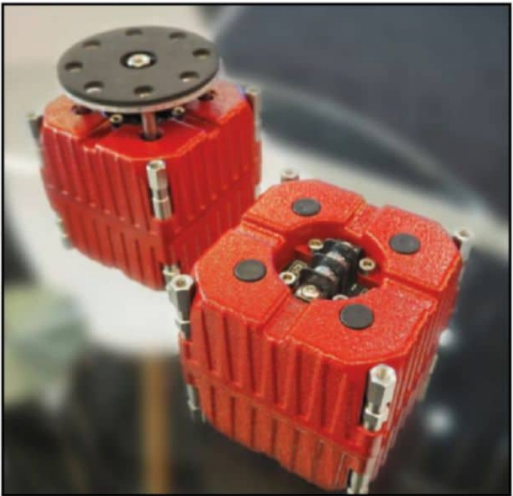


A Crowson VR Haptic Actuator. Crowson Motion Actuators are linear direct-drive tactile transducers that lift and drop the seat allowing for a uniform, infrasonic effect (below 20Hz) with no noticeable resonant modes within the operating range. The solution is used as whole-room floor activation for Virtual Reality (VR) experiences, simulators, and theaters.

with the ability to retrofit existing auditoriums; and all types of seating, including recliner and auditorium-style seating. Most cinemas can upgrade without closing or impacting screenings. BKCS also works with the existing soundtrack from all movies; no encoding required.

ButtKicker gaming products also work with every game, enabling players to feel the low-end vibrations without making the room loud. The ButtKicker Gamer easily attaches to standard center post chairs. The solution enables gamers to turn down or turn off their subwoofer and still feel all the bass. And even while using headphones or a gaming headset, they can still experience powerful bass.

The company's home entertainment products work with all types of furniture, easily installing high-end home theater chairs to comfy recliners. The Individual Seat Control Kits are quick and easy to install adding depth, immersion, and impact to the viewing experience.



Powersoft's Mover is available in a Direct Drive solution (a powerful piston-like motor) and a more traditional Inertial Drive integrated package that can be coupled to any surface.

Clark Synthesis Transducers

Originally developed for military and commercial training applications, these tactile transducers can be used full range or for bass. With a half dozen application-specific transducers, Clark Synthesis has Tactile Transducers variously optimized for swimming pools, drum thrones, flight simulators, home theaters, club dance floors, and even music therapy for the deaf.

Full-range applications include use with hot tubs and wall structures (for invisible speakers) and similar applications. The waterproof version can be used on pool walls. The Tactile Sound Transducers use professional-grade voice coils set into high power ferrite and neodymium magnet assemblies. All components are custom designed and manufactured to exacting tolerances and deliver the high power and the wider frequency range compared to typical large-area haptic feedback devices. This wide frequency range allows the reproduction of fundamental tones and harmonics, making the output pleasantly rich and musical.

Crowson Tech

Crowson Technology has been producing tactile motion actuators for the luxury home theater market for more than 18 years. In that time, its actuator design has evolved but the core technology remains largely the same. Crowson linear-direct-drive actuators go beyond inertial bass shakers by directly lifting and moving the theater seat by as much as one-quarter of an inch. They expand and contract beneath the feet of the seat. This method provides true infrasonic motion/vibration as well as lightning fast transient response and a ruler flat frequency response. Crowson actuators are so accurate they are widely used in research

environments, aerospace vibration simulation, and reliability testing.

Crowson's Shadow-8 Dual Motion Actuator is designed to bring high-fidelity tactile effect to all types of chairs and couches. The compact actuator design is a linear direct drive tactile transducer enabling a uniform effect throughout the chair and True Infrasonic (motion below 20Hz) and as high as 600Hz.

These actuators can be installed by a person with no mechanical skill in less than 15 minutes. No modifications to furniture or flooring are required. The system easily interfaces with standard audio equipment using conventional audio signals.

Crowson is now supplying special virtual reality actuators for wall to wall haptic flooring. An array of 1" tall actuators are placed on the floor and a grid of plywood squares are placed on top. Each plywood intersection is supported by an actuator and creates a zone of haptic feedback. This type of low-profile, easily installed haptic flooring is an ideal way to bring haptics to virtual reality environments.

Powersoft

Powersoft introduced its M-Force linear motor subwoofer technology in 2013 and more recently



morphed this innovative technology into Mover, a powerful but compact shaker. This patented low-frequency direct drive tactile transducer is designed to elevate the entertainment experience by vibrating the surrounding environment, which the human body picks up and, through bone conduction, stimulates the inner ear, translating the vibrations into perceivable frequencies. Adaptable to 4D cinemas, theme parks, or venues with vibrating acoustic floors—as well as applications that go beyond entertainment, Mover is available in two versions: Direct Drive (a piston-like motor) and the traditional Inertial Drive (suspended moving magnet) assembly that can be surface-coupled.

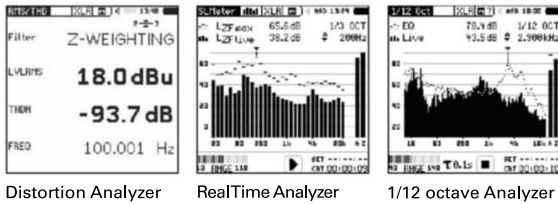
The MEMO bundle combines the Mover ultra-low frequency transducer with a fully optimized Class-D amplifier, specifically designed to enhance home cinema and gaming experiences.

Direct Drive is anchored to an immovable surface

Ready for any Challenge!

Discover an Audio Analyzer you can trust for your application:

- Sound Level Meter class 1
- Level RMS / THD+N
- 1/1 or 1/3 octave RTA (up to 1/12 optional)



americas@nti-audio.com | (503) 684 7050 | www.nti-audio.com/XL2





Flexound introduced the HUMU Augmented Audio Cushion as a proof of concept for its vibro-acoustic technology in entertainment devices. “Hear and feel your favorite content without disturbing others around you,” is the pitch.



Expanding the HUMU Cushion concept into automotive, Flexound allows the sense of touch and personal full-range audio into a variety of seat back or headrests.



Inside view of the Earthquake MQB-1 Tactile Transducer, which uses the same technology as its big brother the Q10B at only 3 1/2" tall and 3 1/8" in diameter.

About the Author

Mike Klasco is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is a graduate from New York University, with post graduate work in signal processing, and he holds multiple patents licensed or assigned. For the past 35 years, Mike has worked on countless R&D projects for large and small companies. Mike specializes in materials and fabrication techniques to enhance audio performance.



and the moving magnet is directly connected to a movable surface (e.g., a floating floor or chair). Next to exciting the surface, Direct Drive is also able to create an offset of $\pm 15\text{mm}$. As a result, the surface will not only shake but also move, giving the audience the sensation of falling off or jumping over an object.

The other tactile product recently introduced by Powersoft is MEMO, a high-performance, ultra-low-frequency transducer and amplifier bundle, specifically designed for gaming and home theater applications. This solution overcomes the physical limitations of traditional systems to deliver sounds that extend well below the sub-harmonic spectrum. The extreme precision of the MEMO bundle enables the gamer to gain an advantage with more realistic and immediate feedback, without the need of cranking up the subwoofer's volume. In home theaters, MEMO enables users to keep the subwoofer's volume down while still delivering the full-frequency spectrum: This means that users can enjoy those bass-heavy cinematic sounds even more, again without cranking up the volume and disturbing the neighbors.

MEMO offers multiple mounting options and can be secured to any gaming chair, or added to a cinephile's favorite seat to achieve a fully immersive experience.

Powersoft has also introduced firmware for Mover, a subharmonic generator plug-in. This new algorithm transposes down the frequency pitch in order to reproduce ultra-low frequency extension all the way down to 5Hz. Note: Safety tip for readers—seat belts are recommended!!!

Earthquake Sound

Earthquake Sound is widely recognized in both subwoofers and tactile transducers for multiple entertainment applications, holding several patents on tactile transducers. Earthquake's bass shakers range from high performance to over the top. All models are “high-fidelity” with exceptional speed accuracy and efficiency, rendering complex low-frequency material in detail for home theater chair, simulation applications (car and flight simulators), casino gaming, virtual reality, and augmented reality (training, classes, and interactive applications).

The transducers boast aluminum alloy aerospace construction for thermal stability and no cycle per life failures. Furthermore, its patented inverted neodymium grade 5 magnets to center the mass, provide for linear spring constant as well as kinetic energy recovery. While the mass is set in motion, the centering magnet generates a force proportional to the motion by generating back electromagnetic force (EMF), which smoothly inverts the motion using the patented DynoMag technique. Another Earthquake

Sound patent uses two magnets at the extreme position to stop the sliders with a linear spring force, while generating an inverting force limited over excursion. The US government also relies on these indestructible shakers for engineering applications.

Leading the brand's available bass shaker range, the Quake Q10B is comparable to costly hydraulics and eliminates low end “resonating leakage” into adjoining areas. The Q10B answers the need for both accurate low frequency response and maximum tactile effect. One Q10B can replace three, four, or even five consumer shakers and move two or three theater seats or a sizeable platform. Compact in size, the Q10B can satisfy more space limitations than other transducers on the market and is almost indestructible and maintenance free, with only one moving part.

Next in the range is the MQB-1 Miniquake, which attaches to individual theater chairs, recliners, sofas, or even large platforms that house several chairs. In addition, it can be mounted to office chairs and thrones for a cool music integration or gaming experience. It uses the same technology as its big brother (Q10B), is half the size including the mounting feet, and it runs cool and efficient for hours.



An Earthquake Sound demonstration at a trade show always attracts the crowds. The company's bass shakers are guaranteed to create visceral impact and immersion to simulation systems.

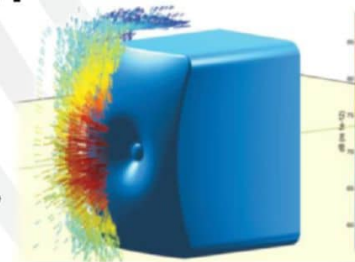
The Shell Shoxx package is specifically designed by Earthquake Sound for serious gamers who want to be “in the middle of the action.” The Shell Shoxx uses the MQB-XX patented shaker/tactile transducer, specifically designed to clamp onto any office chair base with its sturdy, vice-like arm. It complements computers or TV video games, especially first-person shooting, fighting, and war games, car racing games, and action-adventure games. The Shell Shoxx complete set also features the powerful Class J amplifier, an Earthquake design as an evolution on Class D, yielding

Independent Transducer Test & Measurement • Audio Systems
Prototyping • EASE & CLF Speaker Data • Klippel Sales,
Calibration and Service • Microflown Sales & Support



Now offering MICROFLOWN Sales and Support

Warkwyn is proud to offer Microflown Technologies' highly innovative testing equipment for sound visualization, sound power quantification, noise control and troubleshooting. We use Microflown's acoustic particle velocity and sound intensity probes in our own labs.



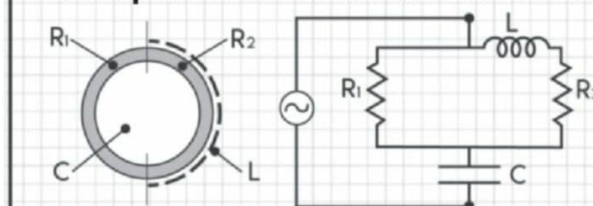
KLIIPPEL
Exclusive North American Sales Representative

Microflown Technologies
Sharing world wide
Sales and Support

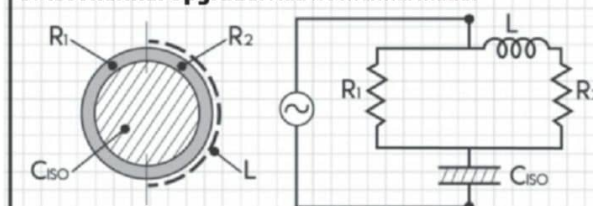
Warkwyn.com • Info@Warkwyn.com • 612.759.4042

IsoThermal TUBETRAP DESIGN KIT #2

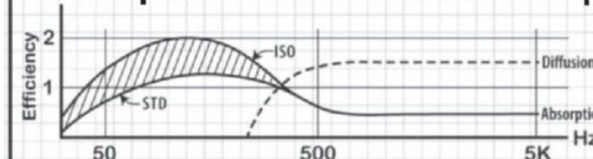
A: TubeTrap: Start with an Acoustic RLC Series Parallel Circuit



B: IsoThermal Upgrade: Add IsoThermal Matrix



C: TubeTrap + IsoThermal = IsoThermal TubeTrap



Save 10% on IsoThermals Use Coupon Code: audioXpress10
Acoustic Sciences Corporation | shop.acousticssciences.com



higher efficiency and unmatched electromotive control over any coil/magnet motor.

Under the Pillow

Flexound Augmented Audio is a Finland-based company working in close cooperation with the Aalto University, with a design office in London, England, and a subsidiary in Shanghai, China. It has focused on developing a strong patent and IP portfolio for personal sound and haptic experiences for entertainment, automotive and health, which they license and have also incorporated in new ventures.

Flexound introduced a pilot consumer product called HUMU Augmented Audio Cushion, which served as a proof of concept for the company's technology in



The Aurras is a thin, discrete, personalized bone-conduction speaker to slip under the pillow, designed to provide enough output to get through to the user and no one else in the bedroom. The Aurras receives the signal from a connected 1W Bluetooth amplifier with a built-in equalizer to optimize the sound, including a bass booster to provide enhanced low-end vibrations.

Resources

M. Klasco and S. Tatarunis, "Bass Shakers (Part 1): Enhancing the Deep Bass Experience with Tactile Energy, *audioXpress*, April 2014
<https://audioXpress.com/article/bass-shakers-part-1-enhancing-the-deep-bass-experience-with-tactile-energy>

M. Klasco and S. Tatarunis, "Bass Shakers (Part 2): Motion Platforms Are Bass Shakers on Steroids, *audioXpress*, May 2014
<https://audioXpress.com/article/bass-shakers-part-2-motion-platforms-are-bass-shakers-on-steroids>

Sources

Clark Synthesis | <https://clarksynthesis.com>
Crowson | <https://crowsontech.com/products>
Dreampad | <https://dreampadsleep.com>
Earthquake Sound | www.earthquakesound.com
Flexound | www.flexound.com
Powersoft S.p.A | www.powersoft.com
The ButtKicker | www.TheButtKicker.com
The Guitammer Co. | www.Guitammer.com

entertainment and therapy. HUMU enables users to hear and feel their favorite content without disturbing others around them, creating an immersive soundsphere with haptic augmentation, which serves both for entertainment, personal relaxation and meditation uses. With this same "cushion" concept, Flexound further explored applications for automotive, wellness and therapy, and integrated into cinema settings.

Flexound sells the key components, offers co-development services and licenses for co-branded end-products with the sales and marketing by the host brands. Its patented modules are intend to be designed into cushions, cinema or car seats, furniture, gaming chairs and more. Promising powerful bass with low listening volume, and hi-fi sound quality (20Hz to 20,000Hz range) from one elastic vibrating element—no loudspeakers needed, the company's approach shows minimal sound leakage, enabling personal "sonic spheres" and even ANC applications in cars or airplanes without headphones.

Flexound's key team members have more than 20 years of experience in international consumer product development, marketing, music business, audio hardware, and software.

Since we have mentioned the HuMu Cushion, we also need to mention Dreampad, which has been delivering high-quality music pillows for relaxation and sleep therapy since 2014. The Dreampad pillow was developed from a discovery that bone conduction sound produces a relaxation response in the body. Based on this therapeutic insight, the vibrational music technology was embedded in pillows to create a relaxing and restful bedtime sound experience using bone conduction technology, which permeates the entire pillow but not beyond, not bothering the person next to you in bed.

Taking the concept further, the Aurras is a new product from Dreampad and a unique audio appliance made specifically to be enjoyed in bed, without cumbersome headphones or earbuds, and without disturbing others. With a Bluetooth amp with integrated equalizer, the Aurras slips under the pillow and lets you listen to the news, movies, podcast, new age music or whatever. Research shows that the Dreampad's unique technology, combined with specially engineered music, can reduce stress and anxiety, helping users get to sleep and stay asleep.

In another article on tactile technology, we will expand the focus to a wide range of unconventional applications from automotive driver awareness, gaming belts and vests, sneakers, haptic headphones, and bone conduction earphones. 



Vintage Tone
Modern Technology
Lightweight Neodymium Magnet

Jensen® N12K

Cone Loudspeaker Radiation and Directionality

A loudspeaker sounds louder if it's pointed at you, right? And if you're behind the speaker, it's not as loud? Whoa, not so fast there! Let's take a look at the real-world behavior of speakers.

By
Richard Honeycutt

Among the “black arts” of loudspeaker and system design is understanding the interaction between the motion of a loudspeaker driver diaphragm and the resulting sound field in the vicinity of the loudspeaker. This article will present the basics of such interactions in a minimally mathematical fashion. Unless indicated otherwise, the principles discussed apply equally to any sort of radiating structure, not only the diaphragm of a familiar cone-type speaker.

Basic Concepts

When a cone vibrates in air, four results occur. The first is that a thin layer of air near the cone will vibrate back and forth along with the cone. This air effectively adds mass to the cone, and makes the cone require more force to vibrate, with that force increasing with frequency. The effect is similar to that of inductance in an electrical circuit, in which more voltage is required to push a current through an inductor as frequency is increased. In neither case is there any work done or power expended. This effect is described as acoustical radiation mass, and is considered as an acoustical inductance. An older name for acoustical inductance is inertance.

The second effect is present whenever any of the air that moves with the cone is forced into frictional or turbular motion (e.g., by obstacles or screens in the air path). In this case, some of the kinetic energy of the air molecules is converted to heat energy. This

effect is similar to the effect of electrical resistance. This effect is acoustical resistance.

The third effect is present whenever there is an enclosed or partially enclosed chamber into and out of which the moving air is forced, resulting in alternating expansion and contraction of a volume of air. Since the enclosure blocks direct passage of air, and since the extra force required as a result of this effect is greater at low frequencies, this acoustical compliance (inverse of acoustical stiffness) is analogous to electrical capacitance.

The fourth effect represents the acoustical energy that is radiated as sound waves. Since these represent power leaving the transducer system, they are also considered a form of resistance: acoustical radiation resistance. Naturally, this effect—often the smallest of the four—is actually the desired effect for which the loudspeaker was built.

Collectively, acoustical inductance, compliance, resistance, and radiation resistance make up the radiation impedance seen by the loudspeaker. In addition, there are mechanical elements resulting from the cone mass, suspension stiffness, and mechanical resistance of the cone and attached moving parts.

If we imagine a piston radiator (speaker cone) in an infinite baffle, the radiation mass and resistance are as shown in **Figure 1** [1]. In both cases, the values depend upon the relationship between wavelength and the circumference of the diaphragm. Note that the horizontal axis, calibrated

in values of ka , represents the value of circumference/wavelength. (For non-circular diaphragms, the results also apply pretty well for squares, with the error increasing for long, skinny radiating surfaces.)

These values are for one side of the cone, and must be doubled if radiation from both sides of the cone is taken into account. The piston in an infinite baffle can be used to represent a speaker mounted in a wall or other surface that is very large compared to the wavelength at the lowest frequency of radiation. Note that the effects of enclosures and obstacles are not considered in this general case, only radiation mass and radiation resistance.

A cone in a baffle that is not large compared to a wavelength has slightly (25%) lower radiation mass and very slightly higher radiation resistance than a cone in an infinite baffle.

At higher frequencies, where the wavelength is not long compared to the cone dimensions, things get much more complicated. The average value of the normalized radiation resistance remains at unity, but there are ripples of about a decibel or so above and below that value. Above the frequency where wavelength equals the circumference, radiation mass decreases.

Notice several implications from this information:

- (1) The effects of radiation impedance are quite small for most direct-radiator drivers in their usable frequency range. This is especially true for the $ka < 1$ region. Thus, the radiation impedance has little effect upon cone motion.
- (2) As a driver in a box transduces higher and higher frequencies, its performance transitions from that of a piston in the end of a tube to that of a piston in an infinite baffle. As this happens, the radiation mass increases by a factor of 1.35, and the radiation resistance decreases by a lesser amount.
- (3) Because the radiation resistance does not vary monotonically with frequency, but the radiation mass does (decreases with frequency above $ka=1$), and because they are represented by a parallel circuit, radiation resistance becomes dominant over radiation mass at high frequencies.

Directivity

In addition to the effects of radiation resistance, any acoustic radiator with constant phase over its surface radiates omnidirectionally at low frequencies, becoming more directional at higher frequencies. Such a loudspeaker is called a point source. The reason for the directionality can be appreciated if we remember Huygens' Principle, which is that any

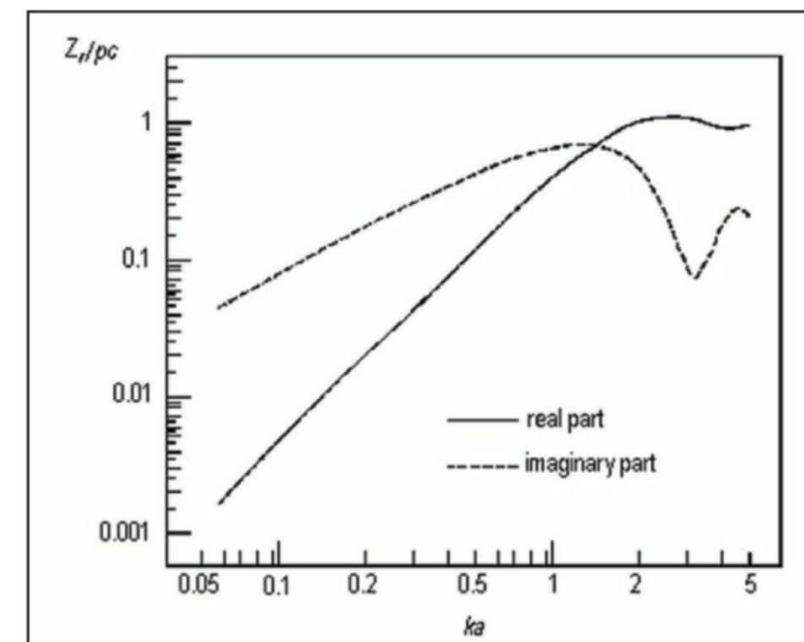


Figure 1: The real (resistive) and imaginary (reactive) parts of the Radiation impedance of a rigid circular piston in an infinite baffle increase monotonically with frequency up to $ka=1$. (ρc is the characteristic impedance of air.)

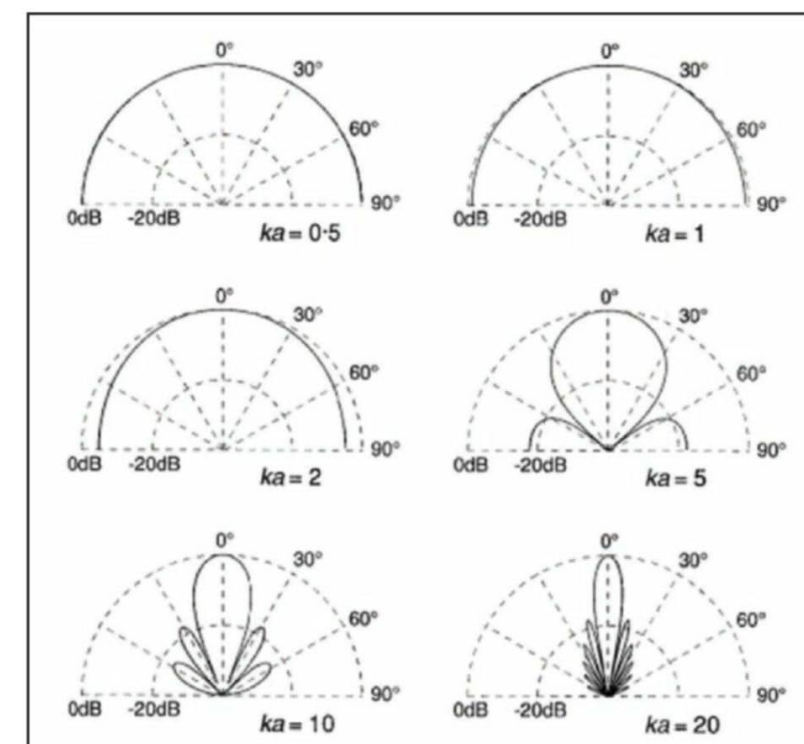


Figure 2: The directivity of a true (theoretical) piston radiator increases with frequency.

radiating surface can be accurately represented as if it were a set of individual small surfaces, each radiating omnidirectionally. If we name one such small subdivision of a cone "surface A" and another, "surface B," then we can identify the effects of travel time from each of these two surfaces to any given listening point. If a listening point is equidistant from both surfaces, sound radiations from the two will add. If at some frequency, the travel time for the wave from surface A is half a period longer than that from surface B, the two waves will arrive out of phase, and cancel. Through the use of calculus, it is possible to determine the combined response to the radiation from all the small surfaces of a cone, for any listening position. These combined responses can be plotted to give a picture of the directional radiation pattern of the cone at one specific frequency (**Figure 2**) [2]. Note the extreme beaming at high frequencies: over a 2-octave range, the coverage angle narrows from 360° to about 10°.

Real cones do not radiate in phase over their entire surfaces, except at low frequencies. So at higher frequencies, the directional pattern of a real cone is different from the theoretical pattern for a piston of the same size. In general, the effects of cone breakup (different parts of the cone radiating in various phase relations with each other) cause a speaker to radiate over a wider solid angle at higher frequencies than would a true piston.

A true pistonic radiator would radiate with all points on the piston in phase. A real loudspeaker behaves similarly to a true piston at the lowest frequencies. In this range, there is no effect from wave propagation in the cone as we move from the

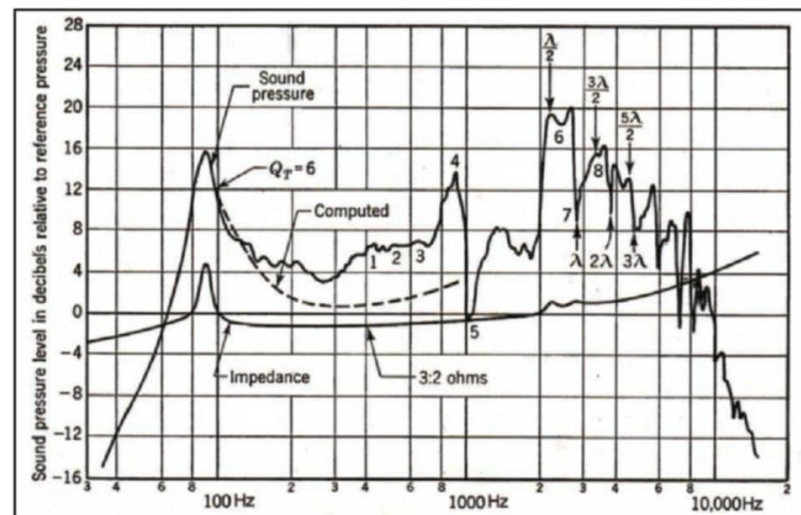


Figure 3: The frequency response of an inexpensive 8" speaker shows the effects of breakup modes.

center to the edge. However, as frequency rises, waves propagate radially outward from the center. At positions where the circumference of the cone is a whole number of wavelengths, portions of the cone can move in opposite directions from each other.

Such behavior is called circumferential mode vibration. Where the radius is a whole number of wavelengths, portions of the cone can move in opposite directions, and we have radial mode vibration. A cone radiating a complex wave (more than a single frequency at the same time) can exhibit both types of modes. These breakup modes strongly affect the frequency response of the cone, as shown in **Figure 3** [3]. The response curve is labeled to show the frequencies at which the cone radius corresponds to various multiples of a half-wavelength. These frequencies are controlled by cone material, angle, profile, and corrugations; the wavelength in question is that of sound in the cone material, not sound in air. Also labeled are cross-references to the cone drawings shown in **Figure 4** [4], which illustrate the breakup modes.

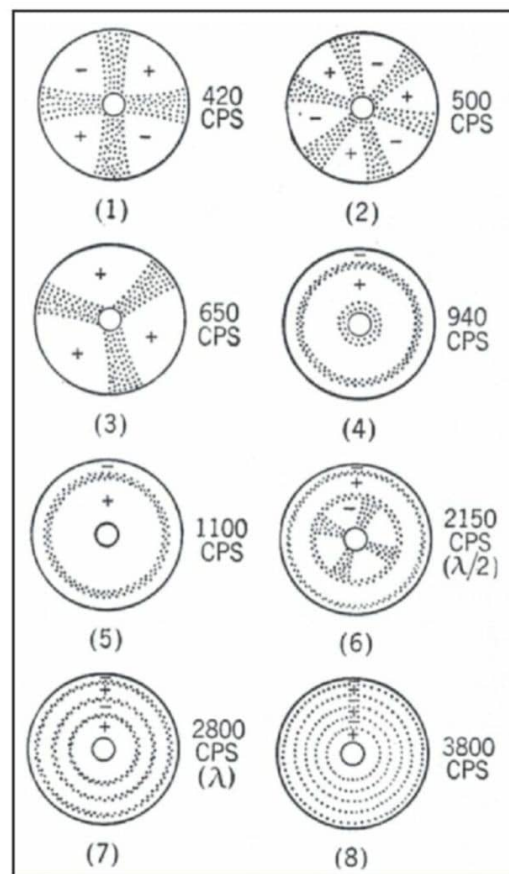


Figure 4: These drawings illustrate the cone breakup modes of the 8" speaker whose frequency response is shown in Figure 3. (CPS means "cycles per second"; i.e., Hertz.)

Modes #1 through 3 are circumferential modes, 4, 5, 7, and 8 are radial modes, and 6 is a combination of modes. The + and - signs indicate cone-surface motion up (out of the page) or down (into the page).

Cone breakup has two effects: It increases the average high-frequency output of the loudspeaker compared to that of a true piston, and it broadens the directivity. **Figure 5** illustrates the directivity of a real 15" speaker at 63Hz, 250Hz, and 1kHz.

One other class of radiator can be used to achieve directivity, and this one does work at low frequencies, even without requiring huge radiators. The simplest member of this class is an unbaffled speaker, also called a doublet radiator. Since the speaker radiates equally, but in opposite phase, from both sides of the cone, all radiation is cancelled in the plane perpendicular to the axis of the cone. Thus, the directivity takes the form of a figure-8 as shown in **Figure 6**.

If appropriate acoustic resistance is placed on the back side of the cone, other patterns, such as hypercardioid and cardioid (like microphone patterns) are available. The cost associated with this technique is much greater cone excursion

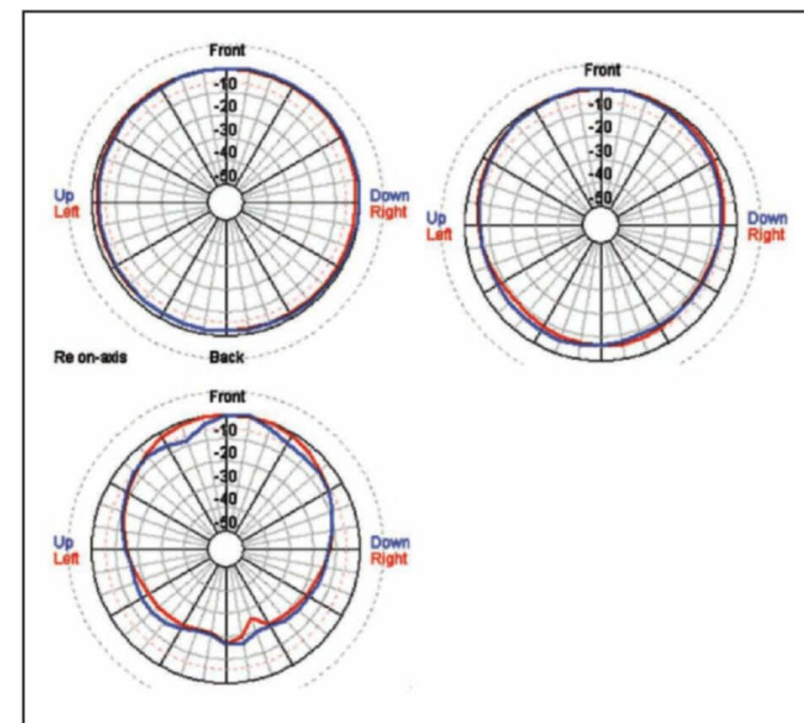


Figure 5: The polar response (directivity) of a 15" speaker shows much less beaming than a true piston would exhibit.

KLIPPEL QC

GET THE RIGHT FEEDBACK

Quality Control of Actuators and Haptic Devices

- ▶ Acceleration, resonance, Q factor, ...
- ▶ Contact-free laser measurement
- ▶ Automatic PASS/FAIL verdict
- ▶ Abnormal sound test
- ▶ Piezo or solenoid actuators, ERM, LRA
- ▶ Smartphones, automotive, gaming, wearables, ...

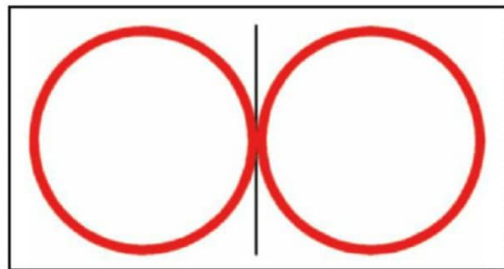
www.klippel.de

E-mail: info@klippel.de

Phone: +49 351 5019390

Dresden, Germany

Figure 6: A dipole speaker (acoustic doublet radiator) has a figure-of-eight directionality.



for a given amount of low-frequency output. This places high demands on speaker mechanical and thermal design. Cardioid, hypercardioid, and arrays of greater directionality can also be created using multiple drivers carefully spaced and/or phased and level-controlled with respect to one another.

Loading: Boundary Effects and Mutual Coupling

We have seen that the radiation impedance of a cone in an infinite baffle is greater than that of a cone in a small box. One way to understand this effect is to think of a cone mounted in front of an infinite wall, and radiating omnidirectionally. Some of this radiation will reflect from the wall behind the speaker, interfering constructively (adding) and destructively (cancelling) with the direct radiation from the speaker.

At the lowest frequencies, the path length difference between the direct and the reflected sound is such a small portion of the wavelength that they add constructively. As the cone is moved closer to the wall, the maximum frequency at which this constructive interference occurs becomes higher, until, with the cone actually mounted in the wall,

there is constructive interference at all frequencies. The result is twice the sound pressure (6dB higher SPL) for the cone in the wall than there would be at the same listening distance if the cone were mounted in the middle of the room.

This effect can be repeated if the cone is now mounted in the corner of two plane surfaces, such as at the intersection of a floor and vertical wall—another 6dB increase. In a three-way corner, such as the upper or lower corner of a room, we get another 6dB, for a whopping 18dB increase compared to the cone in the middle of the room. A corner-mounted cone speaker will have its directivity constrained by the walls of the room at frequencies where the coverage angle would otherwise be greater than 90°. At higher frequencies, the directivity is not affected by corner mounting.

The question naturally arises: What counts as “the corner”? Basically, if the center of the cone is located within a distance x of the geometrical corner of the room, it acts as a true corner radiator for frequencies at which x is under $1/8$ wavelength. At higher frequencies, the response can become ragged, due to interference effects between the direct and reflected waves.

Two points are seldom mentioned in discussions such as this. First is that the corner effect becomes insignificant at frequencies for which the radiator is directional enough that little sound actually bounces back from the walls. Second, when the distance from the cone to the wall is great enough, the reflected waves are significantly reduced in intensity by the time they recombine with the direct waves, and in large rooms they may also be delayed enough so that the ear/brain considers them a separate acoustic event from the direct wave, so they are interpreted as reverberation.

A similar effect occurs when two or more drivers are mounted close together. At frequencies for which the distance between cone centers is under about one-quarter wavelength, each cone increases the radiation loading of the other so that each speaker produces twice the SPL at a given distance that it would produce alone.

Thus, using two speakers gives four times the acoustic power, using only twice the electrical power. The name for this effect is mutual coupling. For 30cm (12”) speakers mounted with rims touching, this effect would persist up to about 300Hz. At higher frequencies, the average output from each speaker will drop by half, and response irregularities caused by interference will occur. And since the loudspeaker is no longer axisymmetrical, the polar pattern will not be circular. 

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.



References

- [1] Keith R. Holland, “Principles of Sound Radiation,” *Loudspeaker and Headphone Handbook*, 3rd Edition, John Borwick, Editor, Focal Press, Oxford, 2001, p.15.
- [2] Ibid, p. 14.
- [3] Leo Beranek, *Acoustics*, McGraw-Hill, New York, 1954, p. 198.
- [4] Ibid, p. 199.

Whatever your testing challenge, GRAS has the right acoustic sensor for your audio application.

Only GRAS offers a complete line of high-performance, test microphones and related products ideal for use in consumer audio and electronics applications.

As the leading global provider of microphones, GRAS has a long tradition of working with audio engineers to ensure accurate data is captured, each and every time.

Microphones from GRAS are designed for the high quality, durability and reliability that our customers demand.

Contact GRAS today for a free evaluation of the perfect GRAS microphone for your application.

grasacoustics.com

GRAS

- > Measurement microphone sets
- > Microphone cartridges
- > Preamplifiers
- > Low-noise sensors
- > High frequency ear simulators
- > Head & torso simulators
- > Test fixtures
- > Custom designed microphones
- > Speech intelligibility
- > THD
- > Frequency response
- > Calibration systems and services



Ask about our new Hi-Res Ear Simulators! Uses a 1/4" microphone to measure up to 50kHz



Need an analyzer?

Audio precision

www.ap.com

GRAS Sound & Vibration

Simple Improvements to a Recent Audio MFB Filter Lowers Harmonic Distortion

By
Michael Steffes
(United States)

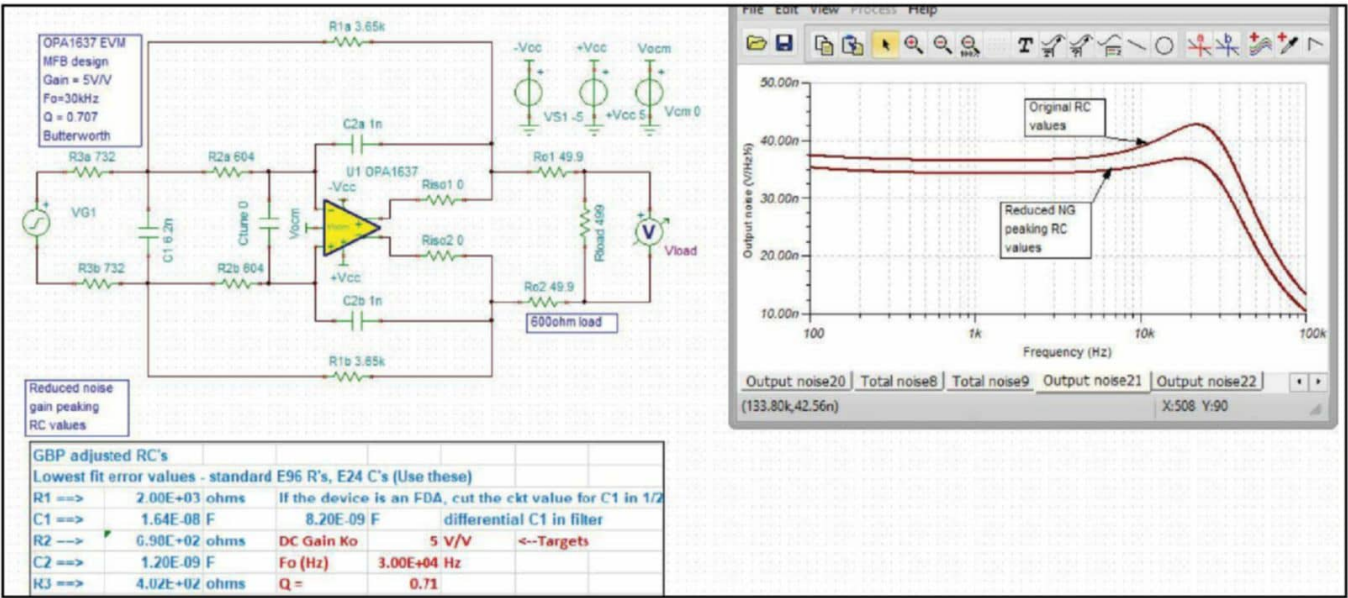
In this article, we try to improve the basic design of a fully differential Evaluation Module (EVM) board that implements a second-order multiple feedback (MFB) filter by improving the noise gain (NG) profile.

A recent audio fully differential amplifier evaluation module (EVM) board implements a second-order multiple feedback (MFB) filter at 30kHz (Figure 1 from Reference 1). This basic design can be further improved looking deeper into the noise gain (NG) profile latent within every MFB design. Free enhancement to this design can be delivered with RC values changes, where a more significant step down in distortion (>10dB) can be achieved substituting a lower power, pin-compatible FDA into the same improved RC value design.

The example design in Reference 1 provides a differential input to differential output gain of 5V/V, DC-coupled, 30kHz second-order Butterworth filter design operating on ±5V supplies. MFB filter designs

can be implemented with literally an infinite number of RC combinations—and still get the same signal path response. Internal to any active filter design lies a NG profile for the filter itself. The NG always peaks at least as much as the desired filter shape but then also adds peaking due to NG zeroes in the response. Equation 1 shows the NG transfer function for the second-order low-pass MFB filter re-cast as much as possible in terms of the desired filter response. Most of the NG response is set by the desired filter where:

k is desired filter DC gain
ω_o is filter characteristic frequency
Q_p is complexness of poles, Butterworth is Q_p=0.707



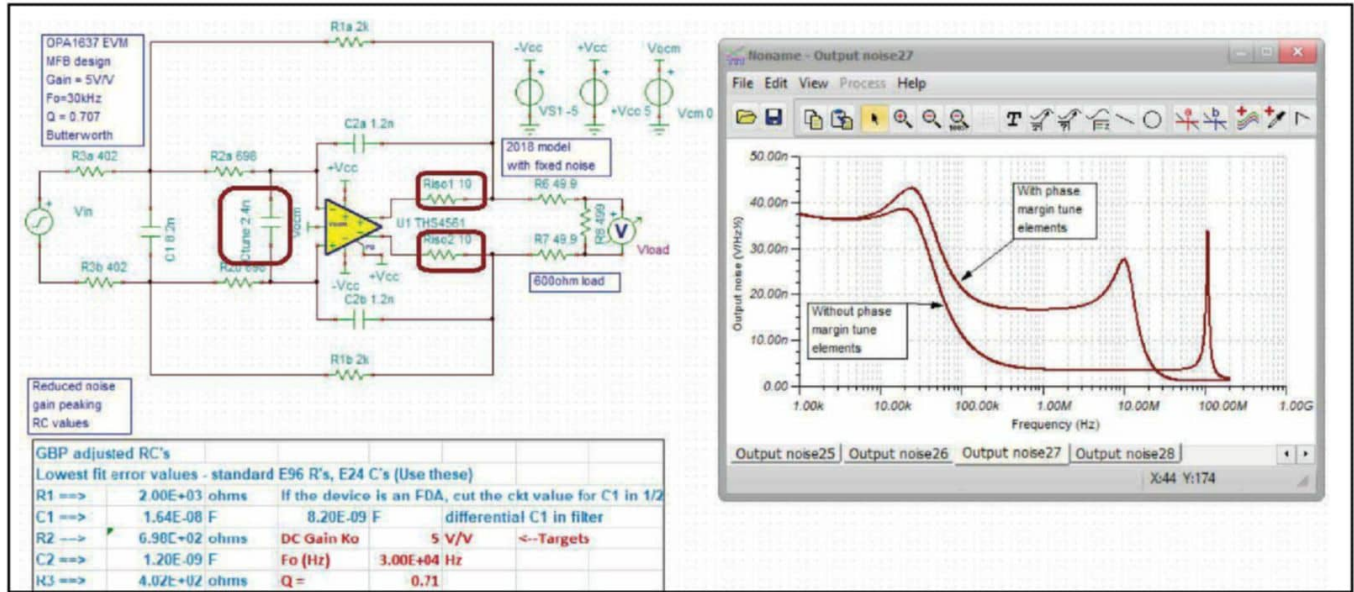


Figure 4: Wideband differential output spot noise simulation showing peaking at LG=0dB crossover.

differential input impedance from the original 1.46kΩ to 804Ω.

Using Equation 3, the Q_z has increased from the original design value of 0.21 to 0.34 which in turn has moved the lower NG zero up from 19.3kHz to 26.4kHz. Generating the LG (in decibels) over frequency is a simple matter of subtracting the NG magnitude (Equation 1, in decibels) from the open-loop gain (Aol) using the TINA model 8.5MHz gain bandwidth product OPA1637 (Reference 3). **Figure 2** shows the LG for both the original RC values along with the slight improvement using the lower NG peaking values in Figure 1. It is this LG over frequency that corrects the open-loop output stage non-linearity to the very low harmonic distortion levels delivered by the OPA1637 in the MFB application of Reference 1.

In this particular example, only slight (but free) output noise and LG improvements can be delivered

with the RC adjustments shown in Figure 1. These improvements are much more pronounced for higher Q filter designs.

Applying a Higher Gain Bandwidth Product FDA to this Audio MFB Filter Design

Using a device offering a higher GBP adds directly to the LG over the audio band of interest. This should deliver even lower harmonic distortion than this already <-105dBc solution using the OPA1637. **Table 1** summarizes some key comparisons between the OPA1637 and the lower maximum supply voltage (faster IC process) THS4561 (Reference 4).

Simply dropping this higher GBP THS4561 into the improved RC value MFB circuit of Figure 1 will provide an increase in LG across the audio band given by $20 \times \log(61/8.5) = 17.1\text{dB}$. All other things being equal, this increased GBP should drop the harmonic

distortion terms by approximately that amount over the OPA1637 solution. **Figure 3** shows this loop-gain comparison sweep for this MFB filter design example.

The increased GBP for the THS4561 on lower supply current comes at the cost of a reduced nominal phase margin. The MFB filter shapes that NG to 1V/V at higher frequencies due to the feedback capacitor inherent in the topology. LG phase margin simulations (Reference 5) show the THS4561, applied directly into the design of Figure 1, to have a nominal phase margin of 25°. While that might be acceptable for production, it is a simple matter to improve this (with no audio band degradation) adjusting the phase margin tuning elements in Figure 1. In fact, setting the differential input C_{tune} to 2.4nF and the output R_{iso} elements to 10Ω will improve the nominal phase margin for this THS4561 implementation to 36°. A quick approach to checking low phase margin is to run a wideband differential output spot noise simulation—a sharp peak suggests a low phase margin condition. **Figure 4** shows this using the 2018 original THS4561 TINA model with and without these tuning elements.

The original 25° phase margin design shows a little lower noise peaking in the 20kHz region but a

very sharp noise peak at LG=0dB crossover around 100MHz. Adding the differential input $C_{\text{tune}}=2.4\text{nF}$ peaks the noise a bit more around 20kHz, and pulls the LG=0dB crossover way back to around 10MHz, with lower noise peaking in that region due to the improved 36° phase margin. Attaching the feedback Cs on the output side of the 2- R_{iso} resistors has also improved phase margin.

While certainly credible, the example MFB filter of Reference 1 can be incrementally improved through simple RC value adjustments. Where a higher GBP device at even lower power (the THS4561) can be applied, a direct improvement in harmonic distortion should be delivered due to the higher LG across the audio band. **ax**

About the Author

Michael Steffes has worked since 1982 in high-speed signal path IC development spanning design roles for Comlinear Corp. (the original current feedback originator) with product definition and launch roles through National Semiconductor, BurrBrown, Texas Instruments, and Intersil. He continues to focus on High-Performance ADC interface solutions, DSL/PLC line driver design, and signal path optimization tools. Providing new product data sheet and applications collateral for more than 150 industry leading amplifier products has led to more than 150 contributed articles and application notes since 1988.



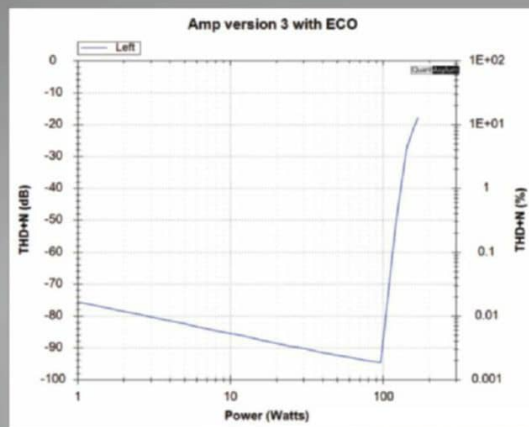
References

- [1] "OPA1637 Evaluation Module User's Guide," Texas Instruments, revised June 2020, <https://www.ti.com/lit/ug/sbou235a/sbou235a.pdf>
- [2] M. Steffes, "Design Methodology for MFB Filters in ADC Interface Applications," Texas Instruments Applications Note, February 2006, <https://www.ti.com/lit/an/sboa114/sboa114.pdf>
- [3] "High Fidelity, Low Noise, Fully Differential, Burr-Brown, Audio Op Amp," Texas Instruments OPA1637 datasheet, revised Aug. 2020, <https://www.ti.com/lit/ds/symlink/opa1637.pdf>

- [4] "Low Power, High Supply Range, 60MHz, Fully Differential Amplifier," Texas Instruments THS4561 Datasheet, revised August 2020, <https://www.ti.com/lit/ds/symlink/THS4561.pdf>

- [5] M. Steffes, "Stability Issues and Resolutions for High Speed Fully Differential Amplifiers (FDAs), Insight #8", Planet Analog, May 1 2019, <https://www.planetanalog.com/stability-issues-and-resolutions-for-high-speed-fully-differential-amplifiers-fdas-insight-8>

Fast Power Amp Characterization



Complex measurements in minutes
Amp efficiency, THD, FR, Output Z
Solutions starting at \$449

QuantAsylum

www.QuantAsylum.com
Designed and built in the State of Washington USA

Four Digital Stylus Force Gauges

If you want to get the most from your turntable, tonearm, and phono cartridge, an accurate stylus tracking force gauge is essential, whether you're playing vinyl LPs, 78-rpm records, or 16" transcription discs. Many manufacturers refer to "stylus pressure" as "stylus force"—the two terms are interchangeable. Gary Galo tests a few of the gauges currently available.

By
Gary Galo
(United States)

- Jelco SG-1 Stylus Pressure Gauge**
www.jelco-ichikawa.co.jp
\$70.00 US from Merrill's Analog Emporium
www.hifigem.com
\$109 US from LP Gear
www.lpgear.com
- Neoteck NTK002 Digital Stylus Force Scale Gauge**
www.neoteck.cn
\$14.95 US from Amazon.com
Available in black, jet black, or silver
- Neoteck EMB-02 Digital Stylus Force Scale Gauge**
www.neoteck.cn
\$15.99 US from Amazon.com
Black only
- Riverstone Audio RL100 Digital Stylus Force Gauge**
www.riverstoneaudio.com
\$26.99 US from Amazon.com
Available in graphite or silver
- docooler YTA229 Chrome-Plated Calibration Weight Set**
1g 2g 5g 10g 20g
\$8.88 US from Amazon.com



Photo 1: Here are two popular analog stylus tracking force gauges—the AR (a) and the Shure SFG2 (b). Both have been discontinued, but Esoteric Sound is selling a clone of the AR, and many online dealers have remaining stock of the Shure. (Source: Photo 1b, Shure Brothers)

In the old days, all stylus pressure gauges were “analog” devices, usually involving some type of balance beam and counterweight (**Photo 1**). The old Acoustic Research (AR) turntables were supplied with one of the most popular; AR also sold them separately. The classic AR gauge consists of a plastic balance beam and pivot, with a resting place for the stylus on one end and a ring depression for positioning small weights on the other. The AR gauge came with four plastic weights: 1/4, 1/2, 1, and 2 grams. If additional force was needed, the manual noted that a penny could be used as a 3-gram weight.

But, that was back in the 1960s and 1970s when pennies were solid copper. Since 1982, pennies have been made with copper-coated zinc. According to the US Mint, a penny now weighs 2.5 grams. If you need even heavier tracking forces, for 78-rpm records, a nickel still weighs 5 grams.

Another popular balance-beam stylus force gauge is the Shure SFG-2. This device has a sliding weight on one end, and two points for the stylus on the other side of the pivot: Times 1 and Times 2. Maximum tracking force is 3 grams. These gauges have been around for decades, but both are now discontinued. Several internet dealers are selling the remaining stock of the Shure, including Esoteric Sound, LP Gear, and Music Direct. Esoteric also sells a clone of the AR gauge under its own Rek-O-Kut brand. Esoteric's version comes with the same weights as the original, plus an extra 2-gram weight. If you need to measure forces higher than 5.75 grams, you can purchase an extra set of weights separately, or add a nickel to the mix.



Photo 2: The Neoteck NTK002 is supplied with a case, batteries, and a 5-gram calibration weight (a). Maximum tracking force is 5.00 grams. The gauge is shown in action with the author's Jelco SA-750D tonearm and Denon DL-103R cartridge (b). (Source: Photo 2a, Amazon.com)

Digital Choices

Digital stylus pressure gauges now dominate the world of analog records, and have the potential for much greater accuracy than the old-fashioned balance-beam types. Prices are all over the map, ranging from less than \$20 US each, to \$250 and more. The very expensive models come from respected high-end audio manufacturers, including Clearaudio, Rega, Project, and Ortofon. The middle ground is occupied by Jelco—Internet dealer prices for its SG-1 gauge range between \$70 and \$109.

But, is it necessary to spend a fortune on a stylus force gauge for accurate results? That's what I set out to determine. I purchased the Neoteck NTK002, Neoteck EMB-02 and Riverstone RL100 gauges from Amazon (**Photos 2–4**; the EMB-02 was previously sold as the DD001). I bought the Jelco SG-1 from George Merrill's Analog Emporium (**Photo 5**). George Merrill is one of the world's premiere turntable designers, and Analog Emporium is the home of



Photo 3: The newer Neoteck EMB-02 is supplied with a protective cover, a case, batteries, and a 10-gram calibration weight. The gauge will operate over a range of 0.03 grams to 100 grams. The photo on Amazon shows the previous model number, DD001. (Source: Amazon.com)



Photo 4: The Riverstone Audio RL100 comes with 5 gram and 20 gram calibration weights and a special cantilever extension arm to place the stylus at the approximate thickness of an LP record (a). The device is shown in use with the author's archival playback setup—a Stanton 500AL cartridge and 3.3 mil truncated-elliptical stylus mounted on a Jelco SA-750L tonearm. The nickel glued to the Jelco headshell adds mass necessary to achieve a tracking force of 7.0 grams for shellac 78-rpm records (b). (Source: Photo 4a, Amazon.com)

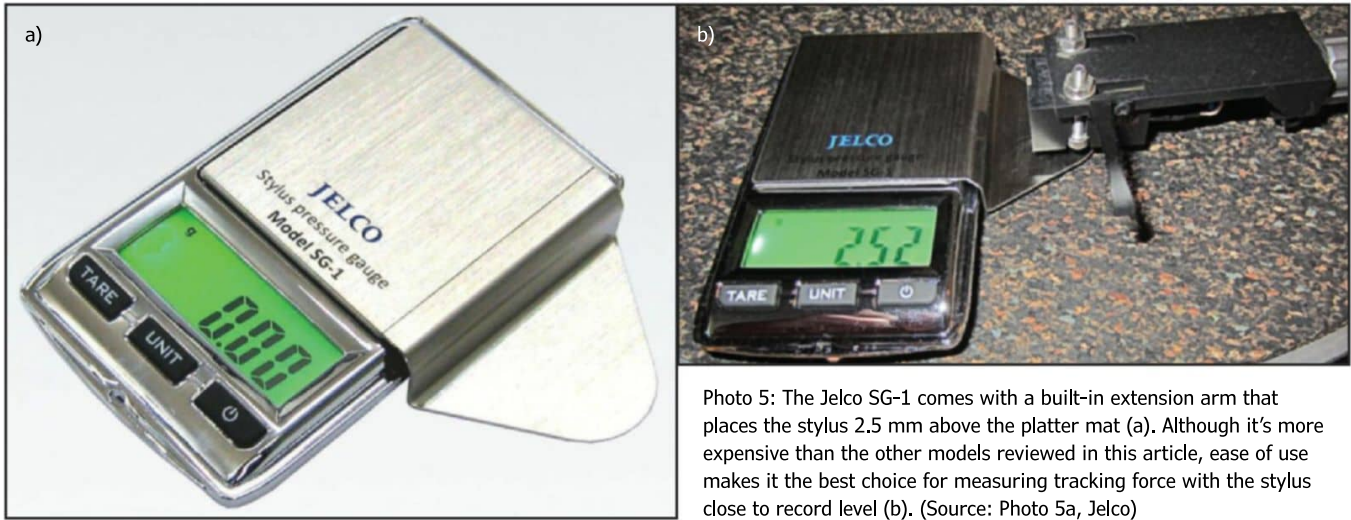


Photo 5: The Jelco SG-1 comes with a built-in extension arm that places the stylus 2.5 mm above the platter mat (a). Although it's more expensive than the other models reviewed in this article, ease of use makes it the best choice for measuring tracking force with the stylus close to record level (b). (Source: Photo 5a, Jelco)

his PolyTable turntables. All these gauges are made in China and based on strain-gauge technology. I bought two of the Neoteck NTK002 units to check for unit-to-unit consistency. Units appearing identical to the NTK002 can be found on the Internet sporting names like Revolv and Pfanstiehl, at roughly three times the price of the Neoteck.

The Neoteck NTK002 operates up to 5 grams, whereas the EMB-02 and Riverstone RL100 will measure as high as 100 grams. Tracking forces above 5 grams are sometimes needed in archival applications, particularly shellac 78-rpm records. I use a Stanton 500AL cartridge with several truncated conical and elliptical styli, which track at up to 7 grams. The Neoteck gauges come with a calibration weight, LR-44 type batteries and a carrying case. The Riverstone has two calibration weights, 5 grams and 20 grams, and a carrying case, but the single AAA battery is not included. The Jelco does not come with a calibration weight—this gauge is calibrated by zeroing the display using the TARE button. The calibration procedures are simple and are explained in the instruction sheets. Manufacturer's specifications

are shown in **Table 1**.

Set up of the Neoteck gauges is simple—just remove the protective cover. The Riverstone RL100 requires some assembly. The RL100 comes with a special cantilever extension arm, which is designed to place the stylus at the approximate thickness of an LP record. Riverstone described its product as a “record level” tracking force gauge. I found the cantilever arm difficult to precisely attach. Once attached, the unit was easy to operate. *RIAA Bulletin E1*, dated 1978, specifies the nominal thickness of a flush-design (perfectly flat) 12” LP Record as 1.9 mm.

Since the mid-1950s, most LPs have been pressed with something called Groove Guard—the edges and label area are raised relative to the playing surface. This was to keep the playing surfaces from touching one another when LPs were stacked up on record changers, a horrifying thought to any audiophile. But, the playing surface on most Groove Guard records is flat. A number of audiophile labels including Analogue Productions, have dispensed with Groove Guard, since their customers invariably prefer all-manual turntables, and record changers have mercifully disappeared from the marketplace. Some LP records are contoured—the thickness of the record changes across the playing surface. According to a drawing from the RIAA document, much of the playing surface of a contoured record can be as thin as 1.1mm. The RIAA document is long out of print, but a PDF can be found in the Supplementary Material section of the audioXpress website (see Project Files for the link).

Riverstone claims that its extension arm places the stylus 2mm to 3mm above the supporting surface, which will usually be the platter mat. I checked the Riverstone gauge with my digital

calipers and found that the resting point for the stylus is approximately 2.5mm above the mat. The Jelco SG-1 comes with a built-in extension arm, but they don't mention putting the stylus resting point at record level. According to my measurement, the resting surface is 2.5mm above the mat. This is in the ballpark for many flush-design audiophile records. I measured a 200-gram audiophile pressing and found the thickness to be 2.25mm. But, most commercial LPs are considerably thinner. The stylus resting point for the Neoteck gauges is about 6.5mm above the supporting surface, and the tracking forces of many tonearms do vary slightly with vertical stylus position.

Measured with the Neoteck NTK002, my Jelco SA-750D drops 0.04 grams if I remove my platter mat, which has a thickness of 3mm. A mat thickness of 5mm on the SA-750D results in a tracking force difference of 0.07 grams with and without the mat. While the Riverstone and Jelco may be optimum for a handful of audiophile pressings, none of these gauges will duplicate the vertical stylus resting position for the vast majority of vinyl LP records. If you worry about 0.05 grams, you can drive yourself crazy going back and forth between vertical tracking

	Neoteck NTK002	Neoteck EMB-02	Riverstone RL100	Jelco SG-1
Range	0.01-5	0.03-100	0.01-100	0.01-20
Accuracy	± 0.01	± 0.01	± 0.01	± 0.01
Resolution	0.01	0.01	0.005	0.01
Cal Weight(s)	5	10	5 & 20	None

Table 1: These are manufacturers' specifications for the four digital stylus force gauges under review. All units are in grams. The calibration weights are those supplied with the units.

angle (VTA) adjustments and stylus force.

None of the product descriptions for the expensive gauges from the manufacturers I cited earlier mention anything about measuring the stylus force at record level, but the Ortofon DS-3 (\$150), the Clearaudio WeightWatcher (\$275), and the Rega Atlas (\$245) all appear to come very close. In fact, the Ortofon looks virtually identical to the Jelco, at roughly double the price (George Merrill, who is also an Ortofon dealer, confirmed that they are essentially the same gauge). But, the only feature that the additional cost gets you is a PCS function.

As Ortofon describes it, the “PCS button balances count by calculating the average weight of one piece-weight called a unit weight, then applying it to the total weight of what you are trying to count. First, weigh one PC, then weigh the bundle, then push the PCS button, and on the

About the Author

Gary Galo retired in 2014 after 38 years as Audio Engineer at The Crane School of Music, SUNY at Potsdam, NY. Since then he has worked as a volunteer in the Crane Recording Archive, transferring vintage Crane recordings to digital. He is the author of more than 300 articles and reviews in more than a dozen publications, and he has been writing for *audioXpress* and its predecessors since the early 1980s. Gary is a long-time active member of the Association for Recorded Sound Collections (ARSC), a frequent presenter at ARSC conferences and the author of numerous articles and reviews in the *ARSC Journal*. He is a member of the Boston Audio Society and a Life Member of the Audio Engineering Society.



The Electrostatic Speaker Design Cookbook
is a complete guide for the novice and the experienced builder to successfully build this unusual sound reproduction device.

THE ELECTROSTATIC LOUDSPEAKER DESIGN COOKBOOK
by ROGER R. SANDERS
with a General Edit Manager by Barry McCune

cc-webshop.com

DO IT YOUR WAY
Cables/Panels/Boxes
WITH REDCO AUDIO'S
“Build Your Own” web pages

Full control for design, editing and more.
AND WE'LL BUILD IT JUST THE WAY YOU WANT!

REDCO AUDIO
www.redco.com 1-203-502-7600



Photo 6: This is the docooler Calibration Weight Set sold by Amazon. The set comes with 1, 2, 5, 10, and 20 gram weights. (Source: Amazon.com)

display you’ll get a number of PCS.” This isn’t likely to be of any use when measuring stylus tracking force. Incidentally, stylus resting points for the two analog gauges were around 4mm for the Shure and 6mm for the AR.

Accuracy Tests

I calibrated the Neoteck and Riverstone gauges with the supplied calibration weights—5 grams for the Neoteck NTK002 gauges and the Riverstone, and 10 grams for the newer Neoteck EMB-02. The Jelco was calibrated by zeroing the unit using the TARE button. I then checked each gauge at 1, 2, and 5 grams using a third-party set of weights—the docooler YTA229 Calibration Weights sold by Amazon (Photo 6).

The docooler weights have a permissible error specification of ±0.003 grams, which should be sufficiently accurate for this purpose. The manufacturer notes that they are suitable for “suitable for general laboratory, commercial, and educational” use. They are not professionally certified.

The test results are shown in Table 2. Nearly all the results are excellent. The Neoteck EMB-02 showed a 4% error at 1 gram, but most of the other readings were within 1% or spot on. For tracking forces of 5 grams or less, the Jelco SG-1, Neoteck NTK002 and Riverstone RL100 provided the best accuracy. The two Neoteck NTK002 samples showed identical results at 2 grams and 5 grams, and only a 1% difference at 1 gram. The EMB-02 is still very good. I’m finding the EMB-02 and the Jelco SG-1 especially useful for the higher tracking forces required by my 78-rpm styli.

Analog Gauge Tests

I also tested the accuracy of the two analog stylus force gauges mentioned earlier. My procedure was to adjust the tracking force using the analog gauge, and then check the stylus force on the Neoteck NTK002, sample No. 1. I used the Neoteck because the stylus resting place is closer in height to the analog models than the Jelco or Riverstone.

With the AR measuring a tracking force of 1.25 grams, the Neoteck read 1.42; with the AR measuring 2.5 grams, the Neoteck read 2.72. These are errors of 14% and 9%, respectively. The AR gauge I tested is an original, from the 1970s, not the clone sold by Esoteric. The Shure fared better.

With the Shure reading 1.25 grams, the Neoteck read 1.3; with the Shure at 2.5, the Neoteck read 2.54. The errors are 4.8% and 1.8%. Shure claimed an accuracy of ±0.1 grams for the SFG-2, so my sample was well within spec.

Conclusions

Digital stylus force gauges offer greater accuracy than the well-known analog, balance-beam types that I tested. The good news is that you don’t need to spend a fortune.

Though the Neoteck gauges will certainly do the job, the Jelco SG-1 occupies a sensible middle ground on price, and offers more elegant appearance and construction than the Neoteck models (that judgement is, admittedly, subjective). If you prefer measuring the tracking force at close to record level, the SG-1 does this without the fiddly assembly required by the Riverstone RL100.

Sadly, as I was finishing this review, Jelco announced that it was going out of business, though its website is still up (see my sidebar article about Jelco for further details). Dealers may have remaining stock of the SG-1, and George Merrill noted that he would not be surprised if it surfaces

Docooler YTA229 Cal Weights	Neoteck NTK002 No. 1	Neoteck NTK002 No. 2	Neoteck EMB-02	Riverstone RL100	Jelco SG-1
1	1.01	1.02	1.04	1.015	1.01
2	2.00	2.00	2.01	2.00	2.00
5	5.00	5.00	5.01	5.00	5.00

Table 2: These are the accuracy test results using the docooler YTA229 Calibration Weights (all units are in grams). Each gauge was first calibrated with its supplied weight—5 grams for the Neoteck NTK002 gauges and the Riverstone, and 10 grams for the newer Neoteck EMB-02. Then, I took measurements with the docooler 1, 2, and 5 gram weights. The Jelco SG-1 is not supplied with a calibration weight.

under another name (it’s an OEM product made in China).

Regardless of which digital gauge you choose, I recommend getting a set of calibration weights so you can periodically check the gauge for accuracy and repeatability. Any gauge that doesn’t give repeatable measurements is defective and should be returned for a replacement or refund. If you still play analog records and want to get the most out of your playback system, you should own a digital stylus force gauge. ax

Project Files

To download the “RIAA Bulletin E1,” visit:
<http://audioxpress.com/page/audioXpress-Supplementary-Material.html>

Resources

M. Fremer, “Is Tonearm Manufacturer JELCO a Covid-19 Victim?”
www.analogplanet.com/content/tonearm-manufacturer-jelco-covid-19-victim

“RIAA Bulletin E1,” Linear Audio, http://linearaudio.nl/sites/linearaudio.net/files/RIAA%20Bulletins%20E1%20%26%20E4_1978%20LP%20dimensional%20standards.pdf

US Coin Specifications,
<https://www.usmint.gov/learn/coin-and-medal-programs/coin-specifications>

Sources

AR/Rek-O-Kut Stylus Force Gauge
Esoteric Sound | www.esotericsound.com/access.htm

Jelco SG-1 Stylus Force Gauge
Jelco | www.jelco-ichikawa.co.jp
George Merrill’s Analog Emporium | www.hifigem.com

Ortofon, Rega, Clearaudio, and Project Gauges
Audio Advisor | www.audioadvisor.com
Music Direct | www.musicdirect.com

Shure SFG-2 Stylus Force Gauge
Esoteric Sound | www.esotericsound.com/access.htm
LP Gear | www.lpgear.com
Music Direct | www.musicdirect.com

Amazon Links

docooler YTA228
<https://www.amazon.com/gp/product/B00NXU106S>

Neoteck NTK002
<https://www.amazon.com/gp/product/B01HRJ9NAY>

Neoteck EMB-02
www.amazon.com/Neoteck-Digital-Turntable-Backlight-Cartridge/dp/B07RGV1Q8W

Riverstone RL100
<https://www.amazon.com/gp/product/B076DFZDS4>

NEXT GENERATION
compression drivers, 1" Exit and 1,7" Edgewound CCA voice coil, with Polyimide Diaphragm. Same no compromise acoustic performance, robust design, high power handling and very linear frequency trend to 20kHz. Cutting edge price-performance ratio comes as standard.



PURE POTENZA

DF10.172K
Ferrite Magnet
120W Program Power
108.5 dB/SPL Sensitivity
1000 - 20000 Hz Frequency Range
114 mm Diameter

DN10.172K
Neodymium Magnet
110W Program Power
110 dB/SPL Sensitivity
1200 - 20000 Hz Frequency Range
85 mm Diameter

YOUR GLOBAL PARTNER
FOR TRANSDUCERS:
FROM CONCEPT TO DELIVERY

Lavoce
I T A L I A N A

WWW.LAVOCESPEAKERS.COM

info@lavocespeakers.com
+39 0733 870 840

Jelco Falls to COVID-19, Aging Equipment, and Engineers

By Gary Galo

In early May 2020 several online audio sites, including the AnalogPlanet forum, reported that Ichikawa Jewel Co. Ltd, the parent company of Jelco, had announced that it was going out of business. In a letter to its distributors, quoted on AnalogPlanet, company President Takako Ichikawa noted:

“The novel corona virus pandemic continues to have a significant impact on our production lines, and so I have decided to terminate our business effective immediately. For the past few years, the business has endured a decline in labor productivity due to health issues arising from our aging skilled engineers. Furthermore, due to our obsolete mechanical equipment, performing maintenance on them has become difficult.”

Ichikawa Jewel Co. was founded in 1920. The company specialized in precision manufacturing of jewel bearings for watches, diamond styli for phono cartridges, jewel bearings for needle pointers in instruments used for tests and measurements, and sapphire bearings for home electric power meters. For many decades, it has been best known in the audio community as a manufacturer of some of the finest tonearms in the world. In addition to its own product lines, Jelco was the OEM source for

tonearms sold by some of the most respected high-end audio companies including Grado, AudioQuest, Linn, Sumiko, and Koetsu. Luxman recently introduced two new turntables sporting Jelco arms, and over the past two years Ortofon has unveiled a new line of tonearms, all manufactured by Jelco, but designed and built to Ortofon’s specifications. In fact, most of the arms sold by these companies were customized variations on Jelco’s basic designs.

Perhaps the most radical modification of the basic Jelco design came from Joseph Grado in the mid-1980s (**Photo 1**). The Grado Signature tonearm took Jelco’s lateral “S” shape and added a downward slope and a high pivot point. Adjustment required patient adjustment of three separate counterweights located at the rear of the tonearm, but the results were worth the effort. Many high-end audio reviewers hailed the Grado Signature as one of the finest pivoted tonearms ever made and, at \$485 US, a steal. I used one for more than 30 years on my custom-built, belt-driven turntable.

Jelco had no regular US distribution network for its stock tonearms, but many dealers imported them directly from the manufacturer, and dealers around the world have been easy to

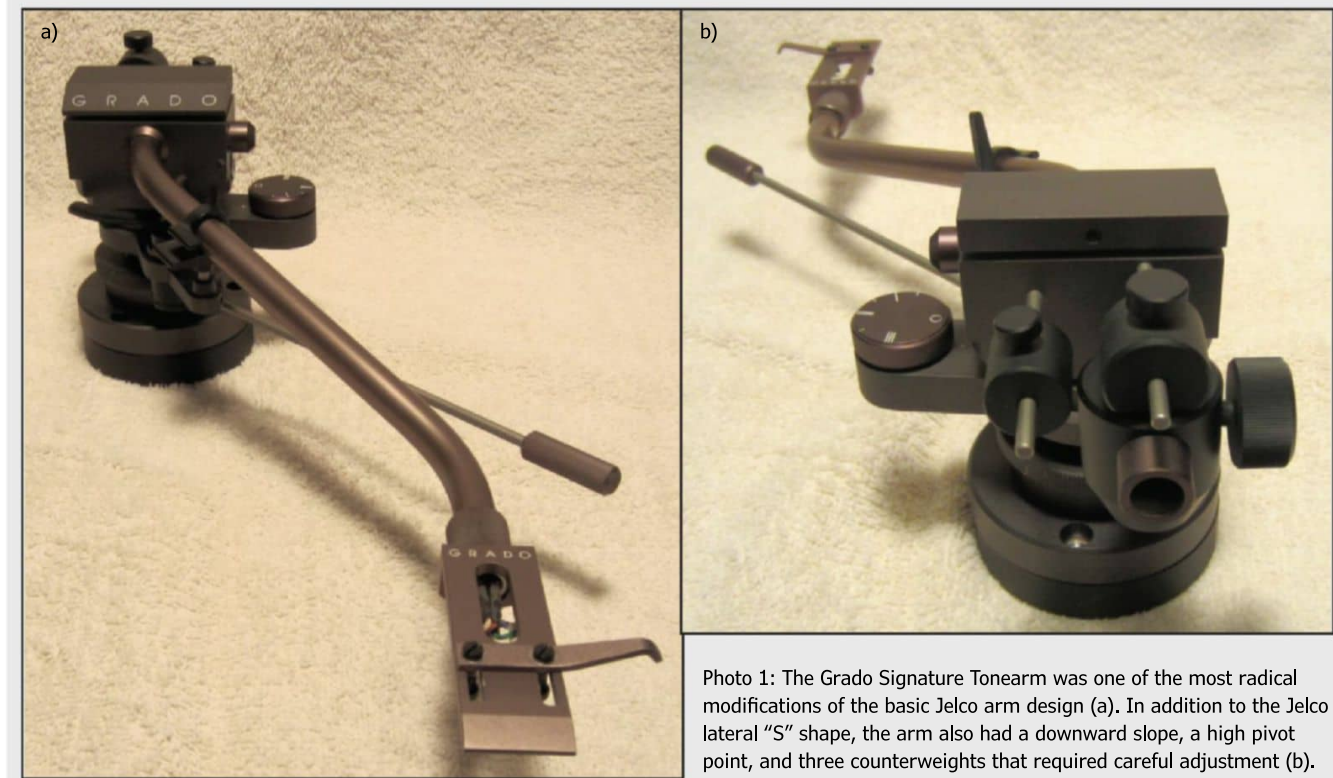


Photo 1: The Grado Signature Tonearm was one of the most radical modifications of the basic Jelco arm design (a). In addition to the Jelco lateral “S” shape, the arm also had a downward slope, a high pivot point, and three counterweights that required careful adjustment (b).

find on eBay. The Jelco arms sold for a lot less than comparable performers sporting major high-end names, and the SA-750, manufactured in three different lengths, became one of its classic designs. Three years ago I decided to replace my Grado Signature with something easier to set up. On George Merrill’s advice, I bought the 9” SA-750D, and have been extremely pleased with its performance (**Photo 2a**). Two years ago I replaced the SME-3012R on my archival turntable with the 12” SA-750L (**Photo 2b**). This arm is long enough to accommodate transcription discs more than 17” in diameter, and performs superbly in this application.

In 2019, Jelco introduced an entirely new lines of tonearms, including the TK-950 and the TK-850 series with knife-edge bearings, and the TK-550 series, which were very similar to

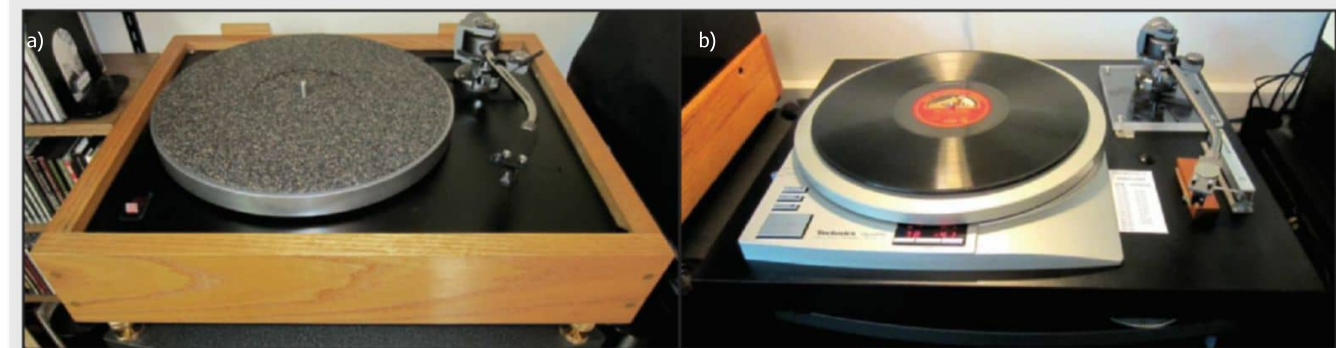


Photo 2: The SA-750 was one of Jelco’s classic tonearms, and was made in three lengths. The author’s custom-built, belt-driven turntable for long-playing vinyl records is fitted with the 9” SA-750D (a). And his archival turntable (b)—a Technics SP-15, custom base, and 12” SA-750L. The 750L will accommodate transcription discs more than 17” in diameter. The nickel glued to the headshell adds sufficient mass to allow tracking forces of up to 7 grams, which is often necessary with shellac 78-rpm records.

CELEBRATING OVER 40 YEARS!
ACOUSTICS BEGINS WITH ACO™



MEASUREMENT MICROPHONES
Polarized & Electret
Windcreens
3 to 18 inch & Custom
Weather/UV Resistant

Community/Industrial
Noise Monitors/Alarms
Indoors and Out





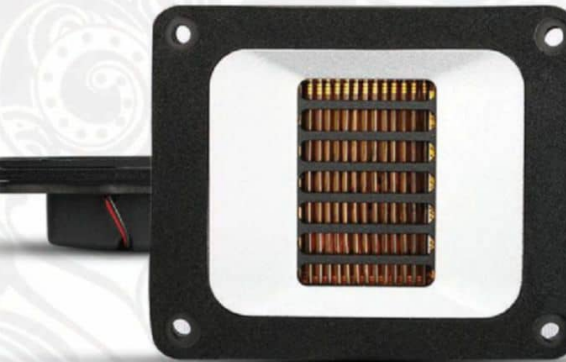
Microphone Systems
Standalone
IEPE Powered
Phantom Power
Custom

ACO Pacific, Inc Tel: +1-650-595-8588
www.acopacific.com sales@acopacific.com
Serving End-Users and OEMs Worldwide



SATORI AT60NC-4
Air Motion Transformer

- Precision diaphragm assembly
- Unique foil damping
- Optimized rear chamber
- Two part faceplate
- Smooth controlled dispersion
- Very flat impedance curve



www.sbacoustics.com info@sbacoustics.com

Vanderveen Trans-SE10 Secrets

By
Menno van der Veen



Recently, a new tube-amplifier DIY kit was brought to market by Vanderveen (design) and Tentlabs (kit manufacturing). The A4-size stereo amplifier is single-ended with 10W output power per channel. “Trans” indicates a method to minimize magnetic distortions in the output transformers and voltage distortions in the power tubes. The technique is based on local feedback with voltage-controlled current sources. The net result is remarkably detailed sound reproduction where the tubes’ second harmonic and output transformer magnetic distortions are minimized. In this article, the “why and how” of the unusual design are explained by its designer.

Through the years, many tube amplifier designs were born in my labs. It appeared to be a long and never-ending search for the real sound. In my youth, I started with standard designs, such as 2xEL84-PP with a switch to select none or strong global feedback. Now, after many years, I present an approach where local feedback is dominant. The scientific opinion is that global and local feedback theoretically are the same. My tube amplifier practice shows that this is certainly not the case. There is for instance a major difference if the output transformer is inside or outside the feedback loop. To briefly summarize my observations on tube amps: Global kills details while local keeps those alive. Therefore, my present design strategy is: per amplifier section, first cure its “faults” by means of local feedback. This article shows how to do that. One extra remark: No feedback at all keeps

details alive, however, it can sound uncontrolled. Local feedback keeps all the details alive and sounds clean and controlled.

A Bit of Valve Amplifier History

The complete audio circuit of the Trans-SE10 amplifier is shown in **Figure 1**. Before I explain it in detail, I want to first delve into valve amplifier history to show the important sources of this design.

In the early days of tube amplifiers, many important research articles were published. Two of them play a major role in the Trans-SE10. The first is from N. Partridge about magnetic distortions in output transformers [1]. The second is from Otto H. Schade about beam power pentodes [2].

Partridge researched magnetic distortions in transformer cores, created by nonlinear behavior (hysteresis) of the magnetic flux inside the core

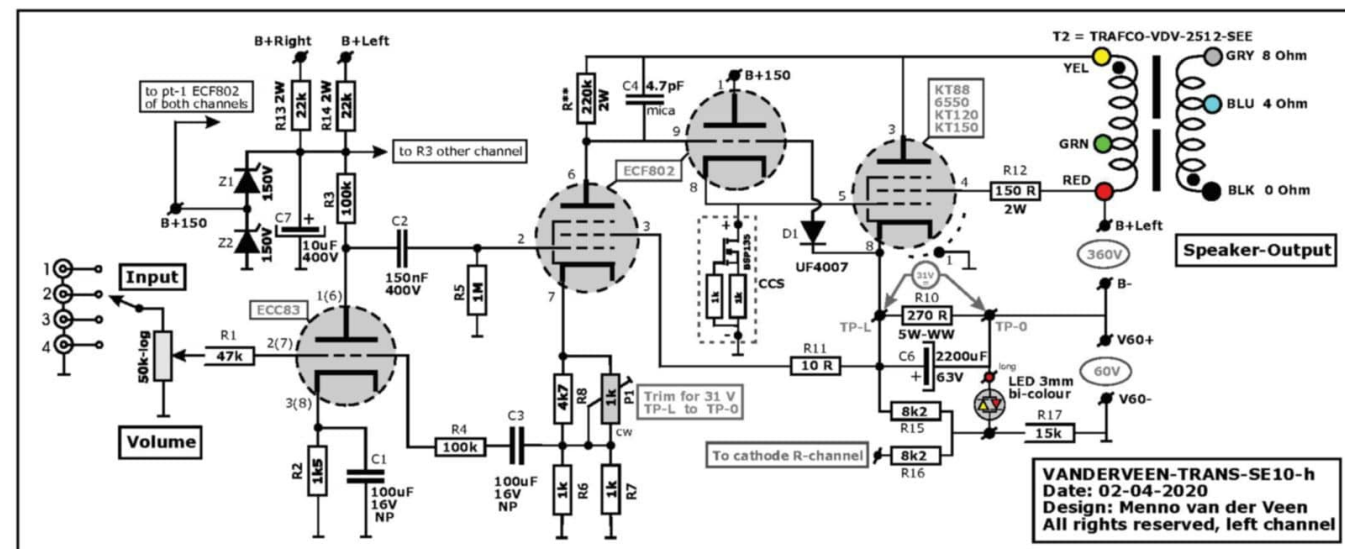


Figure 1: The audio circuit of the Vanderveen Trans-SE10 amplifier

(**Figure 2**). Because amplified audio voltages from the power tube to the loudspeaker pass through an output transformer, the loudspeaker voltage will get distorted as well. He proved that the magnetic voltage distortion at the loudspeaker terminals is largely determined by an external factor: the plate resistance r_p of the driving power tube. The smaller r_p , the smaller the magnetic voltage distortion. This conclusion plays a major role in the SE10.

Schade researched beam-power pentodes. Almost at the end of his publication, he introduced a special local voltage feedback from the anode directly to the control grid of the power tube. See his circuit shown in **Figure 3**. Schade did not apply global feedback from loudspeakers output to amplifiers input (the standard method), but only locally around the power tube. Trained listeners gave a very high score to this method—it sounds much better than global feedback. I tested it and indeed, it sounds very natural.

I made some important observations in my youth at the age of 15 [3]. With the help of an engineer, I built my first EL84 push-pull amplifier. I noticed that the EL84 in pentode, switched to no feedback, was excellent for my electric guitar. However, in triode, the amp sounded much cleaner. Also, with one EL84 removed (push-pull changed to single-ended), I appreciated the sound character as more natural. In all configurations, with global switched on, the sound became flat and lifeless. As youngster, I was unable to explain my observations. Years later, I understood and applied the knowledge in this new SE10 design.

Previously, I mentioned two literature sources and some personal observations. Everyone knows

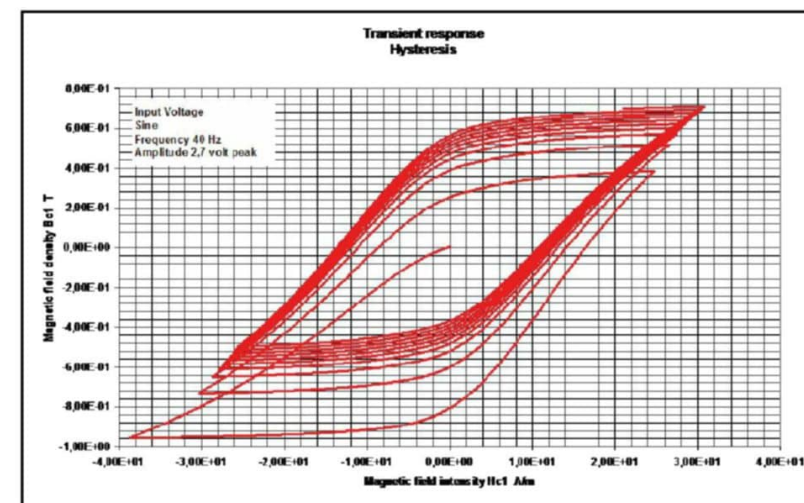


Figure 2: Hysteresis with 40Hz sine wave (Source: Pierre Touzelet)

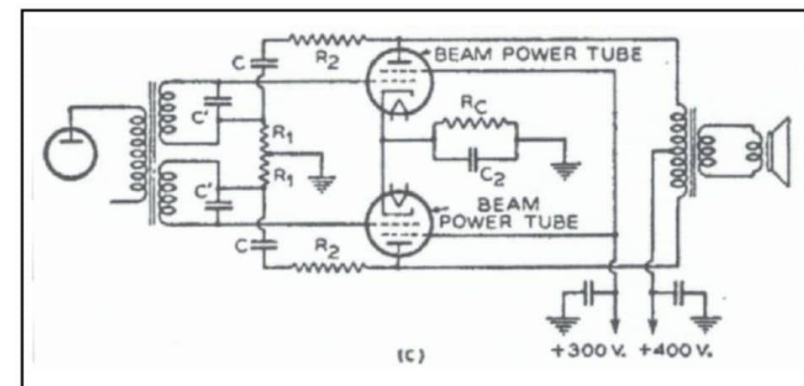


Figure 3: Otto Schade's schematic shows local feedback through R1 and R2.

there are abundance of high-quality articles and observations. But why burden you with a huge literature list? The three mentioned here are enough to explain Trans. So, let's start.

Magnetic Voltage Distortions

I deeply researched toroidal output transformers for valve amplifiers and brought my designs worldwide to market. Toroidals, such as the one shown in **Figure 4**, have the advantage of a large primary inductance, a low leakage inductance, and a small interwinding capacitance. As a result, from



Figure 4: The Trafco-Vanderveen toroidal OPT as applied in the Trans-SE10 amplifier

deep-low to ultra-high, the frequency range is very wide without nasty resonances. In past publications, I explained a lot about the frequency behavior of my transformers (see References). This time, I will focus on the transformer core and its magnetic distortions.

Transformer cores are made of special steels, for example, Grain-Orientated Silicon Steel (GOSS). Inside that steel, we find the magnetic Weiss domains. They change their orientation when the output tube sends its music current through the primary winding. This current creates a magnetic field H , and the core becomes more or less magnetic in the rhythm of this music. The steel has an index that indicates how strong the core supports and amplifies the magnetism because the Weiss domains change their orientation more or less along the field lines. We call this index the relative magnetic permeability μ_r .

In the example about GOSS material, this μ_r has a maximum value of 30,000, which means the core amplifies the magnetism 30,000 times. But it also says that μ_r is not a constant index. It changes when H gets larger or smaller. Consequently, the effective magnetism (flux density B) in the core is not linear with H and also can get saturated and also has a memory ($B = \mu_o \times \mu_r \times H$). This certainly is not linear behavior, and it implies magnetic voltage distortion.

Let's go back to Partridge in 1939. He states: "If the power tube that drives the transformer has a small plate resistance r_p , then the magnetic voltage distortion will be small. But, be aware, when r_p is large, then the magnetic voltage distortion will be large—a linear relation."

What is this r_p ? Let me explain with some examples: A triode such as the famous 300B has a r_p value of 700 Ω . However a pentode such as the EL34 or KT88 has a r_p value environment 10k Ω .

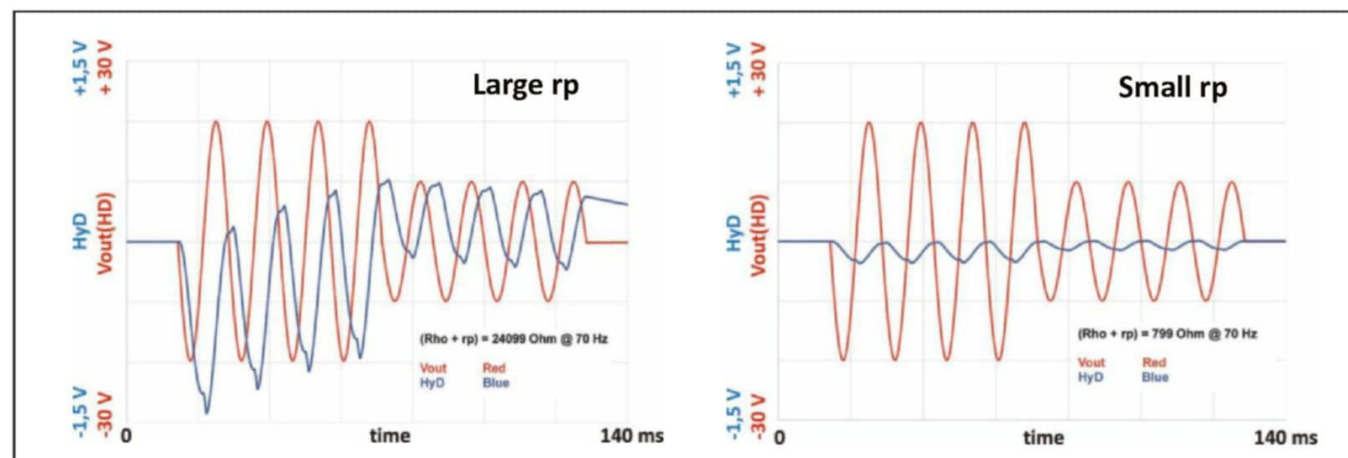
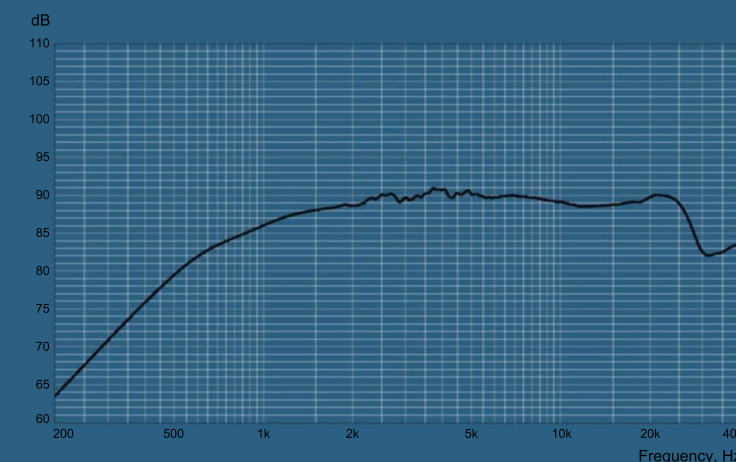


Figure 5: Magnetic distortions HyD (blue lines) for large and small r_p valve amplifiers (Source: Pierre Touzelet and Menno van der Veen)

wavecor

High-End loudspeaker transducers



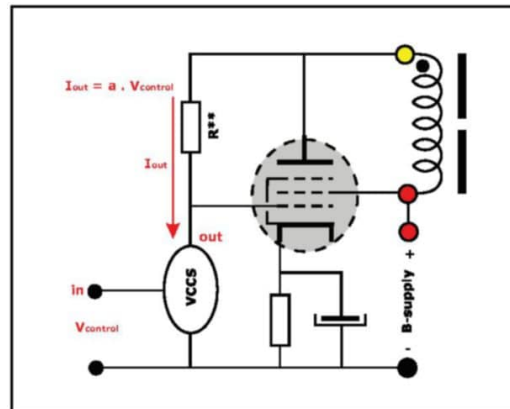
TW022WA
New 22 mm chambered neodymium tweeter, linear 2-25 kHz



WF275BD
New 10" bass driver. Usable for 2-way designs. 27 Hz resonance. 450 W long-term power handling.

www.WAVECOR.com

Figure 6: This the Trans-Tube principle.



Meaning: if we make an amplifier with a 300B and this amp distorts magnetically 0.5%, an equal EL34 amp will have $0.5\% \times 10000/700 = 7.2\%$ magnetic voltage distortion. This is impressive more than with the 300B.

Figure 5 shows some examples of magnetic voltage distortion (indicated as HyD, meaning Hysteresis Distortion). Small magnetic distortion is one of the reasons why 300B triode amplifiers sound so clean compared to dull sounding EL34 pentode amplifiers. (The change in medium 300B damping factor to very small EL34 damping factor is another important factor, not discussed now.)

To be more complete in my explanation, I need to make two remarks: These results are valid in the low-frequency range. Second, the tube itself (300B or EL34) distorts the voltages as well, but this is not magnetic distortion. I will deal with tube distortion when discussing the Trans principle.

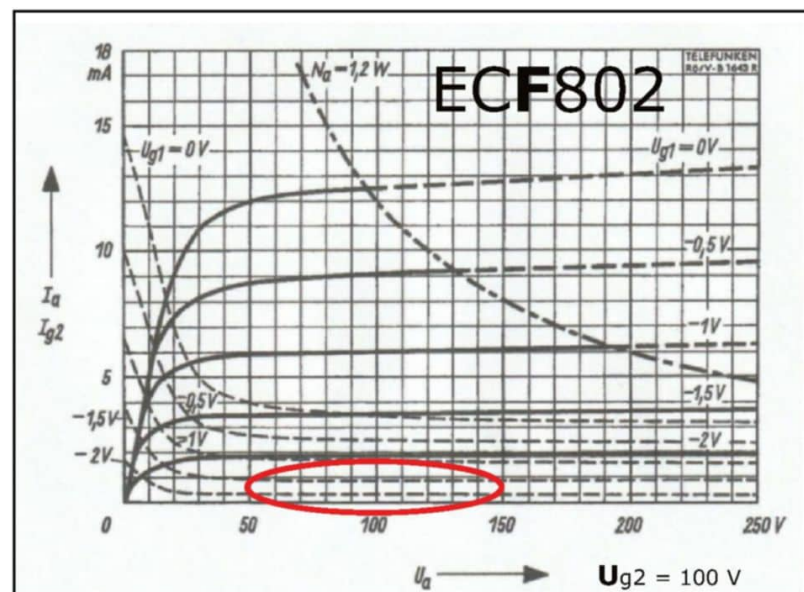


Figure 7: ECF802 characteristics with linear I_{g2} section

Summarizing: if you wish the lowest magnetic voltage distortions in your output transformer, apply low rp power tubes. Which tubes do we find in the world's best high-end amplifiers? The 300B and many other fantastic directly heated triodes, such as the 211 in the world-famous Ongaku amplifier. Top designers know this Partridge rule and have it chiseled in stone above their test bench.

[For the diehards: Make an air-gap in the core and the magnetic voltage distortions are gone. Careful: Partridge proved that per frequency the ratio between rp and Lp determines the distortion reduction. Gapping makes Lp smaller and consequently the effective magnetic voltage distortion is not that simple to calculate.]

The Trans Principle

There are several methods to create low plate impedance power tubes. I will list them and focus next on Trans.

Apply low rp triodes, such as the 300B, the 6AS7, or the 6C33 (voltage supply regulator tubes are excellent in low rp aspect) or the larger 211 (VT-4C) and the like. Or apply local feedback at the tube's cathode (Audio Research and Quad II). Or apply the Ultra Linear principle of local feedback at the screen grid (Hafler and Kerous). Or combine cathode and UL feedback (unity coupled: McIntosh). Or follow the Shade approach shown in Figure 3. Or use Menno's Super-Triode approach. Or apply feedback from the anode of the power tube to the cathode of the driver tube. And so on!

The tube amplifier world is filled with the finest inventions. All these methods have in common that the local feedback is partly applied, not 100%, although McIntosh is extremely close. Trans creates 100% local feedback, as is the cathode follower approach [4], as are the MTA inventions [5],[6]. I know there are more fantastic solutions, but I can't discuss all of them. For more information, see my The-Project study [7], where I evaluate most of these methods in detail. Next, I will focus on Trans only.

Suppose I have a pentode power tube with a plate resistance $r_p = 10 \text{ k}\Omega$, an open-loop amplification factor of 200, and an internal tube voltage distortion of 5%. Suppose I use all its amplification to correct its own mistakes (a correction of 200 equals 46dB local feedback). What will be the specs of this new Trans-Tube? Its new plate resistance will be $10000/200 = 50\Omega$. The new amplification factor will be $200/200 = 1$ and the new tube voltage distortion will be $5/200 = 0.025\%$. These new specs are better than any available tube on the market. So let's do it (**Figure 6**).

Added to the pentode is a voltage-controlled current source (VCCS), which simply means: a device that delivers a current output. The currents magnitude is only determined by the controlling input voltage. At its current output, the VCCS has an infinitive internal impedance. The VCCS output current only runs through a special R^{**} resistor, which is directly connected between the anode and the control grid of the pentode. How does this function?

First, seen from the power tube: All voltage variations at the anode appear 100% at the control grid because the internal impedance of the VCCS is infinitive. There is no loss inside the VCCS. The control grid even does not notice the VCCS. The control grid only sees the R^{**} connection. This is equal to 100% feedback, and the nice specs mentioned are the result.

Second, the variable music current I_{out} runs through R^{**} and creates a changing voltage over this resistor, which drives the control grid of the power tube that next drives its anode.

Third, seen from both sides: a coupling capacitor between the control grid of the power tube and R^{**} is not allowed. Such a capacitor would combine its phase shift with the phase shift created by the primary inductance Lp of the output transformer. This would result in more than 90° phase shift in total, causing ugly motorboating (low-frequency oscillation). For proper operation down to the lowest frequencies, capacitor-less direct-coupling is a must. This condition is difficult in practice, but advantageous because the sound of a coupling capacitor will be absent [8].

See additional article references [9], [10], [11] for more and deeper explanations or other approaches. For now, this short introduction is enough to proceed to the next stage.

VCCS Realization with ECF802

After many experiments I decided to make an all-tube VCCS. With good reasons I selected the ECF802, because its pentode characteristics has a nice linear part, indicated in red in **Figure 7**.

In this linear part the screen grid current I_{g2} of the F-section is constant, which is a mandatory condition. In the next section about linearization, I will improve the linear relation between the control voltage at the control grid and the current source output I_a at the anode of the ECF802 (**Figure 8**). In that circuit I measure the cathode current I_k to know I_a . However, the cathode current I_k contains $(I_a + I_{g2})$. Meaning: when I_{g2} is constant, the alternating (music) part of I_k will equal the alternating (music) part of I_a . See the next section for further details.

I placed a cathode follower between the R^{**} -VCCS connection and the control grid-1 of the power tube. The reason why is the Miller capacitance between grid-1 and the anode of the power tube. In power pentode tubes the screen grid shields the anode from grid-1, but still there is some residual Miller capacitance. Consider this Miller capacitor in parallel with R^{**} and you will understand that the high-frequency bandwidth gets restricted. The cathode follower shields R^{**} from the Miller capacitance and prevents this bandwidth restriction.

Another reason for the cathode follower: grid-1 of the power tube might draw some grid-current (although little, but still) that could create extra harmonic voltage distortion. The cathode follower shields R^{**} from any control grid current. So, it is there for very good reasons. Next, a purist might argue: Why a field-effect transistor (FET) current source at the cathode of the follower instead of a normal cathode resistor? Just to make the distortion of the cathode follower lowest, but you

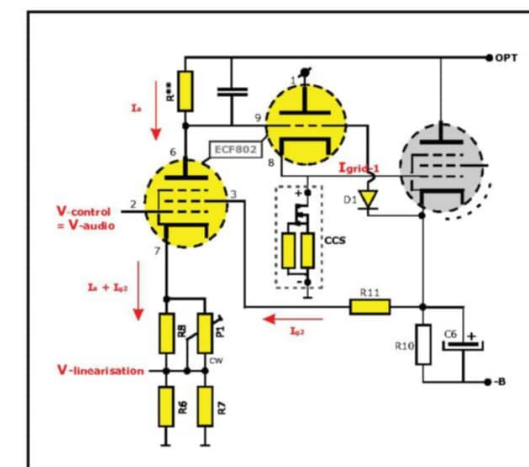


Figure 8: The complete ECF802 current source section

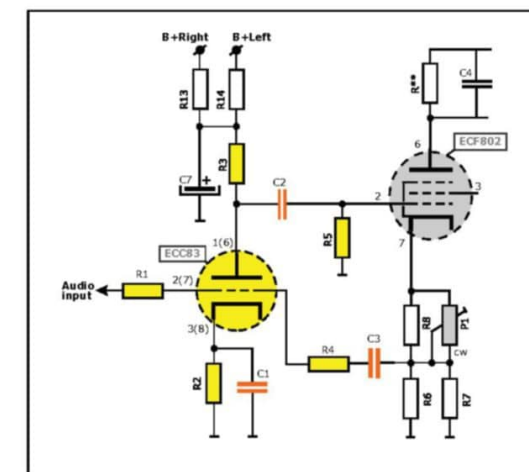


Figure 9: Linearization circuit of the current source

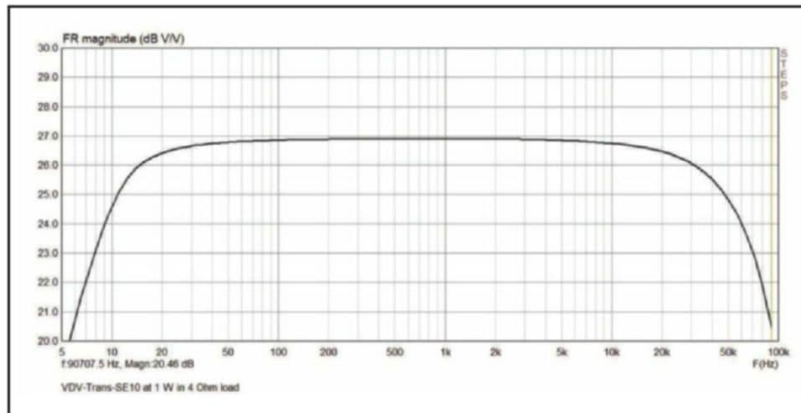


Figure 10: Trans-SE10's frequency characteristic at 1W in 4Ω load

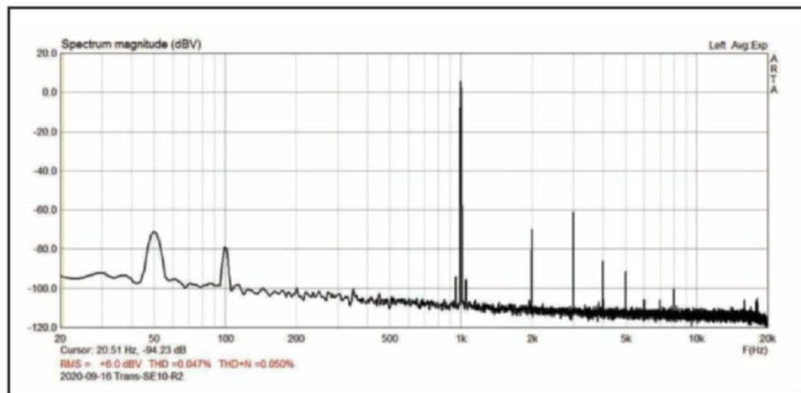


Figure 11: Trans-SE10's harmonic distortion at 1W in 4Ω load

About the Author

Menno van der Veen studied technical physics and electronics at the "Rijks Universiteit Groningen" in the Netherlands. From 1976 to 1990 he taught physics at a high school and a teachers training College. Weekends he worked as sound engineer in a small alternative theater and was sound coordinator for big outdoor events. As reviewer, he wrote many articles for Dutch high-end audio magazines, evaluating audio amplifiers. In 1986, he founded his "Vanderveen" company with focus on valve amplifiers with toroidal output transformers. In cooperation with manufacturers worldwide he designed his new range of Vanderveen-toroidals and regularly reported about research results in Audio Engineering Society (AES) convention papers. Menno was chairman of the Dutch AES section and co-organizer of AES conventions in The Netherlands. In 2005, he founded a bi-weekly Saturday school, called "TubeSociety," with the goal to share his specific tube amplifier knowledge with enthusiast do-it-yourselfers. In his private hobby time, Menno devotes a lot of research on electronic guitars, their valve amplifiers and effect pedals, almost never publishing research results. He says, "When making music yourself, you train your ears in the search for the sublime guitar sound." These skills, combined with electronics knowledge, are the powerful tools he applies when designing new audio equipment.



are absolutely free to replace the CCS by a 27 kΩ 0.5W resistor.

Two additional remarks: The diode D1 between R** and the cathode of the power tube prevents positive grid-1 voltages with respect to the cathode at start-up of the amplifier. Second, the cathode current of the power tube creates an almost constant voltage over the R10-C6 combination. The more quiescent current, the larger this voltage. The screen grid of the F-section of the ECF802 senses this voltage with R11 and creates a quiescent current feedback, stabilizing the operation point of the power tube.

VCCS Linearization with ECC83

In spite of the good ECF802, the linearity of the current source was not good enough; some residual distortion was noticed. To improve linearity, I created an independent second local feedback circuit with an ECC83 (one triode section per channel). This new circuit (**Figure 9**) has two inputs: the audio signal input via R1 and the VCCS current sensing input with R4. (For now, forget the coupling capacitors C1, 2, 3. I will deal with those in a moment). The large amplification of the triode section (factor 60) is reduced to $R4/R1 = 2$. So, a factor of 30 is available to make the current of the VCCS "absolutely equal" to the audio input voltage. Gone were the residual VCCS distortions. This is a local feedback of $30x = 30dB$, highly linearizing the VCCS. Imagine such 30dB plus the earlier mentioned 46dB (76dB in total) as a global feedback from amplifiers output to its input in an old fashioned tube-design. Absolutely impossible! Total instability would be the nasty result. Only with local feedback sections can you do such a trick.

In this extra ECC83 local feedback section, only AC voltages (music) play a role. So, coupling capacitors are allowed, making life (design) much easier. But capacitors do sound [8]. Yes, but their sound is weakened by 30dB feedback, and not noticeable anymore. Well, let's apply cheap ones. Please not; I applied good ones. Recently, one of my TubeSociety students used a very expensive C2 and heard further improvement. This underlines my statement to use good quality capacitors. (It also says: Feedback is not completely able to suppress capacitor distortion. But that is another research topic.)

Also, the local feedback ECC83 circuit should operate perfect from almost DC level to get the optimal VCCS linearity. That explains the rather large capacitor values. Don't change these, or else the VCCS and power-tube and OPT cooperation will not be optimal anymore at the lowest audio frequencies.

The "Must Have" reference for
loudspeaker engineering professionals.

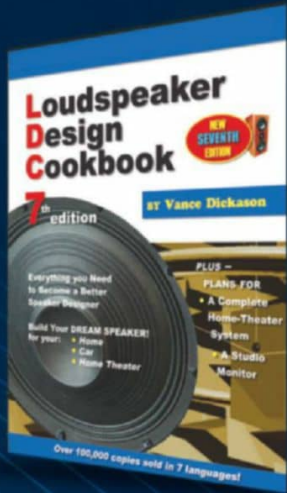
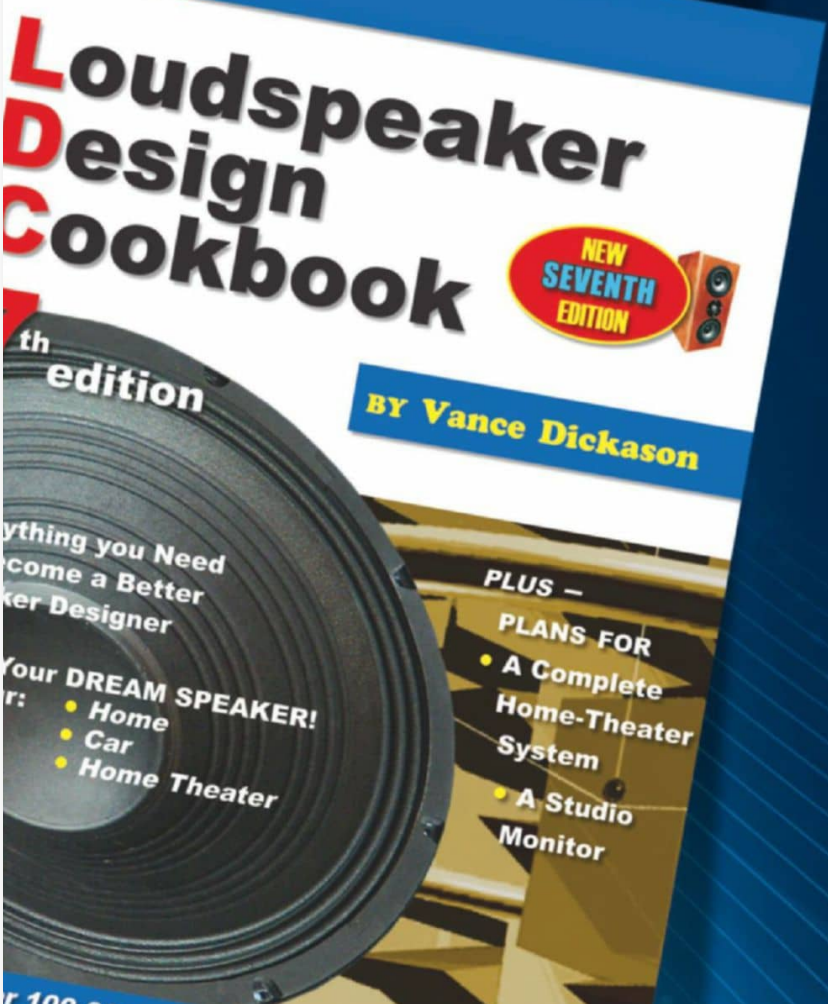
Home, Car, or Home Theater!

Back and better than ever, this 7th edition provides everything you need to become a better speaker designer. If you still have a 3rd, 4th, 5th or even the 6th edition of the *Loudspeaker Design Cookbook*, you are missing out on a tremendous amount of new and important information!

Now including: Klippel analysis of drivers, a chapter on loudspeaker voicing, advice on testing and crossover changes, and so much more! Ships complete with bonus CD containing over 100 additional figures and a full set of loudspeaker design tools.

A \$99 value!
Yours today for just \$39.95.

Shop for this book, and many other audio products, at
www.cc-webshop.com.



(For the diehards: C1,2 are interrelated to create a first-order high-pass behavior of the ECC83 section; see chapter 4.4 in my book High-End Valve Amplifiers [12]. Capacitor C3 creates an extra high pass filter, but its corner frequency is way below the C1,2 corner frequency, and therefore not disturbing.)

Measurements and Explanations

In this article, I will not pay attention to the case and construction of the Trans-SE10 amplifier. Please download the construction manual [13]. I also will

References

- [1] N. Partridge, "Distortions in Transformer Cores," *The Wireless World*, June 1939.
- [2] O. H. Schade, "Beam Power Tubes," RCA Radiotron Division, RCA Manufacturing Co., Inc.
- [3] M. van der Veen: personal observations in 1964
- [4] P. Touzelet, "The Trans-driven Cathode-Follower Amplifier," work in progress
- [5] F. Blöbbaum, "Multiplied Transconductance Amplifiers, Part 1," *Linear Audio*, Vol. 6, pp.83-91.
- [6] F. Blöbbaum, "Multiplied Transconductance Amplifiers, Part 2," *Linear Audio*, Vol. 8; pp.31-50,
- [7] M. Van der Veen, www.menno.vanderveen.nl > research & development > publications > amplifiers > 2005: The Project
- [8] The works of Ari Polisois, who refused to use coupling capacitors, with impressive results.
- [9] J. Macaulay: "Hot audio power"; *EW + WW*; October 1995, pp.856-859.
- [10] M. van der Veen, *Vanderveen Trans Tube Amplifiers*, Elektor, ISBN 978-1907-920349.
- [11] J. Broskie: <http://www.tubecad.com/2014/09/blog0305.htm>
- [12] M. van der Veen, *High-End Valve Amplifiers 2*, Elektor, ISBN 978-0-905705-90-4
- [13] M. van der Veen, "Vanderveen-Trans-SE10 Construction Manual," www.menno.vanderveen.nl
- [14] M. van der Veen, "TubeSociety 2A3-300B-SE Amplifiers, Part 1," *audioXpress*, April 2018
- [15] M. van der Veen, "TubeSociety 2A3-300B-SE Amplifiers, Part 2," *audioXpress*, May 2018.
- [16] M. van der Veen, "Build an 18 W Single-Ended Valve Amplifier with Trans Technology," *audioXpress*, December 2019.
- [17] K. Howard, "Euphonic Distortions" *Stereophile*, www.stereophile.com/reference/406howard/index.html
- [18] P. Millet, "The Sound of Distortion," http://www.pmillett.com/etf_sod.htm

not discuss the power supply section. Equivalent circuits were applied in earlier publications [14],[15],[16].

The frequency characteristic shown in **Figure 10** shows excellent behavior from 9Hz to about 60kHz (-3dB frequencies). No ringing or oscillation is noticed, thanks to the applied toroidals. This all looks great and needs no further remarks, only that at the onset of clipping, the amp is capable of 12W output power. On purpose, the SE10 number is a modest number.

The next measurement (**Figure 11**) shows the 1kHz harmonic distortion spectrum. The second harmonic is equal or lower than the third, depending on the ECF802 tube brand. That is unusual in single ended tube amps. The power tube has such a small Trans-rp (which reduces magnetic OPT distortions) and such small internal tube distortions (due to the local Trans-feedback), that the standard declining harmonics spectrum is not observed anymore. I discussed this effect earlier [14],[15],[16].

When you study my article [14] in detail, I describe my observation that standard declining harmonics is pleasant, but also has a curtain (the H2-curtain; from Keith Howard's article "Euphoric Distortions" [17] calls it "muddy") in front of the soundstage, making finest details not noticeable anymore.

Only with amplifier 4 [14] and amplifier 5 [15], was I able to remove that curtain. I demonstrated this effect to a group of experts at the European Triode Festival (ETF) 2019 in France. They agreed with my observation and referred to studies where similar results are discussed [17], [18].

I had very good reasons to discuss it all at ETF. The general opinion is that declining harmonics gives "The Great SE-Sound." Now I conclude: "declining harmonics hides details; H2 has to be minimal." This is like cursing in the church. Think of the great work of Jean Hiraga, who taught us about the marvelous Japanese single-ended amplifiers, where the declining harmonics rule governs. With Trans, I clearly deviate from this generally accepted rule. I invite you to listen to Trans and judge for yourself.

The next reason to discuss it all at ETF is that magnetic distortions are largely suppressed. Normally, this is partly done by means of global feedback. But everybody knows that global feedback exceeding 20dB surely will create instability due to resonances in OPTs. Remember, Trans creates about 46dB of feedback locally in front of the OPT. Therefore, Trans suppresses magnetic distortion much more. So, one could wonder: is it the Trans H2-reduction or the Trans magnetic distortion reduction that creates the clearer details, or is it

both? I think both. My research in articles [14],[15] showed that H2-reduction removes the curtain. However, amplifier 4 (with Trans) was better than amplifier 5 (with H2 predistortion compensation).

Youth Observations Explained

Let's go back to my observations at the age of 15. I noticed in my first EL84-PP amplifier with global feedback switched off, that EL84's in triode mode sounded cleaner than EL84s in pentode. This is logical: triodes have lower rp. So, the magnetic distortion of the OPT was smaller. Also, triode mode means internal feedback inside the EL84s. So, the tubes operated cleaner. This observation is fully explainable with our present insight. But there is more: push-pull minimizes H2. So, my first EL84-Triode push-pull amplifier was going direction Trans. At that age, I was totally unaware of this.

What is there to say about my guitar preference for pentodes? I liked the brown bluesy distortions and that preference has not changed over the years.

Let's look at my other observation with one EL84 removed from the Push-Pull (PP) amp. In fact, I listened to a non-ideal single-ended (SE) amp with standard declining harmonics and I liked that sound.

How to explain the *unalive* sound with global feedback switched on? Let's say it in normal language: "The feedback was unable to correct all the faults in my starters amplifier and made a mess of it."

Is That All There Is?

I noticed in the reactions of my students and SE10 listeners that they did not recognize the two local feedback sections, nor the fundamental reasons why I created them. I hope this article will support their understanding. How I calculated component values and so on? That will be a totally different story, way beyond my goal to explain its fundamentals. For that reason I will stop now.

Allow me to add one remark overruling my research on tube amps. For years, I followed the global feedback trail. Next I changed to no feedback at all. Now I research local feedback and it feels like having gold in my hands. Its essence is: "Cure before you apply." That's it.

Summary

The new Vanderveen Trans-SE10 is a challenging DIY kit. It clearly demonstrates what benefits can be gained with local feedback in relation to detail reproduction. Its concepts are deeply rooted in tube-amp history. The Trans technology lowers tube and output transformer distortions substantially, allowing to enter new areas of valve amp sound reproduction.

What future progress can we expect? Presently, I am researching Trans with my TubeSociety students in high power (up to 100W) push-pull Class-AB tube amplifiers. When ready, I plan to share it in one way or another. Please be very patient! Research dictates its own pace (and the coronavirus certainly does not help).



NCAS500MP

400 + 100W in 8 ohm

Compatible with our MP-DSP to create the perfect setup for your active speaker

New family Ncore modules

NCAS500MP is the first to be released

Future members:

NCAS250MP: 200 + 50W
NCAS1000MP: 800 + 200W

For further information:

website: www.hypex.nl

mail: sales@hypex.nl



© Hypex Electronics B.V. All Rights Reserved

The Live-Edge Dipoles

By
Perry Marshall

Perry Marshall has built almost every kind of speaker you can imagine. One day during a visit to Big Red Sawmill in Palmyra, NE, he had the desire to build these Live-Edge Dipoles. He bought two slabs of birch and brought them home to Chicago, IL. This system achieves Constant Directivity across the entire audio spectrum.

I have built almost every kind of speaker you can imagine, and though sealed and ported speakers are the most “textbook correct” and straightforward to design... horns, transmission lines, and dipoles are a lot more musical and more fun to listen to. When you really get them right, the sound is incredible. When my friends visit, most have never heard anything like these live-edge dipoles. When you play classic Miles Davis recordings, they have a huge, lifelike, warm, present sound. If you play Rush or Porcupine Tree, they handle the dynamics and bass with ease.

Performance Summary

- Achieves 25Hz to 25kHz with silky-smooth response in a real room that has real reflections. Not just in an anechoic chamber.
- True Constant Directivity sound pattern 25Hz to 25kHz so imaging is superb anywhere in the room
- Open-Baffle Dipole sounds great even behind the speakers
- High Efficiency—95dB 1W/1m on average. Sounds great and plays loud even with “flea watt” single-ended vacuum tube amps
- High Power Handling and High Output—100W and 115dB full range, 1000W and 125dB with subwoofer
- Near-perfect impulse response
- Near-perfect phase response
- <1% Distortion

E.J. Jordan once said, “Art is science with more than seven variables” and this is a great truth. One of the realities that designers of transmission lines, horns, and dipoles accept is that as soon as you put your speaker in a real room, your dreams of perfection go out the window.

The Radiation Pattern

In speaker design, the elephant in the room is the room. Radiation pattern is one of the most important and overlooked aspects of a speaker. A good radiation pattern makes a huge contribution to how realistic the sound is. A speaker with great response on axis and bad response off axis will inevitably sound unnatural. Room reflections are all telling your ears “this is fake sound.” With a good dipole, you frequently find yourself marveling at how realistic the illusion of live sound is.

Constant Directivity is what you really want. Constant Directivity means you have flat frequency response on axis, and as you turn the speaker in any direction, the level steadily drops off axis, but the octave to octave balance remains flat. Then imaging will be great anywhere in the room.

The way to achieve this is with a Constant Directivity horn. Horns are harder to work with—they usually don’t have perfectly flat on-axis

response, and you have resonances and complex impedance loads to deal with. However, CD horns are mathematically optimized toward this ideal radiation pattern, and the good ones achieve it.

A constant directivity woofer is impossible—unless you build a dipole. A dipole gives a “figure 8” radiation pattern where the front and back give strong on-axis response, and the backwave progressively cancels the front wave as you move off axis (**Figure 1**).

So... if you design a dipole using a coaxial woofer where the tweeter is a Constant Directivity horn—and if you add a CD horn tweeter on the back to make the back and front symmetrical—you get a true Constant Directivity response both front and back. The dipole configuration automatically makes the woofer into a Constant Directivity source.

I found that adding a rear tweeter is a significant enhancement because then the dipole radiation extends to both extremes (**Photo 2**). You design the speaker for radiation pattern, dipole Constant Directivity pattern across the entire range... then you fix frequency response problems with digital signal processing (DSP). This gives you an amazing speaker that gives you imaging and sound staging that are physically impossible for cones and domes to achieve.

With the Live-Edge Dipoles, you can walk all over the room—even stand on the far side of the left or right speaker, and you still clearly hear the other speaker across the room and the imaging is still stable. There is no one “sweet spot.” Music sounds fantastic anywhere you go in the room. You can walk around and behind the speakers and they still sound great! This, in addition to careful attention to phase and impulse response makes for fantastic imaging.

I took a group of response curves (**Figure 2**), with great results. I’ve seen a few speakers that achieve this above 500 or 1000Hz; I’ve never seen any other speakers that achieve this from 25Hz to 20kHz.

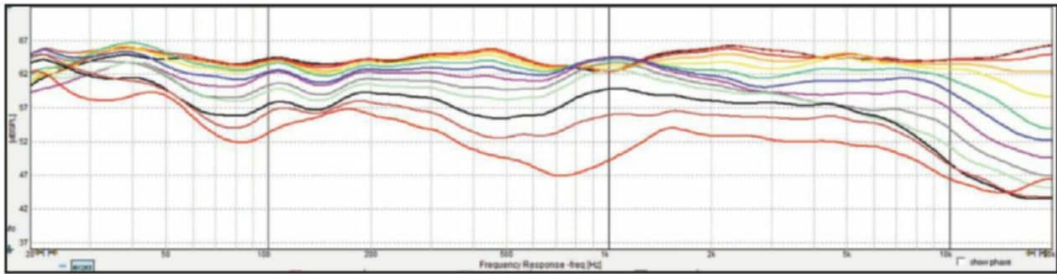


Figure 2: Off-axis measurements at 0, 7.5, 15, 22.5, 30, 37.5, 45, 52.5, 60, 67.5, 75, 82.5 and 90 degrees, 1m from the Radian 8” coaxial. This shows the speaker is true Constant Directivity from bottom to top. This is only possible with an open baffle. The polar pattern is somewhat wider around 1kHz because of the wide baffle (~18”). A baffle that tapers down to ~10” near the 8” driver would yield a more consistent radiation pattern but also force a higher crossover frequency.

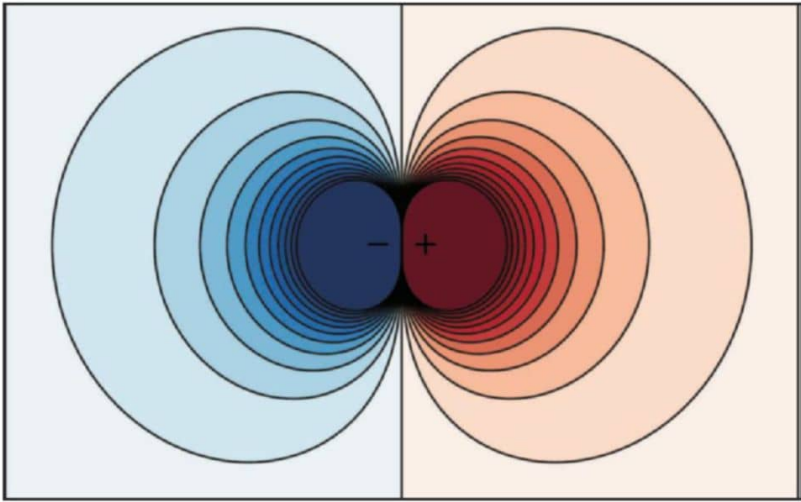


Figure 1: On the side, which is top and bottom in this diagram, the front and back waves cancel out.

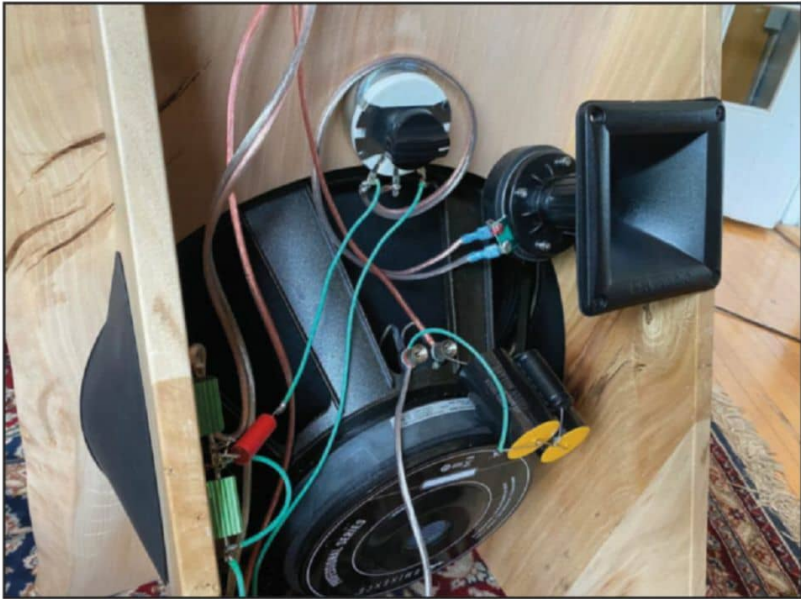
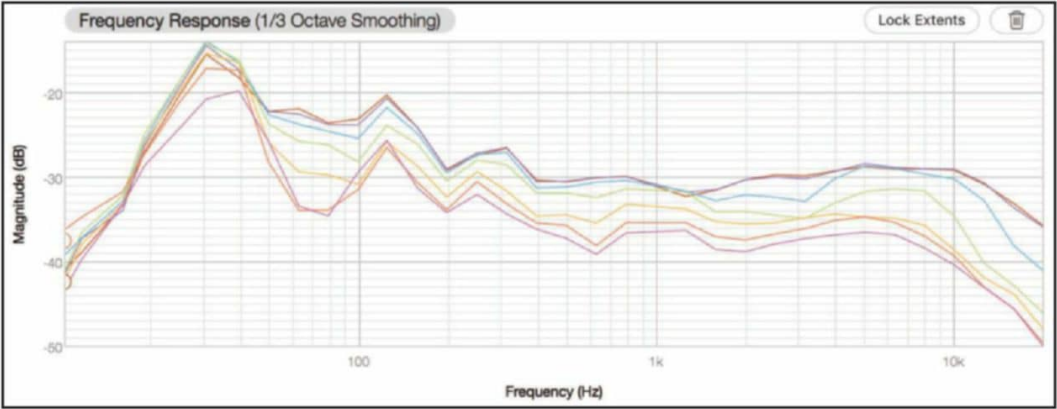


Photo 2: I found that adding a rear tweeter is a significant enhancement because then the dipole radiation extends to both extremes.

Figure 3: This graph shows 1/3 octave frequency response curves in a real listening room at 0, 15, 30, 45, 60, 75 and 90 degrees.



Design Priorities

Accurate frequency response is a major objective, but I have other priorities as well:

- Lifelike presentation of acoustic instruments like string bass
- Extremely high resolution of drums, percussion, and plucked instruments
- A huge soundstage
- High efficiency
- High output
- Very deep bass
- Flat phase response, accurate impulse and step response
- Separate amplifiers for low and high frequencies

In a real room, flat frequency response in the lab translates to the low bass being about 10dB

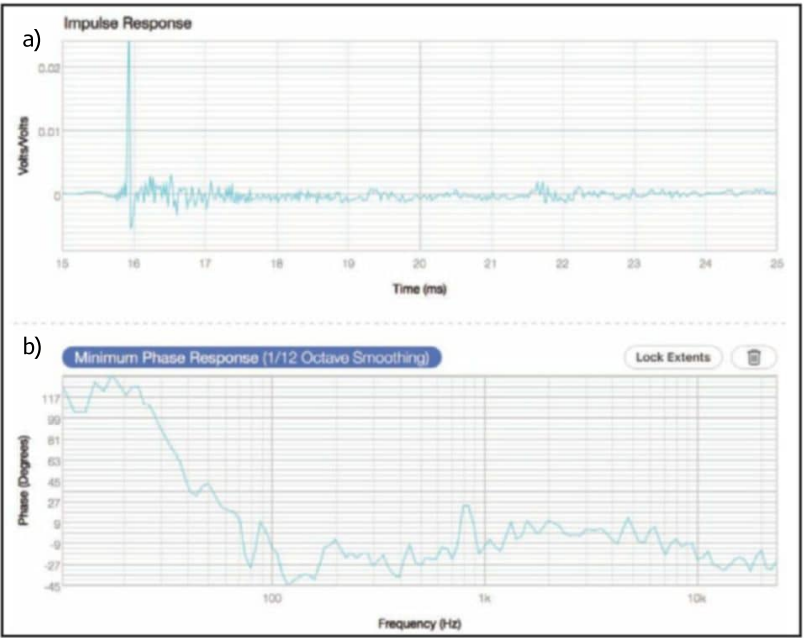


Figure 4: Impulse response for the finished system (a) and Phase response (b)

stronger than the highs. **Figure 3** shows a 1/3 octave average of seven off-axis angles.

In order for the speakers to be highly efficient, I chose the Eminence Kappa Pro 18LF woofer, an 18" pro bass driver with 98dB sensitivity for 1W, ± 8 mm excursion (Xmax). Mounted in my dipole configuration, the resonance is 27Hz with a Qt of 0.5.

Traditionally, dipole designers prefer high-Q woofers. The dipole starts rolling off below 70Hz and drops 6dB per octave. Since none of the available high Q woofers had 8mm of excursion, I used DSP to EQ the bass.

I also chose the 8" Radian 5208 coaxial mid bass + Constant Directivity horn tweeter (16 Ω tweeter with aluminum diaphragm). The 8" Radians have



Photo 3: For speakers, I chose the Eminence Kappa Pro 18LF woofer and the 8" Radian 5208 coaxial mid bass + Constant Directivity horn tweeter.

95dB sensitivity in the midrange and the tweeters deliver 105dB (**Photo 3**).

The Eminence 18" woofer moves a ton of air. Bill Duddleston of Legacy Audio says: "If you can see the cone moving, it's distorting." There is something entirely different about a speaker that has "dynamic range to burn." The music is effortless, powerful and unrestricted at all volume levels.

Crossover Details

I could have triamped these, but biamping dramatically reduced complexity because it enabled me to use a two-way MiniDSP 2x4HD digital crossover. The bass amp drives both woofer and midrange, and the treble amp drives the tweeters. Crossover frequencies are 200Hz and 2kHz. DSP aligns the impulse timing of the drivers to within 0.01ms.

This required a very simple series first-order crossover between the woofer and midrange. A series crossover yields fewer impedance problems than a parallel crossover. This is effectively a 6dB/octave crossover. However, in reality the slope is closer to 12dB. This has too much phase shift, so I wired the midrange in reverse polarity, then compensated for the phase shift using a Finite Impulse Response DSP filter (Eclipse Audio software).

I also used Finite Impulse Response DSP correction on the tweeter. This solves phase shift problems and gives you an incredibly clean impulse response. **Figure 4** shows the impulse response for the entire system as a whole. The impulse response should look like an upside-down T, and it does, with only a touch of overhang. And as you can see, the phase response is ± 45 degrees from 40Hz to 25kHz with no polarity reversals or phase rotations. 99.5% of speakers butcher impulses. 99.5% of speaker designers ignore phase response and only give a shrug to impulse response.

It has long been asserted that you cannot hear phase anomalies, so it doesn't matter. But, I've never understood that argument. Most audiophiles have arguments about stuff you can hear but can't measure... But why would you completely ignore a problem you can easily and obviously measure?

Answer: Because phase shift is actually very hard to solve without DSP. So designers just wimp out on it and say it doesn't matter. With DSP it is solvable, though. And companies like Thiel, Vandersteen and Duntech built product lines around linear phase even with passive crossovers and earned well-deserved respect for doing so. If your speakers have good impulse response, they will also image well.

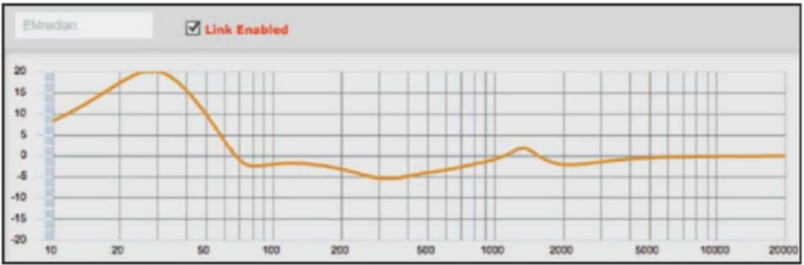


Figure 5: This graph shows the digital EQ curve that sounds best with these woofers.

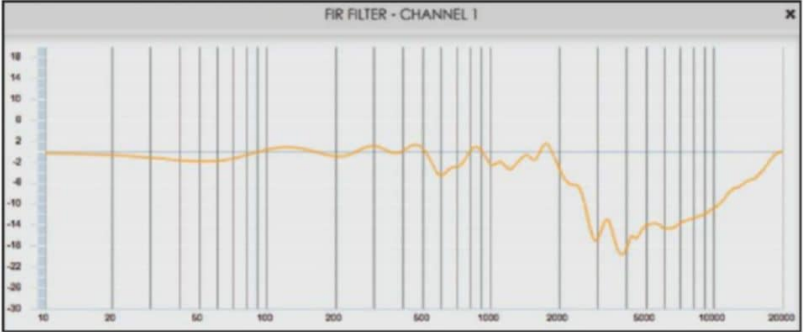


Figure 6: The crossover function for the midrange is performed by this FIR DSP filter.

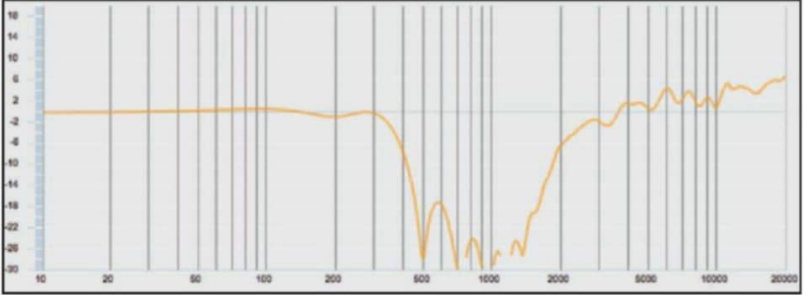


Figure 7: The FIR filter addresses the overlap with the midrange at 2kHz.

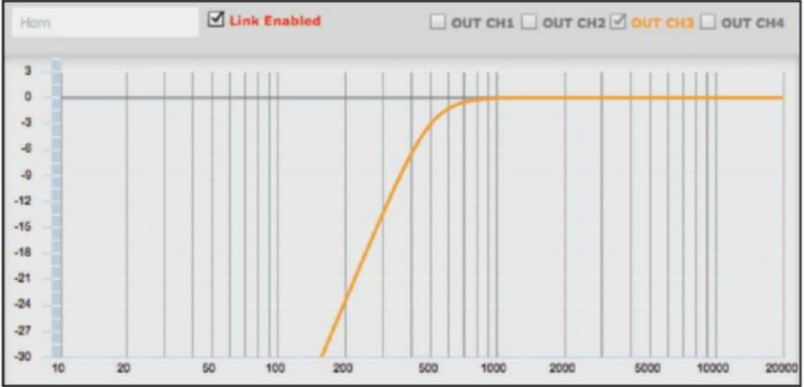


Figure 8: An 18dB/octave IIR filter is used to finish what the FIR filter couldn't.

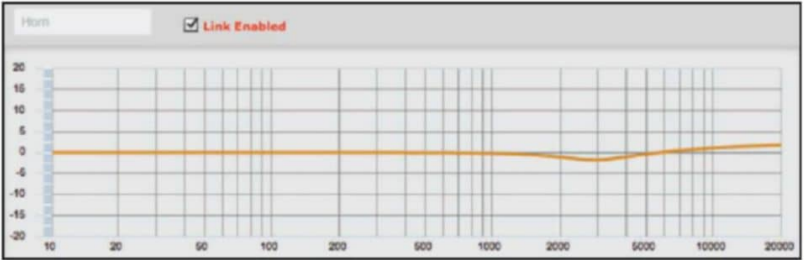


Figure 9: A little bit of EQ is used to dial in the exact tonality.

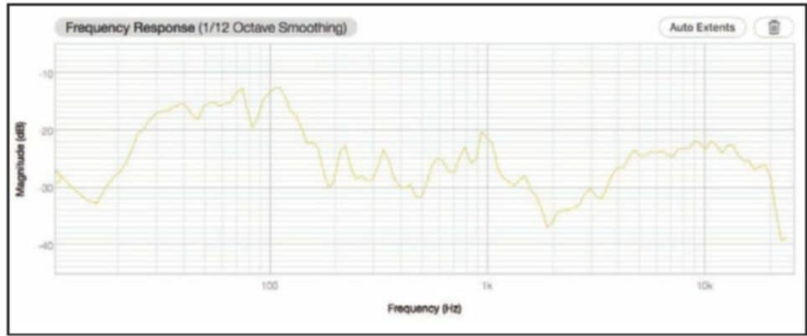


Figure 10: This is the frequency response of the back side of the speaker with the L-Pad at 3 o'clock.

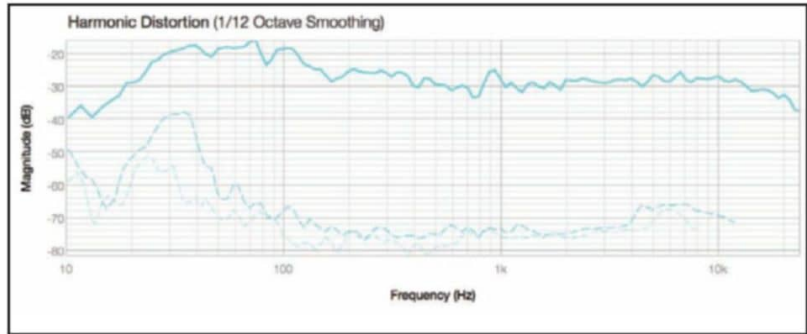


Figure 11: The graph shows the harmonic distortion for the complete system.

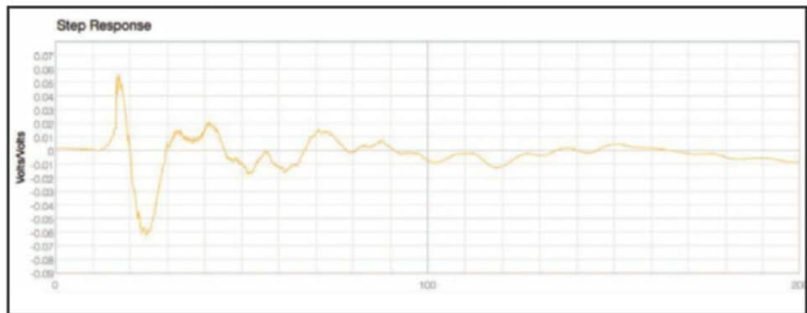


Figure 12: The step response is very clean and well damped.

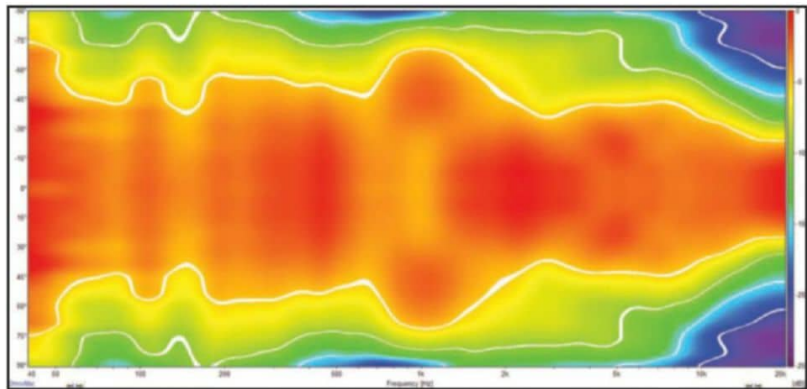


Figure 13: This is a polar heat map with contour lines at -6, -12, -18 and -24 dB. The -6dB beamwidth is consistently 60 to 120 degrees from 20 Hz to 20 KHz. Only dipoles (or very very large horns) are capable of this.

How Do They Sound?

These are truly high-definition speakers because they reveal everything. Stereo images are incredibly detailed and holographic, much like celebrated mini monitors are famous for... except the stereo image is huge.

Most speakers, truth be told, make musical instruments sound like they are being played by people who are 3' tall. With the Live-Edge Dipoles, a double bass sounds the size of a double bass. Instruments and performers sound larger than life, which is gratifying. You could say they sound like electrostatics when you're playing choral music and they sound like Klipsch when you're playing ZZ Top or Mickey Hart. They have the finesse of a KEF, Bowers & Wilkins, or Duntech, with the transparency of a MartinLogan, and the dynamic range of a Klipsch.

And the horns do not sound like horns at all. They just sound like great tweeters. And again, because of the Constant Directivity, they image well everywhere in the room. Planars and electrostatics don't do that. **Figure 5** shows the digital EQ curve that sounds best with these woofers. As you can see, I have 20dB of boost at 29Hz with a gentle roll-off below that point. Bass declines rapidly below 25Hz. Yes, there are a very few albums where you can tell that the dipoles are running out of steam at ultra-low frequencies. For 95% of music, these things shake the house.

The midrange EQ you see in Figure 5 only serves to correct some minor problems. The crossover function for the midrange is performed by this FIR DSP filter (see **Figure 6**). Together these filters tailor the 18" woofer + 8" midrange combination.

The tweeter is a combo of FIR filter, which only addresses the overlap with the midrange at 2kHz

Parts List

Speaker Parts (from Parts Express)

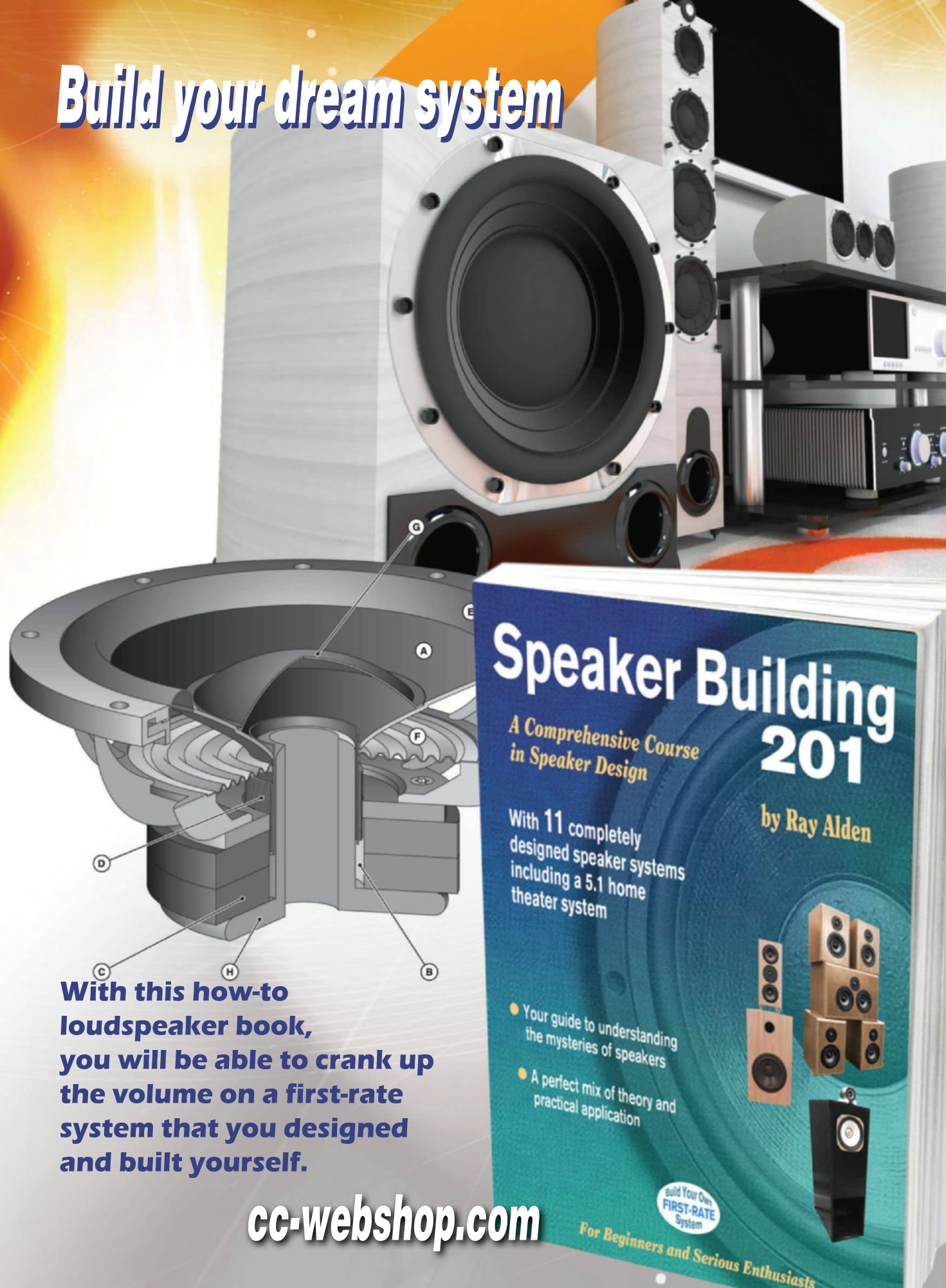
- 2 Eminence Kappa Pro 18LF 18" woofer
- 2 Radian 5208 8" coax mid-woofer with 16Ω tweeter (from Radian Audio Engineering)
- 2 PRV WG175PH horn tweeter
- 4 Binding Posts for speaker cables
- 2 L-Pad (Part #260-255)
- 1 MiniDSP 2x4 HD digital signal processor

* Measurements made with FuzzMeasure software. Power handling is 100W. If you add a subwoofer, they'll handle 500W or more.

Crossover Parts (from Parts Express)

- 5MH air core inductors
- 200μF is a combination of 100μF electrolytic + 2x 47μF film capacitors film capacitors
- 10Ω, 10W resistors
- Dayton 50W L-Pads

Build your dream system



With this how-to loudspeaker book, you will be able to crank up the volume on a first-rate system that you designed and built yourself.

cc-webshop.com

Figure 14: Here is the schematic for my Live-Edge Dipole speaker design.

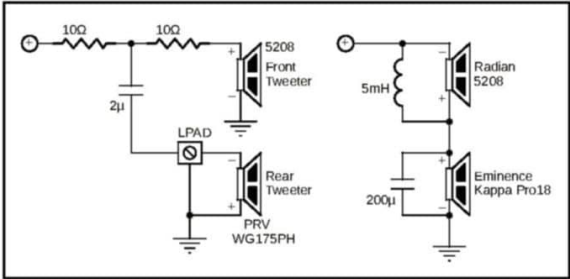


Photo 4: A few weeks after receiving my rough sketches, my cabinet maker sent this photo.

About the Author

Perry Marshall is an electrical engineer and business consultant, and the author of *80/20 Sales & Marketing and Evolution 2.0*. He lives with his family in Chicago, IL.



Project Files

To download additional material and files, visit:
<http://audioxpress.com/page/audioXpress-Supplementary-Material.html>

(**Figure 7**), which is combined with an 18dB/octave IIR filter to finish what the FIR filter couldn't (**Figure 8**), and a little bit of EQ to dial in the exact tonality (**Figure 9**).

On the back is a PRV WG175PH horn tweeter. Its response is similar enough to the Radian Coaxial. An L-Pad adjusts the level. This curve shown in **Figure 10** is when set at 3 o'clock. It was taken on the backside of the speaker. **Figure 11** shows the harmonic distortion. Above 50Hz, it's less than 1%.

Step response is the speaker's response to a sudden change from 0V to 1V. It shows you how much the woofer cone flops around before it comes to a stop. **Figure 12** shows that this is clean and very well damped, which indicates tight, well-controlled bass. The polar heat map shown in **Figure 13** confirms the capability of these dipole speakers.

Active Crossover DSP Files

The active crossover DSP files for this article can be found in the Supplementary Material section of the audioXpress website. However:

- These will ONLY work in the MiniDSP 2x4HD. Any other DSP requires your own files.
- These are based on actual measurements for my exact drivers. If you want to base these on the drivers you buy (you should!), you need to make your own measurements and convert them to a .bin file using a program (e.g., Eclipse Audio FIR designer, <https://eclipseaudio.com>) and upload them to the DSP unit.
- You will have to adjust levels of the drivers, and perhaps even the time delay between them, to your situation.

The passive crossover just does the absolute minimum and the DSP does all the heavy lifting. All crossover parts were purchased from Parts Express. **Figure 14** shows the schematic for my Live-Edge Dipole design, and I have also provided the Parts List.

Cabinets

I bought slabs of birch from Big Red Sawmill in Nebraska. You can use any kind of wood or other material you like. I bet you could find some really cool applications of reclaimed wood. When I bought mine, I didn't realize that the panels were just a little too narrow to comfortably accommodate the support structure behind the 18" woofers.

I hired Seth Cothron in Chicago, IL, to build the cabinets ([instagram.com/studio38designs](https://www.instagram.com/studio38designs)) and all I gave him was some sketch drawings and circles drawn on the wood. The dimensions can be found in the Supplementary Material on the audioXpress website.

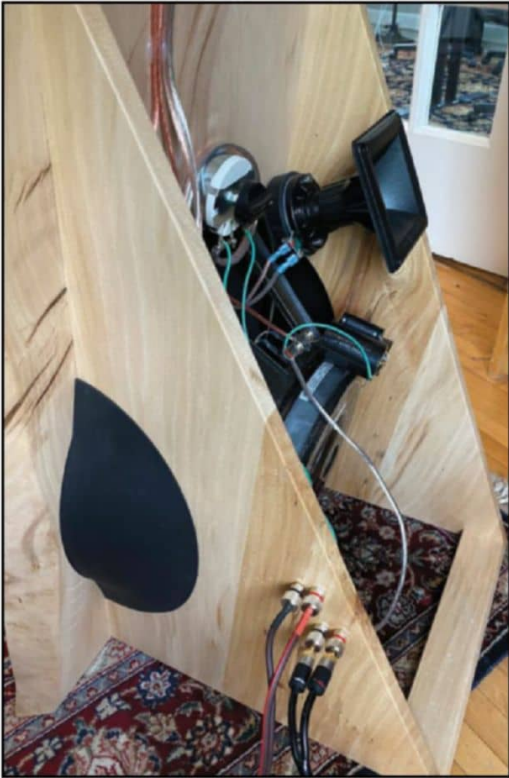


Photo 5: This is a back angle of the partially completed speakers.

Seth proceeded to use his artistic sensibilities and a few weeks later he sent me a stunning photo (**Photo 4**). Unfortunately, Seth shouldn't have cut those circles into the rear triangle pieces. Those would make a huge leak that will cut down my bass response. (Those holes would cost almost an octave of bass.) So, Seth created curved discs out of cardboard and fitted them to plug the holes (**Photo 5**). When you build the speakers, I suggest you choose stock that's an inch or two wider. Seth cut grille frames with a CNC machine (**Photo 6**).

In Summary

Trumpets sound incredibly real, as does upright bass. If you play electronic music like the Matrix soundtrack, they easily keep up with the deep bass and synthesizers. If you play string quartets and choirs, they have a presence and realism that gives me goosebumps and brings my 21-year-old son to tears. My friend Finn Peacock in Australia also built Live-Edge Dipoles. He emailed me today and said, "I couldn't drag my wife away from it—she got out all her favorite CDs that she hasn't touched in years, kicked me out of the garage and just listened in awe." I encourage you to spin off your own variation on this project! **ax**



Photo 6: The flush-mount grille hoops, made on a CNC machine and wrapped with dark blue cloth, complete the design. The live-edge birch is lightly stained with lacquer spray.

QLEX
series

beyma
European drivers since 1969

RISE OF
FORCE
FACTOR

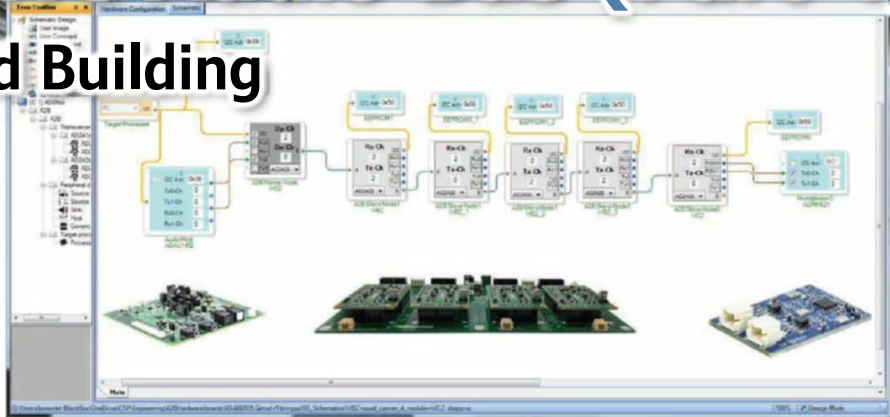
New 21QLEX1600Fe
subwoofer

4 layers Voice Coil
High Total SPL
Light weight - 17.1 kg
Also available in 18" and 15"

Getting Started with Automotive Audio Bus (Part 4)

Designing and Building Systems

By
Brewster LaMacchia



The first three parts of this article series examined the basics of A²B technology, how data moves between nodes with fixed latency, A²B phantom power, and the chip’s LVDS interface. This article takes a step back and looks at creating systems and the tools that are available to speed up that process. One of the nice things about A²B development is not only that its use license/royalty is free, and ADI offers the software libraries and SigmaStudio design tool at no charge.

Editor’s Note: A fully detailed look at off-the-shelf hardware and system building is too long for the magazine version of this article, so supplementary material with this information has been placed on the [audioXpress website](#).

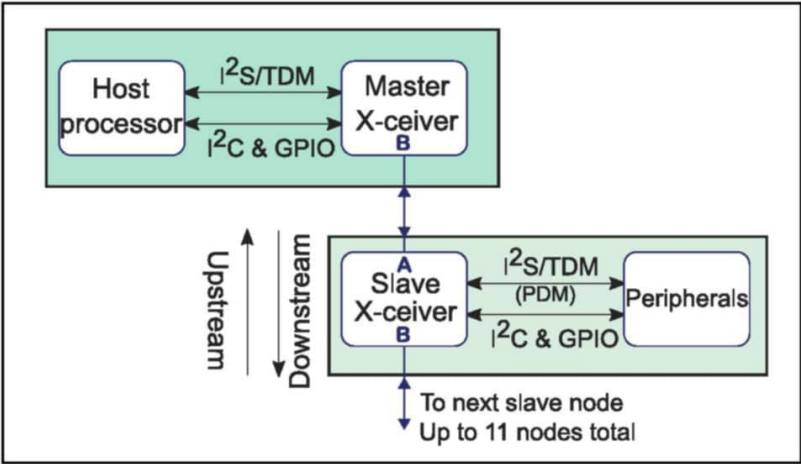


Figure 1: Basic A²B system features

All A²B systems consist of a root (A²B master) node at the head of the single twisted pair A²B daisy chain with one or more client (A²B slave) nodes as illustrated by **Figure 1**. There is an overall limit of 11 nodes with up to 15m of cable between them and a 40m total system cable length.

The decision for a host processor choice is not driven by the needs of A²B control, even a small microcontroller would be fast enough to load the network configuration over I²C and manage status and control when it’s operating—mostly because I²C just isn’t that fast at 400kbits per second. That speed makes for a practical limit on how fast commands can be sent.

This I²C limit is a consideration for the node design. If the peripherals are all chatty over I²C, you’ll quickly run out of I²C bandwidth on the link between the master node transceiver and the host processor. If there are local processors on slave nodes, it is likewise best to have them boot from local flash as downloading code at startup to every slave node with a local processor would slow the startup time. A²B has a provision for holding each node’s local configuration in a small EEPROM to

Engineering Consulting for the Audio Community

Engineering
Success
since 1983

MENLO
SCIENTIFIC



- Liaison Between Materials, Suppliers, & Audio Manufacturers
- Technical Support for Headset, Mic, Speaker & Manufacturers
- Product Development
- Test & Measurement Instrumentation
- Sourcing Guidance

MENLO
SCIENTIFIC

USA • tel 510.758.9014 • Michael Klasco • info@menloscientific.com • www.menloscientific.com

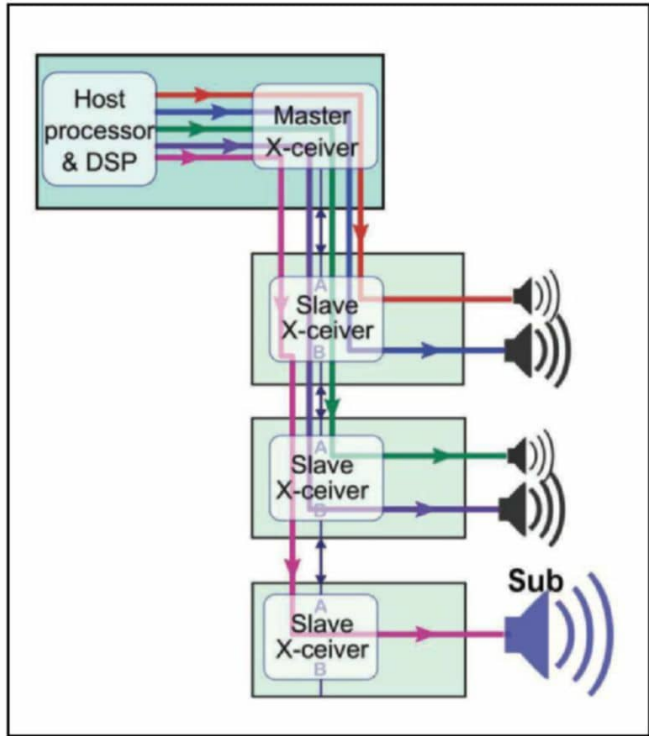


Figure 2: Hypothetical speaker system with centralized DSP

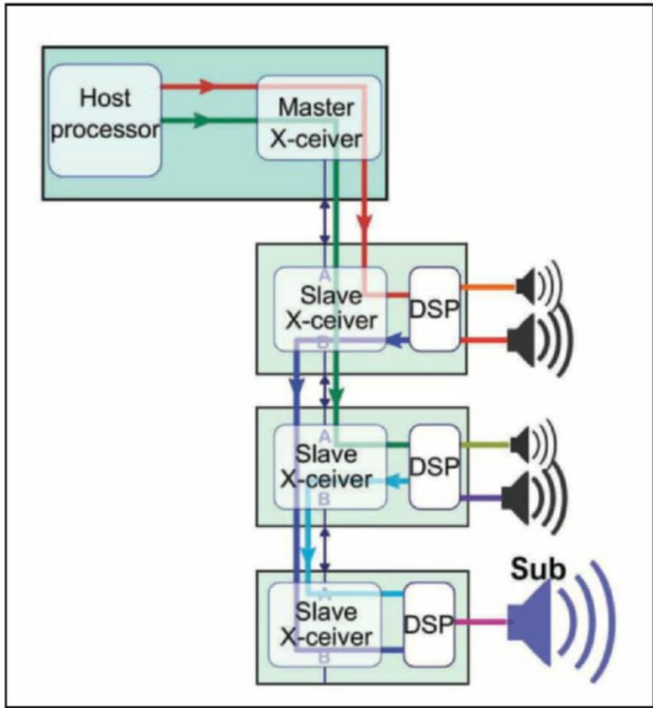


Figure 3: System implemented with distributed DSP reduces host requirements and makes expansion simpler.

further reduce startup time. It's possible to have a local processor configure the slave node from the local side as well as the transceiver supports multimaster I²C operation.

Most likely, the host processor has plenty of compute resources and the additional needs of the A²B bus control will be minor relative to the other processor tasks. What will require consideration in a host is the nature of the audio data and what type of processing must be performed on it.

This may lead to two processors on the host—one that runs general system control tasks and

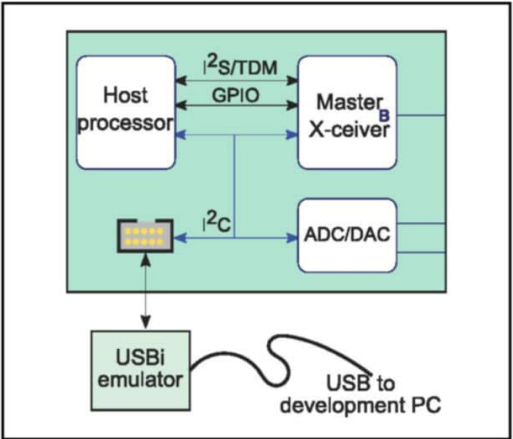


Figure 4: USBi and I²C interconnect for supporting host PC development

the other optimized for the typical DSP processing associated with audio applications that need more than a channel or two of audio. The DSP algorithms may be performed in a general processor core with DSP-like extensions, in a DSP core, or in a FPGA device. The details of that decision are driven by considerations outside of the A²B bus. The only thing to include criteria wise is that the device that is intended to process the audio have I²S ports with TDM modes that support the bit depth, channel count, and sample rates that are required.

If the system is performing active noise cancellation (ANC) or similar tasks that require low latency from input to output, then the device's I²S ports and processing design should provide very low latency—a dozen samples at most is typical of many of these applications. This is where the low latency of A²B can really be an advantage as its two-sample latency gives leeway to the host maybe not being the fastest tool in the shed.

Client/Slave Nodes and Peripherals

The flexibility of A²B offers system designers a huge range of design choices. For example, the node could have a small DSP (i.e., ADI's SigmaDSP) or powerful FPGAs to perform local pre and/or post processing of audio streams.

A common audio speaker configuration is a stereo two-way (mid, tweeter) speaker with a common subwoofer. We could perform the crossovers on the host node and use five A²B slots for data movement as shown in **Figure 2**. If we want to allow more speakers to be added—let's say to create a basic 5.1 surround system—we'll use more A²B channels, and now the host needs to ship with enough DSP resources to handle the worst case system. Alternately, as illustrated by **Figure 3**, we could use one A²B slot per speaker channel and let the local node implement the crossover and handle other local speaker equalization tasks. Non-full-range speakers would send low-frequency content to the subwoofer node, and we can easily add more subwoofers to improve performance there too.

From this basic concept, we could add features like voice input, or a Bluetooth receiver, or even an analog input for legacy source in to add-on units or even right in to the speaker box.

With flexibility comes the risk of featuritis. Creating prototypes and getting user and customer feedback before embarking on a full design effort would be critical for development of a successful product. Counterbalancing that is the time and cost to

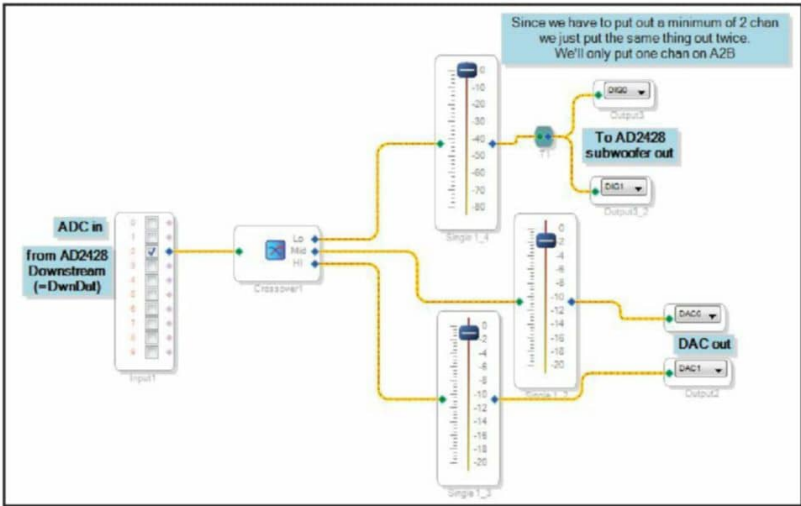


Figure 5: SigmaStudio schematic for speaker node. The speaker node receives a full bandwidth channel from the AD2428 and processes it in the crossover block. The tweeter and mid-range are output to the DAC where they would connect to an amplifier and the drivers. The bass content is output back to the AD2428.

ALL ELECTRONICS CORPORATION

Thousands of Electronic Parts Available Today!

LEDs • CONNECTORS • RELAYS
SOLENOIDS • FANS • ENCLOSURES
MOTORS • WHEELS • MAGNETS
PC BOARDS • POWER SUPPLIES
SWITCHES • LIGHTS • BATTERIES
and many more items...

We have what you need for your next project.

www.allelectronics.com
Discount Prices • Fast Shipping

SpiralAir[™] Interconnects

Coax Is Cool Again!

Cardas Golden Strand
Air Filled Cotton Dielectric
Unique Spiral Litz Shield
ETI Tellurium Copper Connectors

Affordable Litz Interconnects and Bulk Wire
Tonearm - Phono - Line - Loudspeaker

KAB

Preserving The Sounds Of A Lifetime

www.kabusa.com

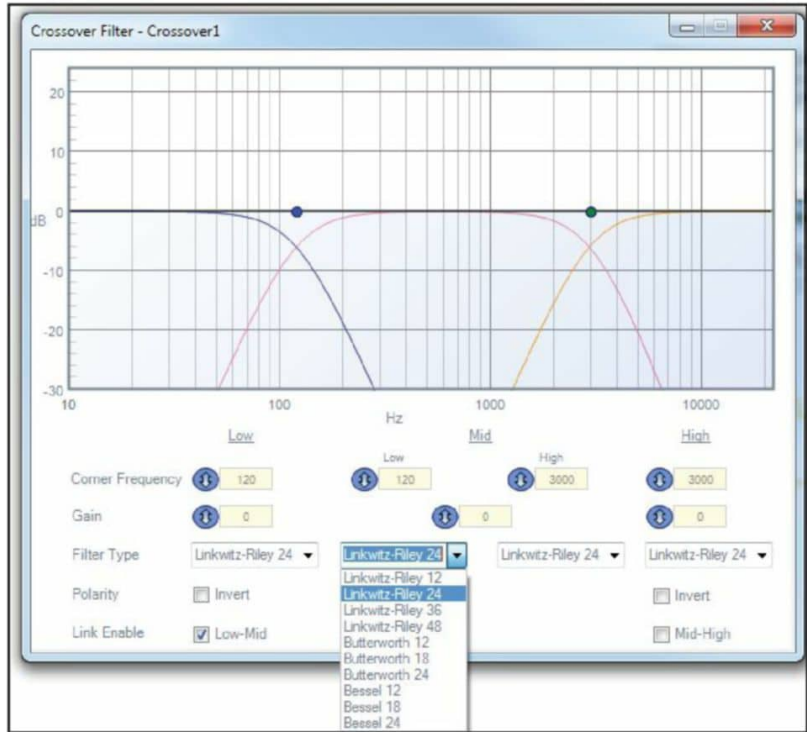


Figure 6: SigmaStudio’s basic speaker crossover controls showing the default filter types and settings.

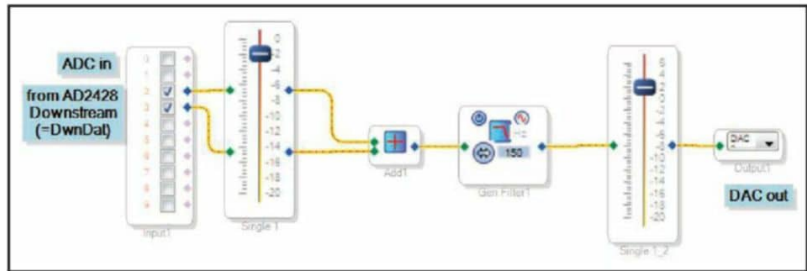


Figure 7: Subwoofer processing takes in the bass data from the speakers, sums it, performs additional EQ if desired, and outputs it to the DAC to the amplifier input. To allow for more than two speaker nodes, additional input channels could be connected to a larger summing node. A real subwoofer would have blocks for a more extensive EQ, a limiter, external gain control, and possibly a compressor for a night mode function.

Project Files

To download additional material and files, visit <http://audioxpress.com/page/audioXpress-Supplementary-Material.html>

Resources

A²B and Automotive Audio Bus are a registered trademark of Analog Devices, Inc. ADI’s main A²B webpage provides links to part information, ADI tools, and design information. www.analog.com/en/applications/technology/a2b-audio-bus.html

develop those prototypes. This is where ADI and its third-party ecosystem can help by allowing creation of bench demos in a few hours and integration in to end user form factor systems for evaluation in a short time period—weeks, not months.

Host/Master Node

There are a number of boards available for evaluation and prototyping. The choice will be influenced by the project’s requirements as well as familiarity with the vendor and the general software tools. In addition to the boards described in this article’s supplementary material, any suitable processor board with I²C, GPIO, and I²S can have A²B Frankensteined on to it with an A²B breakout module.

A host node has two primary responsibilities; setting up the network and providing the clock that defines the audio sample timing. All nodes’ I²S clocks are synchronized to the master node’s clock source.

To use ADI’s SigmaStudio tool for A²B development ADI’s USBi emulator module is used. It provides a USB connection for the development PC and connects to the host’s I²C bus as shown in **Figure 4**. USBi interfaces boards are available from ADI as well as several third-party board vendors and can be used for any SigmaStudio target.

The USBi operates as an I²C bus master to program the A²B transceiver. During development, the host would skip that portion of code as well as not try and master the I²C bus when the USBi is being used. Host boards with ADI’s SigmaDSP chips on them can be programmed through the same USBi interface.

Client/Slave Node

Most of the available host/master boards can also function as A²B client (slave) nodes. Slave nodes may be locally powered or phantom (bus) powered. An important consideration is phantom powered slave boards cannot be easily probed with an oscilloscope because their local ground node is not the same as the system ground (see Part 2 in *audioXpress*, October 2020) unless the audio I/O portion is galvanically isolated from the A²B portion. One workaround is capacitively coupling the analog I/O grounds. The supplementary material details available A²B client (slave) nodes in depth.

Building an Example Application

ADI’s SigmaStudio software tool will be used to build the application pictured in Figure 3, where the speaker crossover is in each node. This simple example could be expanded further by adding things such as multiband EQ to equalize the speaker’s response, or soft limiting to avoid excess

driver distortion at high SPL levels, and so forth.

For the local processing, we need two different DSP applications, one for the small speakers and the other for the subwoofer. The speaker processing is a three-way crossover where we output the mid-range and tweeter locally and the bass is sent over A²B to the common subwoofer (**Figure 5**). Additional level controls are added to simplify system balance when different speakers might be used.

There are a number of options for crossovers and speaker EQ in SigmaStudio, including self-tuning options. Keeping this example simple, a basic three-way crossover will be used. Its settings are shown in **Figure 6**. We’ll overlook details of actually setting the crossover as that topic alone could fill several articles. The subwoofer node’s processing is illustrated by **Figure 7**.

Next the A²B schematic is created (**Figure 8**). It consists of one master node and three slave nodes. One immediate thing that can be noticed is that the peripherals all have the same I²C address on each node. There is an I²C mux facility built in to the A²B I²C subsystem for uniquely addressing each node, eliminating most of the traditional headaches of I²C address conflicts.

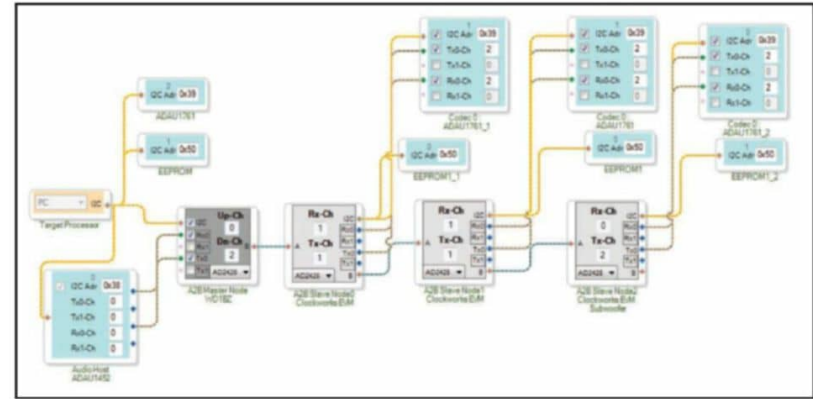


Figure 8:The A²B network topology in SigmaStudio uses off-the-shelf hardware. The three slave nodes each have a ADAU1761 SigmaDSP chip in them; SigmaStudio lets us connect I²S and I²C to match the physical hardware design. The ADAU1452 SigmaDSP-based master node is on the left; it uses an ADAU1761 as that node’s ADC and DAC. As it doesn’t connect directly to the AD2428, its I²S connections are not managed in this setup.

About the Author

Brewster LaMacchia started in signal processing working on military satellite communications systems. Since then he has worked on a wide range of signal processing systems, including a number of years on video processing and surround sound audio for consumer home theaters. Recently, he started Clockworks Signal Processing to develop open-source hardware for multichannel signal processing. He also dedicates time to public astronomy events to expose people to science and critical thinking. In the past 10 years, he’s helped more than 7,000 people look through a telescope for the first time.



Ribbon Loudspeakers
what you need to know.

Order your copy
cc-webshop.com

PURIFI
www.purifi-audio.com

PURE SOUND

Ultra High Performance Class D Amplifier Module
4" & 6.5" Ultra Low Distortion Long Stroke Drivers

1ET400A PTT6.5W PTT4.0W

sales@purifi-audio.com



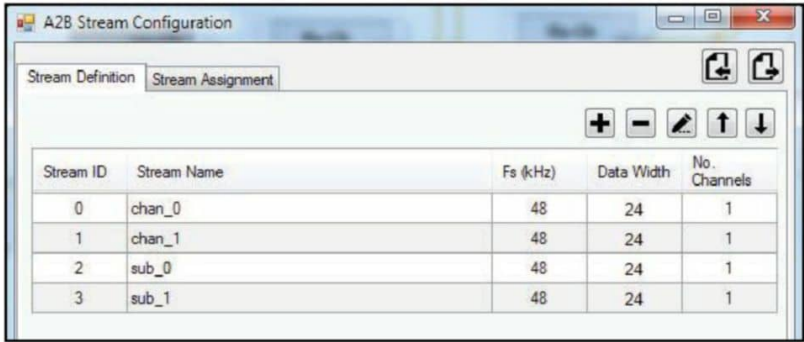


Figure 9: Stream definitions for this speaker example are based on Figure 3.

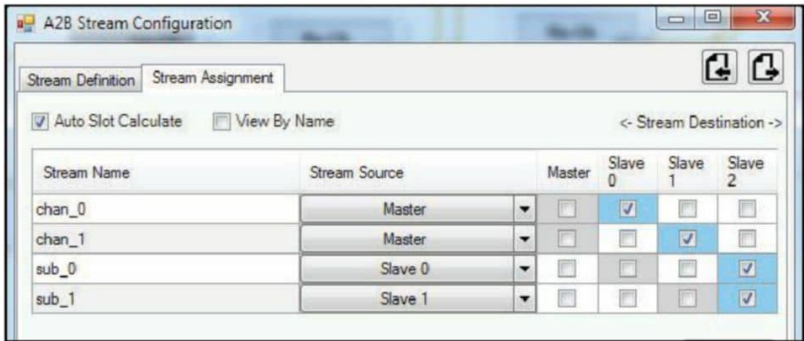


Figure 10: Stream routing where we have assumed that the stereo audio originates in the master node, and the A²B network is wired in a L, R, Subwoofer order

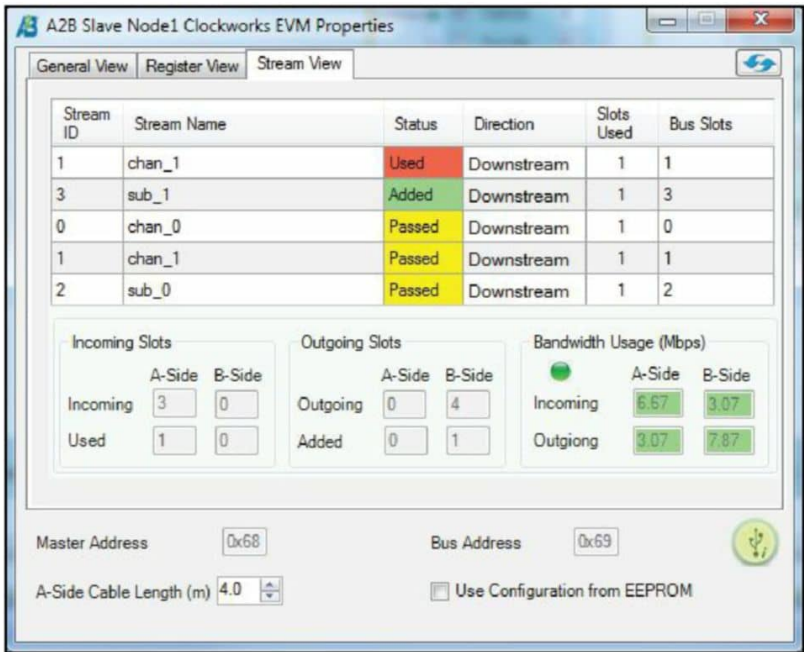


Figure 11: Audio stream detail for the second slave node in this example. This enables an easy-to-comprehend view of data originating from or being output to this node, along with what is being passed in each direction.

The last part of the setup is to define streams for the audio that is to be routed and then indicate which node produces the stream and which node(s) consume the stream (**Figure 9** and **Figure 10**).

SigmaStudio can also look at the node’s properties to set the AD2428’s characteristics through either user friendly settings or raw register bits. For example, in **Figure 11**, we can see the audio stream activity for the second speaker node.

This example illustrates the basics of A²B configuration but overlooks some of the more complex issues that will be present in real-world systems. For example, the A²B schematic defines specific electrical positions in the daisy chain for each function (L, R, and subwoofer). What if the subwoofer was to be placed close to the master node (i.e., first in the daisy chain)?

Each node can configure its AD2428 and the DSP locally. We can assign unique IDs to each node, and could encode in that ID the type of node (speaker or subwoofer in this example). Neither of these help with the data routing over A²B, as A²B is not a network like Ethernet where nodes are defined by an address vs. physical position.

If there are a limited number of configurations then it’s possible to create diagrams for each and determine at runtime which to use. If all nodes route all data to and from the host, then the host can directly manipulate routing.

Beyond that it becomes necessary to use ADI’s A²B software stack along with custom software to dynamically manage the A²B network. Available as a C library for adaptation in to whatever environment the host processor uses, the A²B stack also allows sophisticated fault and error handling versus the simplified load-and-run approach when working exclusively with a SigmaStudio design flow.

Summary

The first three article parts looked at the underpinnings of A²B technology and the electrical design of the A²B interface. This last part of the A²B article series provided details on creating actual products around Analog Device’s A²B technology. Due to space limitations, the detailed review of available hardware for development has been placed on the audioXpress website.

There are many features and capabilities that this article series skipped over. Now when faced with an audio distribution problem, you’ll have enough understanding to know if A²B should receive further investigation as a solution to your design problem. There’s nothing like getting a couple of boards and experimenting. **ax**



**ELECTRONIC
TRANSFORMERS**

- Original Equipment
- Replacement
- Amateur

TRIAD MAGNETICS, INCORPORATED

460 HARLEY KNOX BLVD., PERRIS, CA 92571

www.TriadMagnetics.com

MODERN DESIGN MEETS VINTAGE SOUND



Dayton Audio HTA100BT 100W Hybrid Stereo Tube Amplifier with Bluetooth and Subwoofer Output

Debates may rage between whether solid-state or tube amplifiers produce a better listening experience; however, Dayton Audio solves that dispute! The HTA100BT Hybrid Stereo Tube Amplifier successfully employs contemporary amp technology that faithfully pays homage to vintage amp design. By harnessing the power and efficiency of solid-state class AB amp topology and a vacuum tube preamp section, the HTA100BT authentically reproduces tube amps' warmth and character with the flexibility the modern world expects.



- Authentic vacuum tube warmth and character with class AB amplifier efficiency produces an articulate and natural sonic experience
- Powerful 50 watts per channel of audiophile fidelity that is sure to please even the most discerning ear
- Convenient input connections including USB, optical, coaxial, stereo RCA jack, phono preamp, and Bluetooth V5
- Mono RCA subwoofer output gives you the ability to enjoy a 2.1 system when connected to an active sub
- Brushed aluminum housing with vintage front panel VU meters lends a clean, modern look with a nod to classic aesthetics
- Short circuit protection circuitry to prevent amp failure

Custom OEM inquiries welcome.

SAVE \$12
off \$100 order!

Code: AXMQ121
Expires 2/28/2021

DAYTONAUDIO
Your path to great sound.™

Distributed By:

**PARTS
EXPRESS**
AUDIO COMPONENTS & SOLUTIONS

Tel: 800-338-0531
725 Pleasant Valley Dr.
Springboro, OH 45066