

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Audio Praxis

Audio Archiving, Preservation, and Restoration at the Library of Congress

By Scott Dorsey

Practical Test & Measurement

USB Oscilloscopes LabNation SmartScope Review

By Ethan Winer

You Can DIY!

Reflective-Way, Two-Way System—Part 1

By Peter Lehmann

The Incredible Egg Loudspeaker Part 3—The Proof Is in the Listening

By Thomas Perazella

Back EMF

Reactive Impedances
and Energy Losses—Part 4

By Andy Lewis

Book Review

High Fidelity Circuit Design
by Crowhurst & Cooper

Review by Morgan Jones

Sound Control

Modulation Distortion
in Loudspeakers—Part 3
Detectability Threshold
of Doppler Distortion

Hollow-State Electronics
Screen-Grid Biasing

By Richard Honeycutt



PARTS CONNEXION

The Authority on Hi-Fi DIY

Enter code: **AX0817**
in the comments box at
checkout to receive
25% Off
your next order.

**Your #1 Source for
Vacuum Tubes, DIY Parts,
Audiophile Accessories**

**&
Premium HIGH-END Audio Gear**
(Discontinued • Demo • B-Stock)

CAMBRIDGE AUDIO • MONITOR AUDIO •
CARDAS • REGA • FOCAL • WHARFEDALE •
MUSIC HALL • CYRUS • DEVIALET •
ACOUSTIC ZEN • QED • BDI • SOLID TECH •
APOLLO • TARGET • GRADO

Sales:
order@partsconnexion.com

www.partsconnexion.com

Inquires:
info@partsconnexion.com



905-631-5777

Toll-Free - US/Canada

1-866-681-9602

905-681-9602



5403 Harvester Rd, Unit 1, Burlington, Ontario, Canada L7L 5J7

Debit, Visa, Mastercard, Amex, PayPal, EMT, Money Order & Bank Draft



**HAMMOND
MANUFACTURING®**



MANUFACTURING & DESIGNING TRANSFORMERS & ENCLOSURES SINCE 1917

Chassis, Plates & Covers



Tube Output Transformers

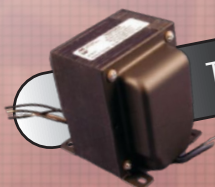
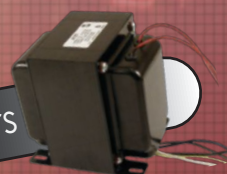


Plate & Filament Transformers



Open & Enclosed Chokes



USA

Tel: (716) 630-7030

Fax: (716) 630-7042

Canada

Tel: (519) 822-2960

Fax: (519) 822-0715

UK

Tel: 01256 812812

Fax: 01256 332249

Australia

Tel: 8-8240-2244

Fax: 8-8240-2255

www.hammondmfg.com



November 2018

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: 800-269-6301

E-mail: audioxpress@pcspublink.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. 2018
Printed in the US

It's About the Sound

There are times when we just need to stop and look back to appreciate and be thankful for those who share their work and lives with us and unfortunately have departed. Just in the past weeks we have mourned the loss of Siegfried Linkwitz (1935–2018) and David Clark (1941–2018). Great engineers, brilliant minds, friends which will be missed. In *Voice Coil* October 2018, Vance Dickason writes and honors the memory of Siegfried Linkwitz.

Unfortunately, we also received news of the passing of a very close friend and contributor for *audioXpress*, Ron Tipton. Born in Chicago, IL, Ron studied electrical engineering in New Mexico State University and was retired from an engineering position at White Sands Missile Range. In 1957 he started Testronic Development Laboratory (TDL Technology, Inc.) to develop audio electronics. During the 1960s and 1970s, TDL built active filters and audio test equipment for well-known companies such as Bose Corp. and Acoustic Research. He remained the TDL president and principal designer until 2015.

Until very recently, Ron directed his audio enthusiasm toward TDL Technology from his home base in Las Cruces, New Mexico. Many of the products sold on his website www.tdl-tech.com were the projects that he carefully designed, built, and explained in his many articles for *audioXpress* (and previously for other titles originally published by Ed Dell).

The website is still online and keeps a collection of links that reflect Ron's diverse personal interests. Unfortunately, the repository of TDL products has been removed since he decided to discontinue sales activities in 2015. Ron frequently would display many of those products in his writings for *audioXpress* and kept using TDL products as the personal reference he knew and trusted.

Ron was an avid magazine reader and would frequently react to projects published in *audioXpress*. In June 2006, he wrote a letter to then-publisher Ed Dell saying, "I'm writing with regard to 'A Solid-State Single-Ended Power Amp' by Ed Simon in the April issue of *audioXpress* (www.audioxpress.com/article/a-solid-state-single-ended-power-amp). I can't resist unusual power amps, so as a result I have an 8' rack full of them."

TDL products were frequently the result of Ron wanting to explore those ideas, measure, and improve them. Sometimes, he would design a product just out of pure necessity, like the TDL Model 108 Noise Generator he did he 2010—and which can be found online (www.audioxpress.com/article/Audio-Testing-With-Noise)

Sometimes *audioXpress'* authors would review TDL products, like the model 4010 Restoration Preamp, which Don Walizer admired in detail in 2009. He wrote, "I have been searching in vain for a modern replacement for the venerable OWL Audio Restoration Module that would incorporate the OWL's functionality with updated electronics and a more refined filter design." At the end of the review Walizer concludes, "I sold my OWL two weeks ago. The 4010 does everything the OWL did, only better."

Just browse all the online references we have online under Ron Tipton's name: www.audioxpress.com/authors/23376/ron-tipton. *audioXpress* has more than 100 entries on record—and some are freely available online. We will make sure to publish many more examples of his work in the future.

His last contribution for *audioXpress* (chronologically) was his Omni Pipe speakers (an omnidirectional design). Ron was inspired by Ken Bird's Omni Hex project in the December 2017 issue of *audioXpress* and decided to give us his take. We just published the article in the October 2018 edition. The project is so much more remarkable given that Ron was already very ill. As soon as his condition slightly improved, he resumed the work and made sure to send it to *audioXpress*, as he did for so many years.

We learned of his passing, through an email sent by Ron's companion Bev Denney, where she shares: "Ron died on September 15. He had held out for a long time." I will always remember the warm welcoming I received from Ron when I started working with *audioXpress*, precisely five years ago. In 2013 he told me, "I wrote my first published article in 1957 or '58 for *Electronics World* (I think)," to which I replied, "Right! I was born in 1965..."

We had frequent email exchanges and Ron would react positively to my suggestions, and eventually we had to enlist him as a columnist—he contributed articles every month (sometimes more than one), which we published in his "Realistic Audio Reproduction" series, subsequently renamed with column title "It's About the Sound." He was always anxious to have his take on the latest technologies and review the latest products, never afraid of exploring areas that were completely new for him. In the last message I received (July 14) he says: "I think a comparison review article of the Waves B360 Ambisonics Encoder with a similar product would be informative. I'm working on a comprehensive article on reverberation time, and I'm ready for suggestions about what to do next."

Thank you, Ron, I'll make sure we keep your passion alive. We will miss you.

Ron Tipton (1936–2018)

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, Joseph D'Appolito, Vance Dickason, Jan Didden, Scott Dorsey, Gary Galo, Gerhard Haas, Chuck Hansen, Richard Honeycutt, Charlie Hughes, Mike Klasco, Ward Maas, Oliver Masciarotte, Nelson Pass, Christopher Paul, Bill Reeve, Fernando Rodrigues, Steve Tatarunis, Ron Tipton, Stuart Yaniger

OUR NETWORK

audioxpress

C circuit cellar

LIS LOUDSPEAKER
INDUSTRY
SOURCEBOOK

VOICE  COIL

SUPPORTING COMPANIES

ACO Pacific, Inc.	51	Marchand Electronics, Inc.	49
All Electronics Corp.	55	Menlo Scientific, Ltd.	31
Avel Lindberg, Inc.	57	Parts ConneXion	2
Celestion	29	Parts Express International, Inc.	68
COMSOL, Inc.	13	Redco Audio	43
Eastech	15	Ross Video, Inc.	23
Hammond Manufacturing, Ltd.	3	Schiit Audio	24, 25
Hypex Electronics BV	61	Triad Magnetics	67
KAB Electro-Acoustics	57	Vidsonix Design Works	51
Linear Integrated Systems	17	Warkwyn	21

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 edition of the magazine.

COLUMNISTS

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

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

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

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

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

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





● Features

10 An In-Depth Exploration of *Electronics for Vinyl* (Part 2) The Remaining Chapters and Final Thoughts

By Gary Galo

Gary Galo, a well-known author on turntables and phono preamps, concludes his review of Douglas Self's recently published book *Electronics for Vinyl*.

16 *High Fidelity Circuit Design* by Crowhurst and Cooper

Review by Morgan Jones

Resurrected 60 years after its debut, Morgan Jones discussed the merits of this book, which is an interesting historical view on the technology and methods of the day.

18 Audio Archiving, Preservation, and Restoration at the Library of Congress

By Scott Dorsey

Scott Dorsey attended the Audio Engineering Society's first International Conference on Audio Archiving, Preservation, and Restoration and details his visit to the Library of Congress.

32 USB Oscilloscopes

A Look at the LabNation SmartScope

By Ethan Winer

When Ethan Winer found himself in need of an oscilloscope, he chose LabNation's SmartScope. In this article, Ethan Winer reviews the oscilloscope.

36 The Incredible Egg (Part 3) The Proof Is in the Listening

By Thomas Perazella

The final part of this article series concludes the build of this intricate Egg speaker project and details the results.

44 Reflective-Way, Two-Way System (Part 1)

Preparing for the Build

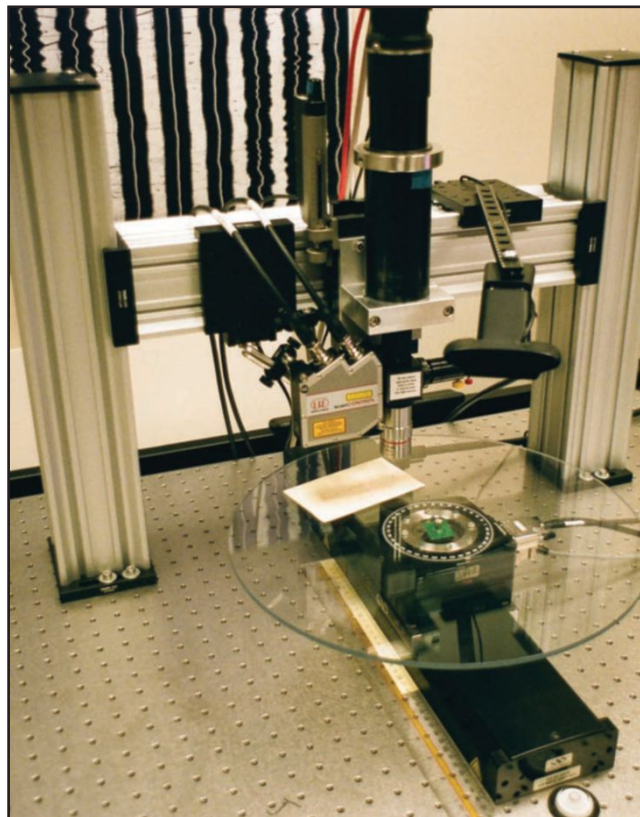
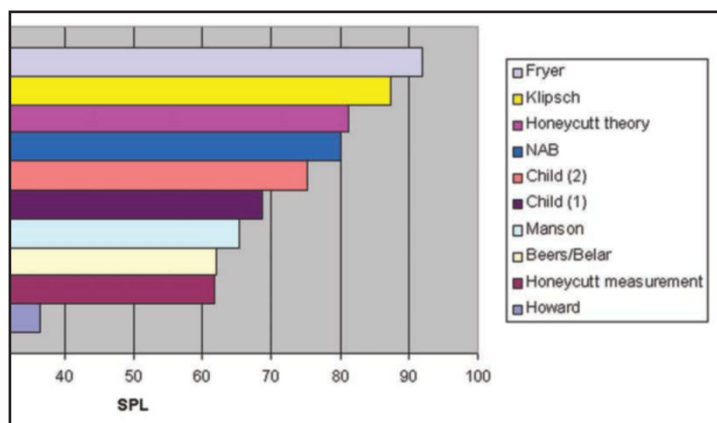
By Peter Lehmann

In this two-part series, Peter Lehman shares his design for a loudspeaker system with an integrated reflector for the tweeter.

52 Back EMF Phase Relationships in Moving-Coil Loudspeakers (Part 4) Back EMF Reactive Impedances and Energy Losses

By Andy Lewis

Andy Lewis expands upon the Back EMF concepts hinted at in the previous three articles in this series.



● Columns

8 The Loss of an *audioXpress* Icon

A Tribute to Ron Tipton

In this issue, we say farewell to Ron Tipton, one of our most esteemed and prolific writers who died at the age of 82.

SOUND CONTROL

26 Modulation Distortion in Loudspeakers and Its Detectability (Part 3)

Experimentation to Find the Threshold of Detectability of Doppler Distortion

By Richard Honeycutt

HOLLOW-STATE ELECTRONICS

62 Screen-Grid Biasing

By Richard Honeycutt

● Departments

4 From the Editor's Desk

5 Client Index

66 Industry Calendar

● Websites

audioxpress.com

voicecoilmagazine.com

cc-webshop.com

loudspeakerindustrysourcebook.com



@audioxp_editor



audioexpresscommunity



linkedin.com/company/audioxpress

The Loss of an *audioXpress* Icon

A Tribute to Ron Tipton



Ron Tipton in his home lab in 2013. He said: I've been building most of my own audio equipment for more than 60 years. Getting into building your own equipment can be addictive so beware! But if you have the inclination and the means, go for it! You will learn a lot in the process, not only about audio techniques but also how to properly do the construction.



This Model 465 6-Channel Volume Controller is one of the many TDL products that Ron designed simply because it was something he needed for himself.

Sadly, on September 15, 2018, the *audioXpress* community lost one of its most esteemed and prolific writers, Ron Tipton, who died at the age of 82.

Upon his passing, Ron's longtime companion, Beverly Denney, shared some details about Ron's early life and provided a more intimate look at his life outside of the audio engineering business.

Bev said, "Ronald Bourne Tipton was born in Chicago, IL, on August 30, 1936. In those days, children were given much freedom to explore. His Chicago

home was only a few blocks to the museums, libraries, and other sources for learning. He participated in Chicago's "Wiz-Kid program in the 1940s. His family moved to Kansas City, MO, in 1946. His house was a block from the wonderful Swope Park streetcar line, which gave its passengers to all points in Kansas City. He explored that city."

"Ron was a 'real great generation' person. He not only knew lots about electronics—his degree was in electrical engineering and chemistry—but he was a house builder, a painter, a concrete worker, a plumber and he was extremely serious about correct usage of the English language. He became an early organic gardener and later earned a MS degree in herbology and a PhD in nutrition."

Ron entered the White Sands cooperative program in 1956 to work half time and to go to what is now New Mexico State University. He finished the course work for the PhD studies but did not write the dissertation since he was already self-employed.

Bev recalled that Ron loved reading and going places and had a good sense of humor. His fantasy life was to live in the Caribbean on an island with a good sailboat. He had that life briefly with his first wife, Virginia, who died too soon.

"I had lost my husband the same year. We both were seeking companionship and we found each other. We had all sorts of adventures and experiences throughout the 36 years that we were companions," shared Bev.



Ron's listening room was a familiar space for many *audioXpress* readers since he would show it frequently in his articles. In this image, we can see Ron's three-way speakers with a subwoofer on the floor and the omni-directional speakers, the Omni Pipes, which was the last project he wrote for *audioXpress* (October 2018).

Bev concluded, "Ron enjoyed the research and writing the articles for *audioXpress*. He was determined to finish the latest speakers and write the article. The magazines arrived here on that Saturday that he passed away. It made me feel very proud."

Throughout his long and illustrious career, Ron shared his knowledge and his designs with other like-minded audio enthusiasts and they appreciated his willingness to lend a helping hand.

Chuck Hansen said, "While I never met him in person, I knew Ron fairly well, and we kept in touch by email starting in 2009. You may remember him as the champion of the TL072 op-amp in his Testronics Development Labs (TDL) designs. He started TDL in the late 1950s and I liked his practical design approach. He worked at White Sands Missile Development Range at one time, so we would discuss military design and qualification testing in detail."

Chuck shared a story that really encapsulates what Ron Tipton meant to the DIY community and audio enthusiasts in the industry. He was very knowledgeable and always willing to lend a hand, or in this case, an instruction manual.

"In 2010, I bought a PM-720 Phase Meter on eBay. It had the usual caveat about the seller not being able to test it, but the power light came on, so I took a chance. There was no real information about the manufacturer on it anywhere," recalled Chuck. "There was no schematic or other information with the PM-720, but the unit appeared to work based on oscilloscope measurements of two phase-shifted sine waves. It even had an internal shunt so you could compare voltages and currents. I called Ron to see if he ever heard of it."

Chuck continued, "Not only had he heard of it, but he owned one and had a manual for it. He sent me a scanned copy, which had the calibration information but no schematic. I took measurements of many of the circuit voltages and scope waveforms, and sent Ron a copy.

Chuck said, "The meter worked well for a few years, and I used it to measure phase shift between voltage and current in transformers for work projects. Then one day it stopped working, and I decided it was time to reverse-engineer the circuitry so I could attempt to



Ron Tipton, in 2015, is shown in his element, working on a project in his shop.

repair it. I turned to another industry veteran, Bob Pease at National Semiconductor. He faxed me some datasheets for the analog ICs, and was very helpful in describing why the early analog ICs only used NPN transistors. I guess I should get the PM-720 working again someday, in honor of all the information Ron sent me, as well as the Fairchild semiconductor datasheets and discussions of early analog IC design I had with Bob Pease. We are losing the early contributors to our analog and audio family far too often now."

Chuck concluded, "Over the years Ron and I would email each other with comments about articles we had in *The Audio Amateur* (the predecessor of *audioXpress*) and its successor magazines. Ron had a series of articles in *audioXpress* about audio measurement and recording computer programs in the past number of years. He was a great designer and I really appreciate his friendship."

Stuart Yaniger, another of our prolific and expert contributors, shared the following thoughts about Ron Tipton.

"One of the great delights in being part of the *audioXpress* family is the people by which I'm surrounded. Ron Tipton's articles were always a delight to read, showing deep technical mastery and a rare ability to write clear and uncluttered English. He had a knack of explaining difficult concepts, and his prose was unpretentious and direct. I was lucky enough to have the opportunity of collaborating with Ron on a review of some room correction hardware and software, where his expertise really helped me gain understanding. This was doubly gratifying because Ron also set a shining example for us: Many people's knowledge ossifies as they get older, but certain minds are always curious, always open to new ways. Ron was all of that."

Stuart added, "Collaboration with Ron initiated a correspondence where he was amazingly patient and generous in sharing his knowledge and bringing me up to speed in areas where he had expertise—and I didn't. His emails were packed with information, as delightful to read as his articles, and conveyed the information in a wonderfully clear and unpretentious way."

Stuart Yaniger summed up the collective thoughts of the *audioXpress* family after learning that Ron Tipton had passed away when he said, "I was greatly saddened to hear of Ron's passing; although I celebrate the life he led and what he has left behind, I'll always regret not having had the opportunity to meet him face to face and share a drink or two over friendly conversation. Thanks, Ron, I'll always remember how you helped me become slightly less ignorant and have fun in doing so." 



Another speaker built by Ron, which he described as: This model TSMD1 speaker is half of a stereo pair. The closed box measures 23.75" tall by 9" wide by 12" deep. Construction is 3/4" thick MDF for sides, top, bottom, and back with a 3/4" thick 13-ply birch plywood front panel. The drivers include a pair of Dayton Audio #295-337 7" aluminum woofers and a single Dayton pt2 planar tweeter crossed over at 3 kHz with a passive second-order Linkwitz-Riley network. Response is flat to within 3 dB from 100 Hz to 20 kHz.

An In-Depth Exploration of *Electronics for Vinyl* (Part 2)

The Remaining Chapters and Final Thoughts

In the second half of this two-part article series, Gary Galo reviews the remaining chapters and shares his final thoughts on Douglas Self's recently published book *Electronics for Vinyl*.

By
Gary Galo

(United States)



Douglas Self is a well-known engineer and author specializing in audio equipment design. *Electronics for Vinyl* is Self's latest book on audio. In Part 1, I reviewed the first half of the book. Here are my thoughts on the remaining chapters.

Pre-RIAA EQ

In addition to RIAA, there's a short chapter titled "Archival and Non-Standard Equalization," an area where I've spent considerable time. Self makes a couple of historical errors in this chapter. He states that "Disc records appeared in 1901, first 10-inch, then 12-inch." Emile Berliner received his first patent for the flat-disc record in 1887, and was selling 7" records by 1895, along with a hand-

cranked Gramophone to play them. Self notes that acoustic records have a constant-velocity characteristic and were intended for mechanical reproducers, "such as the Orthophonic Victrola with its folded exponential horn." Acoustic records were, indeed, intended to be played on acoustic phonographs—there

were no other choices—and should be played flat when electronically reproduced using a magnetic cartridge (judicious post-phono preamp equalization is sometimes desirable to compensate for the limitations inherent in the recording process). But, the Orthophonic Victrola was an acoustic phonograph designed by Western Electric for the playback of electrical recordings. (Victor called the new electrically recorded discs "Orthophonic Recordings.") Prior to 1925, Victrolas and other acoustic phonographs did not have exponential horns. ("Victrola" was a trademark that applied only to internal-horn machines manufactured by the Victor Talking Machine Company; many incorrectly use the term generically to describe any acoustic phonograph.) Before Western Electric developed the math for the design of exponential horns, horn expansion rates were determined mostly by trial and error. Maxfield and Harrison, the inventors of electrical recording, described the exponential horn design in their 1926 Bell System Technical Journal paper. Self includes two topologies for early equalization curves—one being the design by D. Michael Shields, which I incorporated into the archival phono preamplifier that I described in

Electronics for Vinyl
by Douglas Self
New York: Routledge, Taylor & Francis
Group, 2018
328 pages
ISBN 978-1-138-70545-6,
www.routledge.com

Linear Audio. Though we're properly credited, my name is misspelled "Galow" several times.

Noise & Distortion

Self devotes a rather lengthy chapter to noise and distortion in moving magnet preamps. He points out that even relatively modest phono preamps will have a noise floor well below the noise inherent in the preamp itself. So, he asks: "Is it not a waste of time to strive for low electronic noise when the groove noise is much greater, and the contribution of electronic noise negligible? If obtaining a good electronic noise performance was difficult and expensive this argument would have more force, but it is simply not so. This chapter will show how to get within about 2 dB of the lowest noise physically possible using cheap opamps and a little ingenuity."

His treatment of the noise issue is exhaustive. He offers a chart with coil resistance and inductance for a number of popular moving-magnet cartridges, and includes several Grado moving-iron designs on this list. He provides noise modeling data for a wide range of moving magnet inputs using discrete field-effect transistors (FETs) and bipolar junction transistors (BJTs), plus a dozen or so popular op-amps. Only the discrete transistors beat his favorite devices, the 5532 and the 5534.

Other topics include noise in balanced phono inputs, weighting curves, and noise measurements in RIAA preamps. In the latter section, he stresses the importance of making noise measurements with a real cartridge providing the source impedance. Because of the coil inductance, the source impedance provided by the cartridge rises with frequency. At high frequencies, most of the input noise comes from the 47 k Ω cartridge load resistor rather than the cartridge itself. He credits Marcel van de Gevel for making him fully appreciate this issue. Self also covers noise when RIAA phono preamps are driven by moving-coil head amplifiers, which provide the phono preamp with a much lower source impedance than a phono cartridge.

Perhaps this chapter, more than any other, defines Self's design priorities. The chapter is over 21 pages long, yet discussion of distortion in moving magnet RIAA preamps is confined to little more than 1.5 pages. This discussion is limited to total harmonic distortion (THD) vs. frequency comparisons of the 5534A and the LM4562 op-amps used in RIAA phono preamps with 30 dB of gain at 1 kHz. But, offering nothing more than THD vs. frequency is way behind current audio measurement capabilities. Most people who work in the audio field, whether objectivist or subjectivist, know that the nature of harmonic distortion components greatly

affect the sound of audio circuits. Low-order THD will elicit very different subjective responses from listeners than high-order THD. Self's graphs show the differences between the 5534A and the LM4562 to be negligible at high frequencies. But, without measurements on the spectral content of the distortion, the usefulness of these graphs is limited.

In Chapter 22 of *Designing Audio Power Amplifiers*, Bob Cordell notes that "Single-number THD specifications are of limited use in characterizing the sound quality of an amplifier because they do not convey the nature of the distortion. For example, they do not distinguish between low-order and high-order distortions." He goes on to state that "THD is a much better indicator of amplifier performance when it is measured at high frequencies (THD-20) and a full spectral analysis is presented of the amplitudes of the individual harmonics."

Self avoids the issue of intermodulation distortion (IMD)—even the ancient Society of Motion Picture and Television Engineers (SMPTE) measurement, which mixes sine waves of 60 Hz and 7 kHz at a 4:1 ratio. Measuring IMD using two equal-amplitude tones, spaced 1 kHz apart, at a high output level, is very meaningful. The International Telephonic Consultative Committee (CCIF) method uses 19 kHz and 20 kHz test tones. And, a number of variations on it have been used by audio designers and manufacturers, including sliding the two tones across the entire audible spectrum. Cordell again stresses the importance of viewing the spectral content of the distortion: "Proper use of the CCIF IM test requires the use of a spectrum analyzer. This enables the odd-order products to be viewed." He also describes other distortion measurements, including Transient Intermodulation Distortion (TIM), Slewing-Induced Distortion (SID), and Dynamic Intermodulation Distortion (DIM). The vast array of measurements available with the test equipment manufactured by Audio Precision and others, including CCIF IMD measurements, go a long way toward quantifying the differences that listeners hear among amplifiers, whether power amps or IC op-amps. All of Self's graphs of THD, here and in the earlier chapter on op-amps, bear the Audio Precision logo, so he should have the capability of making any type of distortion measurements he chooses. John Atkinson provides an example of some of the measurements that can be made on phono preamplifiers in his *Stereophile* online review of the Sutherland Engineering Duo phono preamp.

Self notes that RIAA cartridge noise is a "complicated business" and this chapter certainly illustrates that point. He concludes that the 5534A is "cheapest and easiest way to get within 3 dB of the

theoretical best noise performance.” Although the dual 5523 is cheaper, it only gets you within 5 dB of the theoretical best, and he also notes that having two channels in the same package will likely increase crosstalk. “For these reasons, the 5534A is used in all the opamp designs in this book.” Though few would argue with his well-documented conclusions regarding noise, I find his priorities to be one-sided, especially considering his well-founded views on the noise levels found on LP records compared to those of even modest phono preamps. Most audio equipment designers realize that the design of any piece of audio equipment involves trade-offs. Self seems obsessed with noise at the expense of other, equally-important design criteria, avoiding the considerable array of distortion measurements available to designers.

Benchmark Media Systems is a very measurement-oriented company (I must point out that they also have a listening room in their facility). The company’s Vice President and Director of Engineering John Siau has expressed the view that good measurements equal good sound, and their equipment is highly-regarded by both measurement objectivists and subjective listeners. Why? In my view, it’s because they measure the right things (remember Heyser!), and don’t focus on one aspect of performance at the expense of others.

Discrete Designs

Self also includes a chapter on moving-magnet preamps using discrete circuitry, providing an interesting historical perspective on the early evolution of solid-state phono preamps. In the oldest, single-transistor designs the cartridge was effectively part of the RIAA feedback loop. One

designer, Jack Dinsdale, actually used the cartridge inductance as part of the low-frequency portion of the RIAA EQ. Unfortunately, every cartridge would yield a different RIAA response, and there was really no practical way to make it “universal.” Two-transistor circuits popular in the 1960s and early 1970s are also shown, and by the time we arrive at five and six transistor designs, the preamps begin to look like discrete op-amps with RIAA networks in the feedback path. Self notes the trend toward increasing power supply rail voltages in discrete preamps, including in the famous Technics SU-9600 stereo preamp, which ran the RIAA phono preamp on a 136 V supply rail.

The extra gain required for moving-coil cartridges is addressed in its own chapter. Self describes the characteristics of various low-output moving-coil cartridges, noting that impedances and output levels vary considerably from one model to the next. The chapter covers various fully active topologies for moving-coil head amplification, including the common technique of multiple, paralleled transistors where the gains add and the uncorrelated noise partially cancels. He offers a complete, working design using three paralleled bi-polar transistors followed by a 5532 gain stage. This is the only design in the book that incorporates a DC servo integrator—a TL072 is used as the servo amplifier around the 5532.

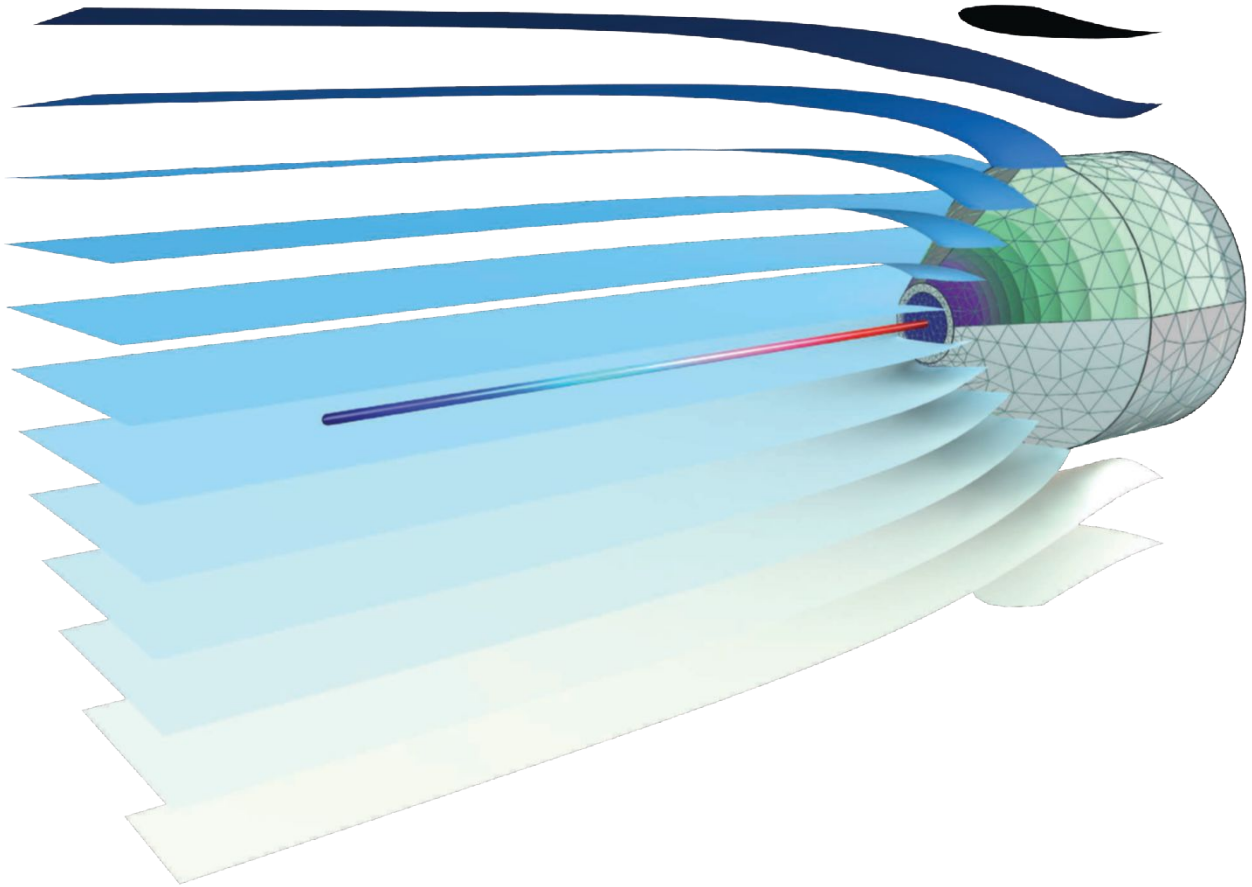
Self notes that a step-up transformer can be an elegant solution to the problem, with the caveat that transformers also can suffer from frequency-response problems and high cost. I should point out that the Jensen JT-44K-DX transformers that I use with my Denon DL-103R cartridge have -3 dB points of below 2 Hz and 200 kHz with that cartridge’s 14 Ω source impedance, combined with a deviation from linear phase of around 3° at 20 Hz. Self mentions, without giving the part numbers, that Jensen also makes transformers with multiple primary windings that can be connected in various series/parallel combinations to accommodate a wide variety of cartridge outputs and impedances. These are Jensen’s JT-346-AXT and JT-347-AXT transformers. These premium designs are an order of magnitude better in terms of low-frequency response and phase linearity than the JT-44K-DX, but at around \$200 US each, they aren’t cheap. Sonically, my JT-44K-DX transformers are very impressive, with excellent inner detail, with a smooth, open, airy treble and solid, well-defined bass. They also have double-nested shields and are silent in operation. I make these subjective statements based on actually listening to music through these transformers.

Self devotes two chapters to filtering information

About the Author

Gary Galo retired in 2014 after 38 years as Audio Engineer at The Crane School of Music, SUNY at Potsdam, NY. He now works as a volunteer in the Crane Recording Archive doing preservation, restoration, and digital transfer of vintage Crane recordings. He is also a Crane alumnus, having received a BM in Music Education in 1973 and an MA in Music History and Literature in 1974. Gary is a widely-published author with more than 300 articles and reviews on both musical and technical subjects, in over a dozen publications. Gary has been writing for *audioXpress* and its predecessors since the early 1980s. He has been an active member of the Association for Recorded Sound Collections (ARSC) since 1989, and a frequent recording and book reviewer for the *ARSC Journal*. He has given numerous presentations at ARSC annual conferences, many of which have been published in the *ARSC Journal*. He was the Sound Recording Review Editor of the *ARSC Journal* from 1995-2012, and co-chair of the ARSC Technical committee from 1996-2014. Gary has also published numerous book reviews in *Notes: Quarterly Journal of the Music Library Association*, written for the *Newsletter of the Wilhelm Furtwängler Society of America*, *Toccata: Journal of the Leopold Stokowski Society*, and he is the author of the “Loudspeaker” entry in *The Encyclopedia of Recorded Sound* in the US. He has also written several articles for *Linear Audio*. He is a member of the Audio Engineering Society, the Boston Audio Society, and the Société Wilhelm Furtwängler.

Hearing aids can't solve the cocktail party problem...yet.



Visualization of the total acoustic pressure field around and inside an elastic probe tube extension attached to a microphone case.

Many people are able to naturally solve the cocktail party problem without thinking much about it. Hearing aids are not there yet. Understanding how the human brain processes sound in loud environments can lead to advancements in hearing aid 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 hearing aid design.

comsol.blog/cocktail-party-problem

outside the audible spectrum. He makes a common error in describing information below the range of human hearing, calling it subsonic rather than infrasonic (subsonic refers to travelling below the speed of sound, and has nothing to do with frequency). The chapter is quite exhaustive, noting that the design of such filters gets complicated once you get beyond a second-order Butterworth characteristic. Regarding the phase shifts caused by analog filters, Self states that there are “mountains of evidence” that phase shifts are inaudible, especially at low frequencies. This view requires further clarification, and it’s important to distinguish between phase shift and phase distortion, the latter having been shown to be audible in double-blind testing. For a thorough explanation of this subject, and its audibility, I suggest the articles by Deane Jensen, as well as Stanley Lipshitz, Mark Pocock, and John Vanderkooy. The filter types covered by Self include multiple orders of Butterworth, Bessel, linear phase, and elliptical. He also discusses a passive, very-low frequency crossfeed technique, and phase correction of low-pass filters. Many formulas and tables are included to facilitate DIY design, and he offers many sample circuit designs all using 5532 and 5534 op-amps.

The chapter on filtering above the range of human hearing is correctly called “ultrasonic” rather than “supersonic” (these two terms rarely, if ever, cause confusion). Self offers sample circuits and design tables for Butterworth, Bessel, and linear-phase filters, all through sixth order. He recognizes the need for additional bandwidth above 20 kHz, so his designs have a -3 dB point at 50 kHz. Scratch filters are covered separately. Self notes that these filters usually operate in the 3-to-10-kHz range, and

offers several examples of variable-slope types. Both chapters on filtering contain multiple designs based on unity-gain followers using the 5532 which, again, seems inconsistent with his recognition of common-mode distortion problems with this op-amp.

The chapter on line outputs is dominated by discussion of various types of balanced outputs, both active and transformer-coupled. This is a bit surprising, given Self’s negative views on balanced inputs, expressed earlier in the book. There’s also a chapter on power supplies, all involving variations on fixed or variable 3-terminal IC regulators. He also devotes a short chapter to level indicators, offering a complete design for an LED bargraph meter with switchable peak/average response.

The Culmination

Self concludes with a chapter on practical designs, offering five complete preamplifiers with varying levels of complexity. Project 1 is an economy version using a single 5534A op-amp and RIAA network “C” made with single, off-the-shelf parts. Gain is 40 dB. Project 2 takes this concept and fine-tunes the RIAA response using paralleled capacitors and resistors. Project 3 is a 30 dB gain preamp using RIAA network A, and incorporates a third-order active high-pass filter as a variation on the IEC amendment. One of the resistor values is altered so the filter response deviates from maximum Butterworth flatness, though Self doesn’t show how it deviates. Project 4 is an update of Project 3, with the RIAA circuit changed to network C, and the infrasonic filter changed to a true third-order Butterworth. Project 5 uses network A, incorporating a switchable moving coil head amplifier, and increasing the complexity of the signal path by adding a switchable IEC rolloff to a second-order infrasonic filter. This is followed by a switchable gain stage providing up to 20 dB of additional gain. The filters and gain stage use LM4562 op-amps for reduced distortion. The caption for Project 5 is incorrect, noting a “switchable IEC amendment 3rd-order subsonic filter.” Only the first-order, IEC portion of the filter is switchable; the second-order filter is fixed.

Self supplies graphs of the RIAA accuracy for Project 1 and Project 2, and notes in the text that Project 3 and Project 4 are within ± 0.1 dB and ± 0.05 dB, respectively, from 20 Hz to 20 kHz. He notes improved accuracy for Project 5 but doesn’t offer any data. Project 4 uses four parallel capacitors each for C1 and C2. For Projects 3 and 5, the parallel capacitor count for C1 is increased to five. I much prefer my own Lipshitz implementations, which never require more than a single capacitor for C1. As noted earlier,

Resources

J. Atkinson, “Sutherland Engineering Duo Phono Preamplifier Measurements,” *Stereophile*, August, 2017, www.stereophile.com/content/sutherland-engineering-duo-phono-preamplifier-measurements

B. Cordell, “Distortion and its Measurement,” *Designing Audio Power Amplifiers*, New York: McGraw Hill, 2011.

G. Galo, “An Archival Phono Preamplifier,” *Linear Audio*, Volume 5, pp. 77-103.

D. Jensen, “High Frequency Phase Response Specifications – Useful or Misleading?” presented at the Audio Engineering Society (AES) 81st Convention, Los Angeles, CA, November 1986, AES Preprint 2398. www.aes.org/e-lib.

S. Lipshitz, M. Pocock, and John Vanderkooy, “On the Audibility of Midrange Phase Distortion in Audio Systems,” *Journal of the Audio Engineering Society*, Volume 30, No. 9, September 1982, www.aes.org/e-lib.

Self objected to the increased PC board real estate required for series resistors, but the number of parallel capacitors required in many these circuits take up far more space than a couple of series resistors. And, I should note that Project 3 also uses two series resistors to make up both R1 and R2.

Aside from RIAA accuracy, Self doesn't supply any performance data for these projects. Given the high priority he places on noise throughout this book, it's surprising that he doesn't supply any signal-to-noise specifications for the culminations of everything he has laid out in this rather extensive volume.

It would also be interesting to see if the use of series resistors in Project 3 degraded the noise performance compared to the parallel resistors he seems to prefer in order to reduce noise. Of course, THD and IMD figures would also be useful, with CCIF-style IMD being far more meaningful than extremely dated SMPTE method. Personally, I would like to see a complete set of distortion and noise measurements for at least one of these projects, comparing the 5532 and up-to-date op-amps such as the LME49860. This data would illustrate the

design trade-offs involved with the use of Self's favorite op-amp and those preferred by other designers.

Appendix 1 gives component values for the standard E3 through E96 series. Appendix 2 is a list of articles on phono preamp design published in *Linear Audio*, including Bob Cordell's "VinylTrak – A Full Featured MM/MC Phono Preamp" (V. 4), Stuart Yaniger's "The Equal Opportunity – A Balanced Moving Magnet Phono Stage" (Volumes 7 & 8), and my own "RIAA Equalization for Displacement-Sensitive Phono Cartridges" (Volume 11, with my name spelled correctly!).

Electronics for Vinyl is certainly one of the lengthiest volumes ever assembled on the subject. It contains a considerable amount of technical information and will surely be of interest to anyone involved in phono preamp design. That said, it represents the design philosophies of one engineer, and leaves a number of avenues unexplored. *Electronics for Vinyl* should not be considered the last word on this subject, but rather one source among many that phono preamp designers should have in their libraries. 

Proudly produced by Eastech Holding Limited Group of companies



Designed by a team of transducer's engineers with over **45 years of hand-crafted design** in Denmark.

Offering **Danish premium quality** ranges of transducers sizes from **0.75" tweeters to 15" woofers**.

We offer **OEM/ODM** choices for your transducers or systems engineering design needs.



E marketing@eastech.com

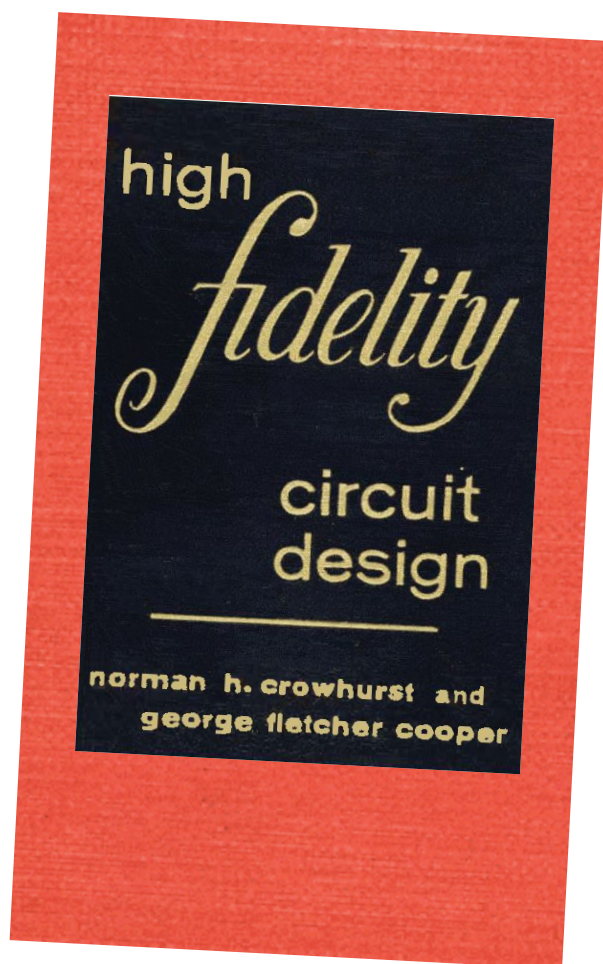
W www.eastech.com

High Fidelity Circuit Design by Crowhurst & Cooper

Review by

Morgan Jones

(England)



Like many readers, this reviewer initially flicks randomly through an unfamiliar book to gain an overall impression, and my immediate thought on this book was, "They've been reading too much Lewis Carroll" which, to be fair, the authors admit on page 31.

Quirky style aside, this is a reprint of a 1956 American book born from previous magazine articles aimed at an interested but not professional audience. The authors don't explicitly state their backgrounds, but the content and terminology strongly suggests cable telecommunications. This reviewer has a number of Norman H. Crowhurst books, but their style seems different, suggesting that this book was largely written by George F. Cooper.

High Fidelity Circuit Design

By Norman H. Crowhurst and George F. Cooper

Originally published by Gernsback Library, Inc.

Republished by: Linear Audio

Date: July 22, 2018

Pages: 304

www.lulu.com/shop/norman-h-crowhurst-and-george-f-cooper/high-fidelity-circuit-design/paperback/product-23729826.html

In judging a technical book more than 60 years after publication, we must understand the limitations of its time. Today, computer modelling is routine, and in the unlikely event that a standard modelling package is unable to perform our task, we resort to a dedicated mathematics package. Working further down the scale of sophistication, we write our equations into a spreadsheet and plot the results. At worst, we scribble on paper, punch calculator buttons, discover that 1.8 μF isn't available, recalculate to accommodate 2.2 μF , and the job is done. In the 1950s, even the simplest calculation was hard work (can you remember how to calculate square roots by hand?), so the book is littered with abacs and charts that were welcomed in their day but now look quaint.

Test equipment improved immeasurably after World War II, but few could afford even a 5 MHz oscilloscope, a signal generator was expensive, and audio test sets were unknown. Even plotting measured results was difficult because few stationers stocked (expensive) four-cycle log-lin graph paper. Thus, although the book's measurements were of a good standard for their time, we would be unlikely to repeat those techniques today and our presentation would be better.

Surprisingly, some electronics knowledge is

assumed, so the first five chapters are not about introducing triodes and pentodes, circuit blocks, and load lines, but about amplitude and phase analysis against frequency as a precursor to achieving stability when negative feedback is applied.

Chapter Six introduces phase splitters (concertina, see-saw, and long-tailed pair). Chapter Seven introduces power supplies, and interestingly notes that a choke-input supply benefits from an input capacitor “from 0.1 to 0.5 μF ” before the choke to ground.

Chapter Eight returns to passive components and considers attenuators, most particularly, the constant impedance attenuators that form the basis of cable equalizers. Chapter Nine briefly mentions cable equalizers before considering the simpler equalizers found within audio amplifiers.

Chapter Ten is devoted to the use of Lissajous figures on an analog oscilloscope, and can be ignored, as can the primitive treatment of loudspeakers in Chapter Eleven.

Chapter Twelve is entitled “advanced techniques” (sic capitalization) and begins by investigating pre-Baxandall tone controls from Leak and Quad. The (careful) use of positive feedback as a means of producing negative resistance is considered. The difference between small-signal and large-signal operation is discussed by considering 600 Ω lines. (Note that 600 Ω lines became obsolete decades ago, and because they needed to be driven from a precise 600 Ω , line amplifiers were actually designed to have near-zero output resistance before adding a 600 Ω series resistor, so the amplifier actually drove 1200 Ω —a fact missed in this book and elsewhere.)

Chapter Thirteen might be entitled “high-power amplifiers” but it’s about public address work and begins by considering public address loudspeakers on 70 V and 100 V lines, then considers audio distributed on 500 Ω lines (oddly, not 600 Ω). Next appears an early form of what would now be called a Voltage Controlled Amplifier (VCA). Finally, a parallel push-pull EL34 amplifier is presented, intended for 50 V line public address. Quality is not high, as the EL34 are operated as pentodes, but details of the transformer design are usefully given and the need for low-leakage inductance in a Class B amplifier is explained.

Chapter Fourteen considers design of oscillators and oscilloscope amplifiers, including a long-tailed pair having a pentode tail stabilized by neon reference.

The final chapter “improving old amplifiers” comprises seven pages and summarizes the book by covering adding negative feedback to an existing amplifier.

As can be seen, the contents are unconventional and a more accurate book title would have been, “Cable communications theory as applied to making stable negative feedback audio amplifiers.” Such a verbose title would not have sold many books. With only 303 pages (including index), coverage of the non-amplitude and phase analysis topics is distinctly skimpy, and it was annoying that very few diagrams were adjacent to their accompanying text.

Today, amplitude, phase, and stability analysis would be conducted in SPICE, rendering this book’s charts and calculators interesting mainly from a historical perspective, although the manual methods described provide an insight that blindly running a simulator does not. There are other bits and pieces of design information, but this book is best regarded as an interesting historical view on the technology and methods of the day. 

About the Author

Morgan Jones, well-known as the author of two standard texts on tube amplifiers (*Valve Amplifiers* and *Building Valve Amplifiers*), is a former BBC engineer. Morgan lives in a quiet suburb of London, England, where he currently works as a design engineer specializing in high sensitivity measurements. Besides electronic design, he is keen for anything on two wheels.

A New Part Introduction for 2016

1.8 nV Low Noise Dual P-Channel JFET

4pf Low Capacitance

Complement to N-Channel LSK489



The image shows three LSJ689 components: a TO71 package, an SOIC-8 package, and a SOT23-6 package. To the right is a graph titled 'LSJ689 Series' showing Noise Voltage (nV/√Hz) vs Frequency (Hz) on a log-log scale. The noise voltage is constant at 1.8 nV/√Hz across the frequency range from 10 Hz to 10,000 Hz. An arrow points to the curve with the text 'New LSJ689 Dual P-CH JFET'.

- Low Noise < 1.8nV
- Low Capacitance: 4pf max
- Low Offset: 20mV max
- Complement to Dual N-Channel JFET - LSK489
- Ideal for Differential Amplifier Applications when uses with the LSK489
- Ideal for Voltage Controlled Resistor Applications
- T071, SOIC-8, SOT23-6, ROHS Packages
- Contact the Factory or Visit the LIS Website for Data Sheets/Pricing/Samples

www.linearsystems.com 1-800-359-4023

Audio Archiving, Preservation, and Restoration at the Library of Congress

LIBRARY OF CONGRESS
PACKARD CAMPUS
AUDIO VISUAL CONSERVATION

From June 28-30, 2018, the Audio Engineering Society held its first International Conference on Audio Archiving, Preservation, and Restoration at the Library of Congress National Audio-Visual Conservation Center (NAVCC) Packard Campus in Culpeper, VA. Scott Dorsey attended the conference and he shares the experience offered by this exclusive look at the unique facilities and its treasures.

Photo 1: The Library of Congress is the largest library in the world, with millions of books, recordings, photographs, newspapers, maps, and manuscripts in its collections. The Library is the main research arm of the US Congress and the home of the U.S. Copyright Office.

By
Scott Dorsey
(United States)

The world's foremost experts on records are at the Library of Congress (see **Photo 1**). These are people who spend all day at work collecting records, repairing, and preserving records, and playing records. These are people who care about making sure that recordings are preserved as well as currently possible. As someone who has always dreamed of having a job where I could listen to records all day long, I was amazed to recently tour the Library of Congress' media archives in Culpeper, VA, courtesy of the Audio Engineering Society (AES).

The archives in Culpeper have a tremendous number of items in it. LP records, 78s, instantaneous acetate recordings, tapes in every format that can be imagined, and a huge amount of films dating back to the early days of Edison's experiments and going forward to the latest Star Wars movie that may be shipped to their vaults before it even appears in theaters. Whenever a record company issues a CD, it sends a reference copy to the Library

as proof of copyright. Just about every format you can imagine is in those vaults (see **Photo 2**).

Requesting Materials

Anyone can request access to materials in the archive, and if you were to go to the reference library in Washington, D.C., you could listen to a large body of already-digitized material, which is made accessible onsite. But if you want to listen to something that isn't already digitized, a request is sent down to Culpeper to transfer the material, and you will be informed when it's available for you to come and listen to it (see **Photo 3**).

It starts when somebody requests something from the archives. They are removed from cold storage, and then they might be catalogued. Some older collections have been carefully catalogued and indexed, and only when something is requested it this done. The items requested go to the cleaning room Intake section where they are carefully

prepared for transfer. This article is only about what happens to records. Maybe you want a CD, maybe you want a 1/4" tape. Procedures are also in place to get those to people who want them, but in this article I focus on records.

Prepping the Records

The Keith Monks record cleaning machines are used almost exclusively because they are so gentle on records (see **Photo 4**). These machines apply a fluid and vacuum it off but unlike most machines the vacuum is only a pinpoint spot, so instantaneous lacquer recordings won't be damaged. Lacquers are made of a base material (usually aluminum today, but in the past, materials as diverse as glass, steel, and cardboard have been used), coated by a thin layer of nitrocellulose lacquer and cut in real-time on a recording lathe. These might be intended for making stampers for pressing records or for direct playback like today's dub plates.

Shellac-based 78s come in. Modern LPs come in. Strange variants come in, and it's the job of the media people to know what they are, how to take care of them, and to make sure that both they and the metadata are preserved (see **Figure 1**). By



Photo 2: Whenever a record company issues a CD, it sends a reference copy to the Library of Congress as proof of copyright.



Photo 3: If you want to listen to something that isn't already digitized, a request is sent down to Culpepper to transfer the material.

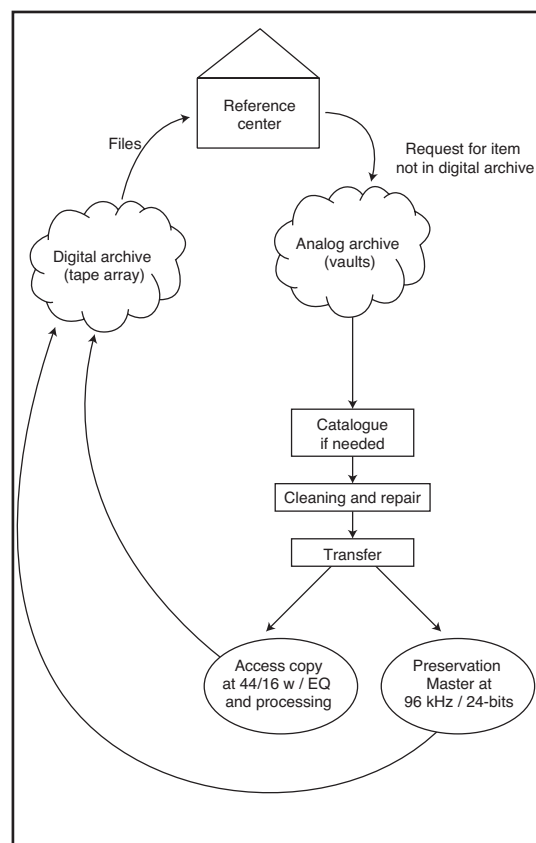


Figure 1: This diagram helps explain the process when it comes to accessing material from the Library of Congress.



Photo 4: The Keith Monks record cleaning machines are used almost exclusively because they are so gentle on records



Photo 5: Sometimes severely damaged materials like these acetate come in and need to be stabilized and made safe for transfer.

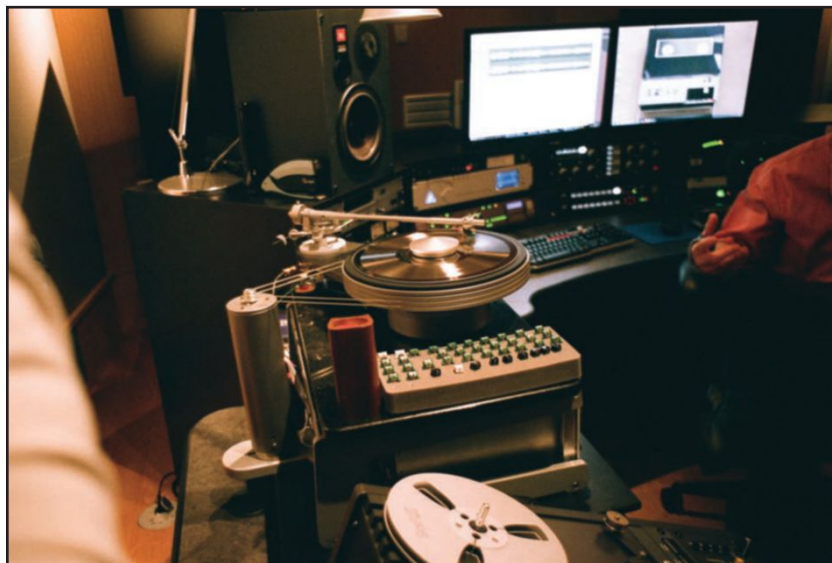


Photo 6: Here is the playback of a lacquer recording in a typical suite.

About the Author

Scott Dorsey has a degree in electrical engineering, during the pursuit of which he worked in the broadcast and recording industries. After several years working at a major studio, he took a job with a defense contractor. This left him time to do live concert recording for acoustical music and to design and build audio devices for personal use and on contract to several audio manufacturers and importers. Scott is a regular contributor to several audio magazines. He has been publishing equipment reviews and DIY projects since the mid-1980s. He is probably best known in the general audio community for his retrofit electronics designs in inexpensive Oktava, AKG, and Feilo microphones.

metadata we mean any information associated with them, from liner notes and labels to handwritten inserts and the information written in the lead-out. (Ever looked at the lead-out of a modern LP? You'll see a matrix number and the signature of the mastering engineer at the very least).

The cleaning room has a variety of machines available for different kinds of records, to avoid possible cross-contamination between disc types. Shellac 78s that may possibly shed debris shouldn't be put on the same machines as LPs, even if the operator is as careful as possible. Some records are cleaned by special technicians whose job is to prepare records. Some of them are cleaned by the transfer engineers.

Sometimes severely damaged materials like the acetate shown in **Photo 5** come in and need to be stabilized and made safe for transfer. Recordings that are in shockingly bad condition can often be made playable. There are experts on staff to deal with all manner of specialized problems, including things such as a sticky shed of tapes and acetate film base breakdown, but there is seemingly an army of experts just dealing with discs of various types. Once the record has been prepared as well as possible, it is taken to a transfer suite where the record is played back and digitized.

The Playback

Photo 6 shows a playback of a lacquer recording in a typical suite. Note the specially modified turntable that is able to handle a variety of speeds and the large display of styli of different sizes and shapes to accommodate earlier records and more worn records. Everything about the playback is specifically set up for the exact record being played back. If it's a 78 or an older transcription with wider grooves, or a worn disc, or a stamper, specially sized styli are used. If the record is a modern stereo LP, the engineer will carefully adjust the vertical tracking angle to get the best possible stereo separation. Every effort is made to get a clean transcription because anything lost in the transcription process is gone forever.

Transfers are all done at 96 kHz per second at 24 bits with some very high-end A/D converters. Listening to the engineers talking about converters is very much like listening to folks at an audiophile show. Nobody is ever completely happy with what they have and everybody has slightly different tastes, but everybody is working hard to push the state of the art.

We asked one of the engineers about direct stream digital (DSD), and apparently there is some internal discussion going on about the possible benefits of making a parallel DSD version of the

pulse code modulation (PCM) audio with direct DSD conversion, but it's not yet happened. These people care about sound, but they're intensely concerned about what transfer formats future generations will find most useful—and of course, we can argue with that, but we can't know for sure until it's too late.

The direct transfer is made with no processing at all, and it's marked as the Restoration Master. The 24-bit/96-kHz restoration masters are put directly into the digital archive with as short and as direct a signal path as possible. In many cases, it is even made without any de-emphasis to that effect. The majority of recordings that come through are older materials, some not in good condition, which may require some clean-up processing to be listenable. I know this seems like a terrible thing to audiophiles, who are worried about getting the purest copy of the original. The engineers at the Library are similarly worried, which is why that restoration master is retained.

The Access Copy

Often a lot of processing has to be done to make a listenable copy, and when that processing is done, an Access Copy is created. The access copy is only

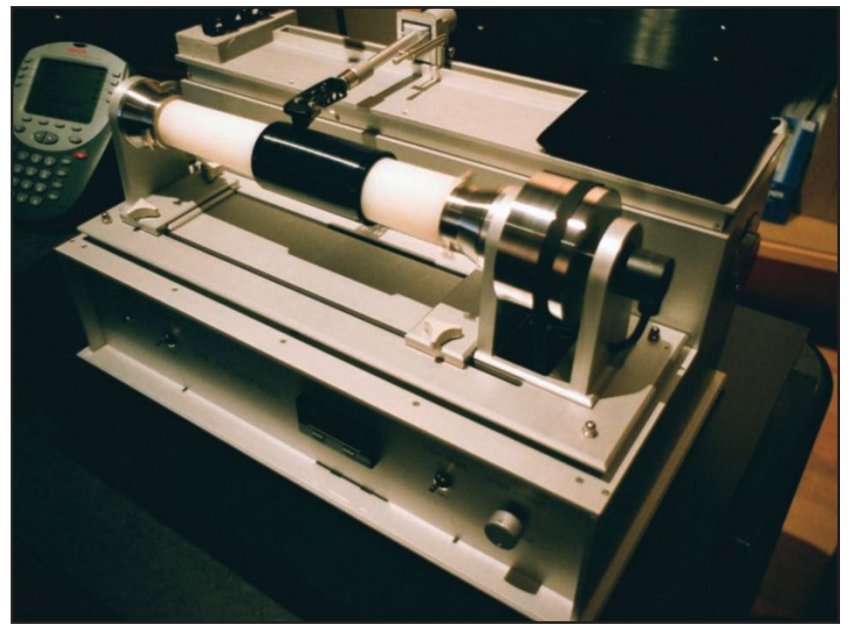


Photo 7: This Archeophone machine is a modern electronically controlled playback device for Edison cylinders.

at 16 bit/44.1 kHz, and it may have light processing or extensive processing to improve listenability.

The engineers in the transfer suite have a huge array of different software available for this purpose, from Cedar to Izotope. But, if some engineer should

GETTING THE MOST OUT OF
VACUUM TUBES

By Robert B. Tomer

Vacuum Tubes—The Important 4/5ths of Tube Circuits
Over 80% of all tube equipment defects result, directly or indirectly, from tube failures. Why do tubes fail? What can be done to prevent them from failing before their time? How can you determine whether a tube is good or bad, or how well and how long it will work in a given circuit? Should tubes be replaced periodically, whether they've failed or not . . . or should they be tested every so often, and replaced if indications show them to be below par? This book supplies the answers to these profound questions...plus many, many more.

Like Watching a Play From Backstage
Author "Bud" Tomer, an international authority on tubes, is well-known for his candid "Tech Tip" bulletins and his slide-film presentations to service, military and engineering groups. Within this book you'll find a virtual encyclopedia on vacuum tubes—yet the author has not delved into mathematical details that would be of interest to only a few tube engineers. Everything is focused toward the interest of those who use tubes—either for replacement purposes or in original equipment designs.

In his refreshing, down-to-earth style, Mr. Tomer sheds light on the mysteries of tube testing. "Why so many tube types?" "What About Tube Testers?", etc.

Other Interesting and Informative Topics:

- 10 reasons why tubes fail
- Predicting tube life
- Selected and preferred tube types
- Lengthening tube life

Anyone who has anything at all to do with tube equipment design, or proper use—should read this book. It is a must for anyone who designs, tests, maintains and maintains consumer goods such as radio, TV, and other electronic equipment, and for those involved in production and manufacturing processes—or even for hobbyists and amateur enthusiasts.

You'll also learn about:

- Catastrophic failures—what they are, when and how they happen
- Degenerative failures—what happens to tubes over time
- Subjective failures—why a seemingly normal tube may not perform as expected
- Characteristic variables—differences that make one tube type better than another
- Special-purpose tubes—considerations for filament and thyratron types.

Order your copy at cc-webshop.com

**PRECISION
ACOUSTICAL
MEASUREMENT
AND ANALYSIS**

- Transducer Optimization
- EASE Data and Modeling
- Polar Plots and Directivity Balloons

KLIPPEL

*Sales & Support
North American Representative*

**NOW OFFERING KLIPPEL
CALIBRATION SERVICES IN THE U.S.**

WARKWYN

Minneapolis, MN
612.759.4042
info@warkwyn.com
www.warkwyn.com

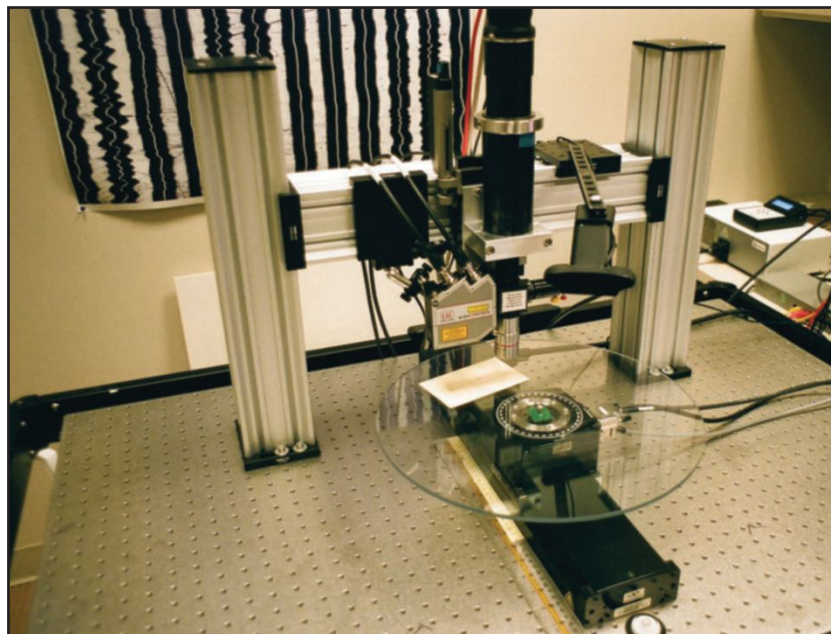


Photo 8: IRENE is a 3D optical scanning system that relies on arrival time differences to determine the shape of a record.



Photo 9: Peter Alyea of the Library of Congress demonstrates the IRENE optical imaging system.

Resources

Audio Engineering Society | www.aes.org

Library of Congress National Audio-Visual Conservation Center (NAVCC)
Packard Campus | www.loc.gov/avconservation

decide that an old Packburn would be the best tool for the job, they can likely find one. One of the finest things about the Library is that it is very agnostic about tools and will purchase whatever software or hardware a given engineer finds necessary to do a job, so all of the editing suites are equipped differently, all for specific engineers. This enables comparisons between different software and equipment and it also means that whatever system is most appropriate to solve a given sonic problem is probably available somewhere in the building. And, if the system doesn't exist yet, then the restoration master is sitting online in digital storage, ready to be reprocessed into a better-sounding access copy.

When I say that the Library has very specialized equipment to deal with just about anything that should come up, I am not exaggerating. In **Photo 7**, we can see an Archeophone machine, a modern electronically controlled playback device for Edison cylinders. Edison records were very popular for quite a while and were capable of surprisingly high fidelity, and the Edison recorders were portable enough to be used for dictation and for recording speeches and live performances in the field. So, there are a lot of very interesting recordings from the early part of this century. A lot of them aren't of interest to audiophiles, but there is something amazing about listening to Teddy Roosevelt's own voice.

For recordings that are too damaged to play, there is an experimental system called IRENE, which is a 3D optical scanning system that relies on arrival time differences to determine the shape of a record (see **Photo 8**). Then a digital system can simulate the path of a stylus down the groove to reconstruct the signal.

Constructed in conjunction with Lawrence Berkeley National Laboratory, IRENE is still an experimental system. It sees pieces of dirt that a stylus would brush away, so the playback is a bit noisy. But it allows very careful manual inspection of the groove, it allows some tricks to be performed on worn records, and on damaged records with cracks, chips, and missing sections it will play through without having to play each rotation of the groove manually at low speed and reconstruct from the pieces. It's a tool for the future but one that is already finding some use (see **Photo 9**).

The Transcription Files

After all this, the two transcription files—the access copy and the restoration master—are put on a giant disk/tape farm that can be accessed from the reference center in Washington. A short time, sometimes just a week or two after requesting material, the original requester can

come back and listen to the digital transfer of the recording for which they have asked.

In addition to making materials available for listening onsite, requesters who can prove that an item is in the public domain or that they have permission from the rights holder of the material can request digital copies to be taken offsite.

A very tiny portion of material is currently available for listening on the National Jukebox, at <http://loc.gov/jukebox>, although at this time only a very small number of pre-1923 acoustic recordings from RCA Victor are currently available. This is so sad, given the vast amount of already catalogued and digitized material that is known to be in the public domain, which are in the Library archives. Hopefully more material will become available soon.

Much of our country's history is preserved on Edison recordings and on radio transcription discs, such as the 1948 political speech by then-South Carolina Governor James Strom Thurmond, Sr. shown in **Photo 10**. So much of the history of music has been preserved on 78s and LPs, but more than just music. Records tell the story of the nation, the story of industries, they help us to understand how we got to where we are and where we are going. 🇺🇸



Photo 10: Instantaneous transcription discs of radio broadcasts such as this 1948 political speech by then-South Carolina Governor James Strom Thurmond, Sr. (December 5, 1902–June 26, 2003) were made as single copies in the days before audio tape was available for radio production and deposited with the Library of Congress and make an important contribution to our understanding of 20th-century history.

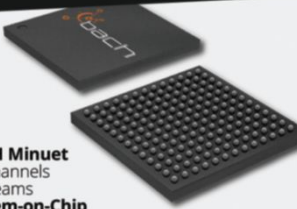




BACH Virtual Sound Card
Audio on Linux
256 Channels
32 Streams



BACH openModule
≤ 512 Channels
≤ 64 Streams
3 x GigE
Ultimate Capacity. Customizable.



BACH Minuet
16 Channels
8 Streams
System-on-Chip

Quickly give your products the freedom and power to perform across markets with our embedded live audio networking ST2110 & AES67 modules, chips and software solutions.

BACH™ offers a comprehensive set of embedded audio networking solutions, enabling your products to play in multiple audio networking environments, including AES67 and ST2110 complete with control from Dante/SAP, RAVENNA, Livewire+, NMOS, and DashBoard. Our BACH™ platform also optionally offers Ethernet AVB networking. BACH's broad technology spectrum enables your single product investment to perform across markets.



BACH Liberty Module
64 Channels
8 Streams

PIN COMPATIBLE TO DANTE BROOKLYN II





www.rossvideo.com/oem-developer-solutions



ROSS
LIVING LIVE!



NO MONEY
NO BETTER

SPEND MOSTEST
FOR BESTEST SOUND



SCHIIT

WWW.SCHIIT.COM

 FIRE

OBHEY

DO NOT QUESTION AUTHORITY

Isn't it time for a change?
Discover our affordable,
US-made audio components,

Modulation Distortion in Loudspeakers and Its Detectability (Part 3)

Experimentation to Find the Threshold of Detectability of Doppler Distortion

In the two previous articles, we have discussed the detectability of Doppler distortion, examining a few theoretical and experimental studies, plus a variety of opinions. Now we will present the results of a small human study, and compare its results with those of previous researchers. This study was reported at the May 28, 1971, meeting of the North Carolina Regional Chapter of the Acoustical Society of America.

By
Richard Honeycutt
(United States)

The experiments discussed in this report used five observers. Each observer was placed 3 m from the front of the loudspeaker, in a 2000-Hz, 70-dB sound field. A 60-Hz signal was then introduced and the observer indicated at what level of the 60-Hz wave he first began to detect modulation (warbling) of the 2000-Hz wave. The test was repeated with

the observer placed in a 70-dB sound field beside the speaker. (This required a distance of about 1' from the speaker). The arrangement is shown in **Figure 1**.

From the averages of the two sets of values thus obtained, a correction factor of 3.85 dB was derived in the manner described in my previous *audioXpress* Sound Control article (October 2018). These experiments yielded a corrected critical level (for a 12" speaker, a 60-Hz modulating frequency, a 2000-Hz high frequency, and a speaker-to-listener distance of 3 m) of 61.65 dB. There was a precision of 1.2 dB, and an expected error of 1 dB. The disagreement with theory is 19.4 dB.

The experimentally determined critical cone velocity to produce the threshold SPL for various sizes of speakers is plotted vs. modulating frequency shown in **Figure 2**.

Initial Experiment

This small initial experiment indicates that, for a pure tone, the perceptibility of modulation distortion is much greater than is the perceptibility of unidirectional, long-term pitch change. (It may

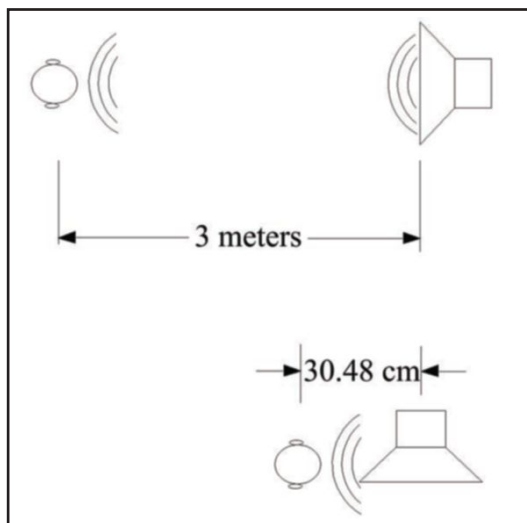


Figure 1: The experimental arrangement

be noted at this point that if we define percent distortion as the ratio of undesired, distortion-produced frequency amplitudes to desired signal, the theory predicts a threshold of perceptibility of about 5.25% distortion; whereas our experiments show the threshold to be 0.2%.)

Therefore, the theory outlined in the previous Sound Control article (*audioXpress*, October 2018) predicts far less sensitivity of the ear to this distortion than actually exists, and the theory is in need of repair. The repair may be effected by our simply changing the value of a to 0.000182. It must be remembered, of course, that most music consists of complex waveforms, and that the perceptibility of modulation distortion of these waveforms may very possibly be considerably less than pure tones. Another unknown is the dependence of the modulation distortion detection threshold upon the modulating frequency.

Threshold Studies

As mentioned in my previous articles, the similarity of Doppler distortion (Dd) to flutter in tape recorders was noticed early by Paul Klipsch and others. More recently, studies have been performed to establish a threshold at which phase jitter in digital clocks becomes perceptible in digital audio. **Figure 3** shows the NAB standard limit on flutter in a 15 IPS tape recorder, plotted on the same graph as the phase jitter limits proposed by W.I. Manson of the BBC in the early 1970s.

The flutter has been translated from the usual units—%RMS flutter—to the units used to measure phase jitter—RMS jitter in seconds. The latter can be transformed to the instantaneous frequency change Δf by dividing the RMS phase jitter in seconds by the period of the modulated wave. For example, if we are examining the detectability of Dd of a 2-kHz wave by a 60-Hz wave, we see from the graph that the RMS phase jitter is about $1.46\mu\text{s}$. Dividing this by the period of the 2-kHz modulated wave ($500\mu\text{s}$), we have: $\% \Delta f = 1.46\mu\text{s} / 500\mu\text{s} = 0.292\%$. Notice that the $\% \Delta f$ is not the same as the $\% \text{Dd}$ based upon the amplitudes of the modulation sidebands and the “carrier” amplitude. Unfortunately, the phase jitter studies did not examine detectability of modulation of a 2-kHz wave by a 60-Hz wave, which is arguably the most useful combination of frequencies at which to look.

In order to compare many of the Dd thresholds determined over the years, I have converted the results to the SPL produced by a 12” speaker reproducing a 60-Hz wave at the threshold level

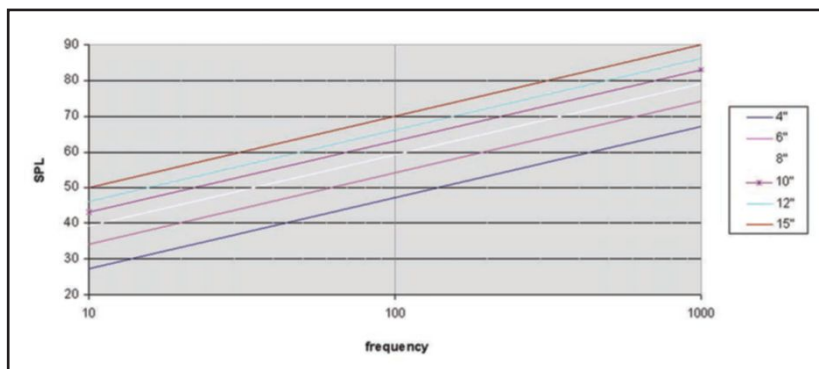


Figure 2: The experimental SPL thresholds for various speaker sizes, for detectable Doppler distortion

that just produces detectable Dd. The bar graph in **Figure 4** shows this comparison. Although the work of these researchers was described in previous Sound Control articles, it is useful to present some comments for each one (see **Table 1**).

For speakers of diameters other than 12”, the thresholds will be different:

- 15”—3.8 dB higher
- 8”—6.9 dB lower
- 6”—12 dB lower
- 4”—19 dB lower

Next, let us see what all this means in terms of the high-fidelity listener. H. K. Sivian, S. D. Dunn, and L. K. White have compiled graphs relating the overall time-averaged (wideband) sound pressure level (SPL) produced by various

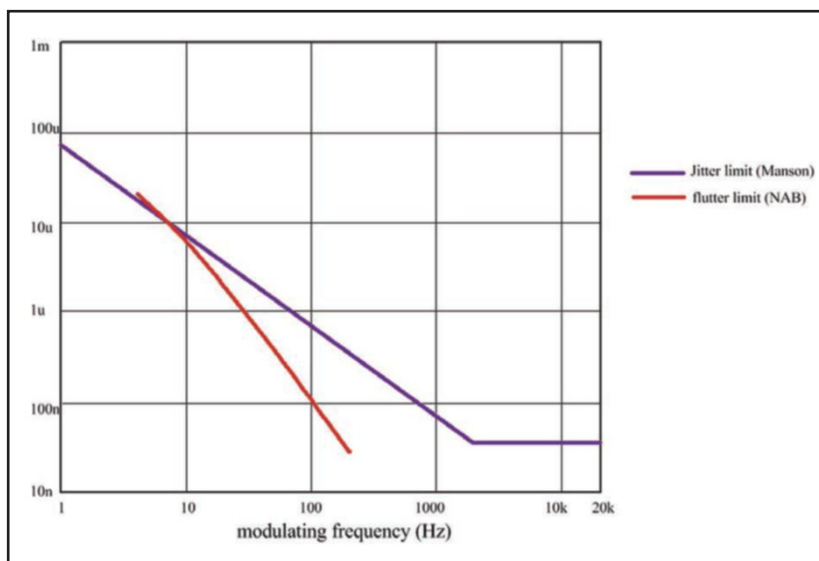


Figure 3: The detectability thresholds for phase jitter and flutter

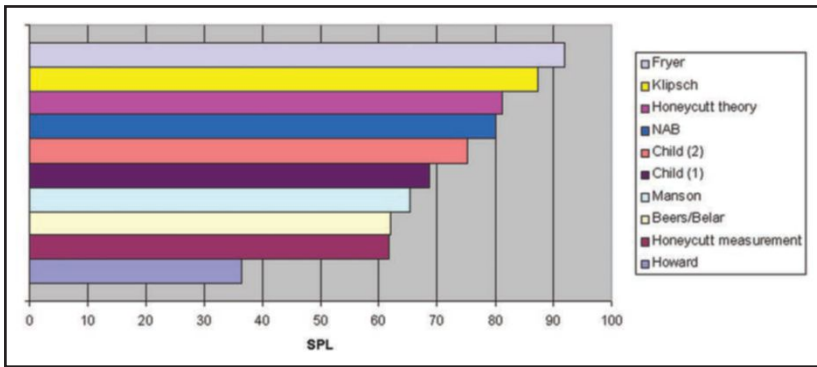


Figure 4: The experimental results for Doppler distortion detectability thresholds

instruments and combinations of instruments to the average pressure in given frequency bands. (These are called critical bands, and the audible range of frequencies is spanned by 20 of these bands. The bands were selected by reference to perceived pitch. The two lower bands are 0-90 Hz and 90-180 Hz. For further information, see the Hewlett-Packard Application Note 100 and Harry F. Olson's *Acoustical Engineering*) Further, Sivian, et al., related the peak pressure level in the various critical bands to the overall average pressure level.

If we choose examples from this data, we may obtain some idea as to the likelihood of perceptible Dd for various sizes of speakers, if music is being reproduced at realistic concert levels. We will choose:

- A 75-piece orchestra (overall average SPL = 87.2 dB; distance to observer = 15')
- A 15-piece orchestra (stage or dance band)

(overall average SPL = 92 dB; distance to observer = 10')

- A pipe organ (overall average SPL = 80, 4 dB; distance to observer = 12')

Figure 5 shows the average and peak concert levels in the 0-to-90-Hz range for organ, a 25-piece orchestra, and a 75-piece orchestra. Clearly, for "live" listening levels of these music sources, all the threshold levels shown in Figure 4 are higher, except for Howard's. Thus no perceptible Dd will occur.

Perceptible Dd will result whenever a speaker reproduces any level which exceeds the critical level for that frequency, provided that the speaker is simultaneously reproducing a higher frequency. It is clear that occasional peaks will drive a 12" speaker or smaller into perceptible Dd, using any except the two highest thresholds graphed in Figure 4. (Compression and tastes in listening levels are huge unknowns in predicting the likelihood of perceptible Dd.)

Conclusions

It should immediately be obvious that the larger speakers require larger low-frequency peaks to produce detectable Dd than do the smaller ones. Even when signals are such that the larger speakers do produce Dd, they produce less of it than do the smaller speakers. Perhaps somewhat less obvious is the fact that pipe organ music, which is most susceptible to Dd due to the high incidence of low-frequency peaks, also often presents the best opportunity for detecting this distortion due to its tendency to contain relatively pure tones, which are held for fairly long periods of time. These comments could also pertain to rock music, which often contains organ notes and fuzz-box/limiter-held guitar notes that are rather extended, coupled with a very strong bass.

Now we are still faced with one question concerning Dd: How much is the perceptibility of Dd affected by: (a) the masking effect of the extremely complex musical waveforms, (b) the rapid changing of musical sounds, and (c) the relative shortness (in much, but significantly, not all music) of the low-frequency peaks that are responsible for the Doppler modulation, affect the perceptibility of the Doppler distortion?

Here, multiple tests with many human observers are indicated if accurate answers are to be obtained. In the present absence of these, I will offer some subjective opinions. First let us note that the effect is perceivable for very low

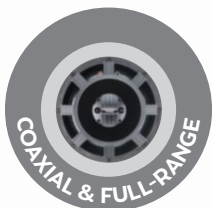
Researcher	Theory
P. A. Fryer	Test frequencies: 1 kHz/40 Hz
Paul Klipsch	Tested with music; level is not the detectability threshold, but rather a level at which Dd was described as "unpleasant" and "extremely irritating"
Richard Honeycutt (Theory)	Test frequencies: 2 kHz/60 Hz
NAB	Test frequencies: 2 kHz/60 Hz
Roy Childs (2)	Test frequencies: 1 kHz/40 Hz
Roy Childs (1)	Test frequencies: 1 kHz/40 Hz
W. I. Manson	Piano and glockenspiel/60 Hz
G. L. Beers/H. Belar	Test frequencies: 5 kHz/40 Hz? (paper was not specific)
Richard Honeycutt (measurement)	Test frequencies: 2 kHz/60 Hz
Keith Howard	Flute tone, 100 Hz

Table 1: The test signals used by the various researchers

DESIGNATED DRIVER

Leading Pro Audio brands choose Celestion

Innovation, performance and reliability make Celestion the drivers of choice for a growing number of big-name PA brands. So when you need superior quality compression drivers or professional loudspeakers, check out about the speakers the pros use.



celestion.com

CELESTION

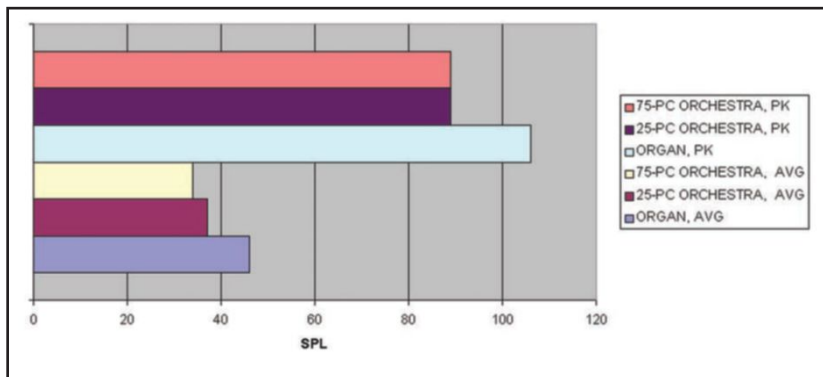


Figure 5: The average and peak concert levels for organ and orchestras in the 0-to-90-Hz range

modulating frequencies, as vocal vibrato is an example of perceptible frequency modulation. It is, I think, quite possible that the “roughness” noticeable in playback of essentially all violin recordings, except on the very finest (notably electrostatic) speaker systems, as contrasted with the striking smoothness of violins in live performance, may be due at least in part to Dd.

And, Paul Klipsch pointed out that the confusion of frequencies that results after the introduction of all the various inharmonic modulation distortion components “loses the midrange instruments in a sea of mud.” Perhaps modulation distortion is responsible for at least part of that hard-to-pinpoint but still undeniable difference in midrange performance between a “clean” and a “muddy” speaker.

Resources

R. F. Allison, “Doppler Distortion Again,” *Audio*, March, 1971.

R. V. Childs, “Doppler Distortion in Loudspeakers,” *Audio*, Volume 54, No. 8, August, 1970.

P. W. Klipsch, “Loudspeaker Performance,” *Wireless World*, February 1970.

———, “The Mud Factor,” *Audio*, October, 1970.

V. O. Knudson, “The Sensibility of the Ear to Small Differences in Intensity and Frequency,” *Physics Review*, XXI, 1923.

H. F. Olson, *Acoustical Engineering*, Van Nostrand and Co., Princeton, 1947.

Sivian, Dunn, and White, “Absolute Amplitudes and Spectra of Certain Musical Instruments and Orchestras,” *Journal of the Acoustical Society of America*, Volume II, 1930.

E. G. Shower and R. Biddulph: “Differential Pitch Sensitivity of the Ear,” *Journal of the Acoustical Society of America*, Volume 3, 1931.

H. Van Hessen, “Doppler Anyone?” *Audio*, April, 1971.

Several final considerations are worthy of mention. The calculations and experiments above have been based upon a listener in what was chosen to approximate a typical living room situation: the reference point for calculating and measuring SPL for a given cone velocity was 3 m from the speaker. Further, these calculations have assumed that listening is done at a level comparable to that of the original performance. Also, the statistics on orchestral frequency-loudness distribution were averages. Therefore, the use to which any given system is to be put must be kept in mind.

If (1) much 1812 Overture-type music is to be played (tympani, cannon, tubas, and trombones produce strong signals of relatively long duration in the 90-to-180-Hz band); (2) music is to be played at louder than realistic levels (or at realistic levels for rock music); or (3) the listeners are to be farther than 3 m from the speakers (as in a theater or auditorium), the Doppler distortion could be not only perceptible, but downright objectionable. Methods of avoiding Dd then include:

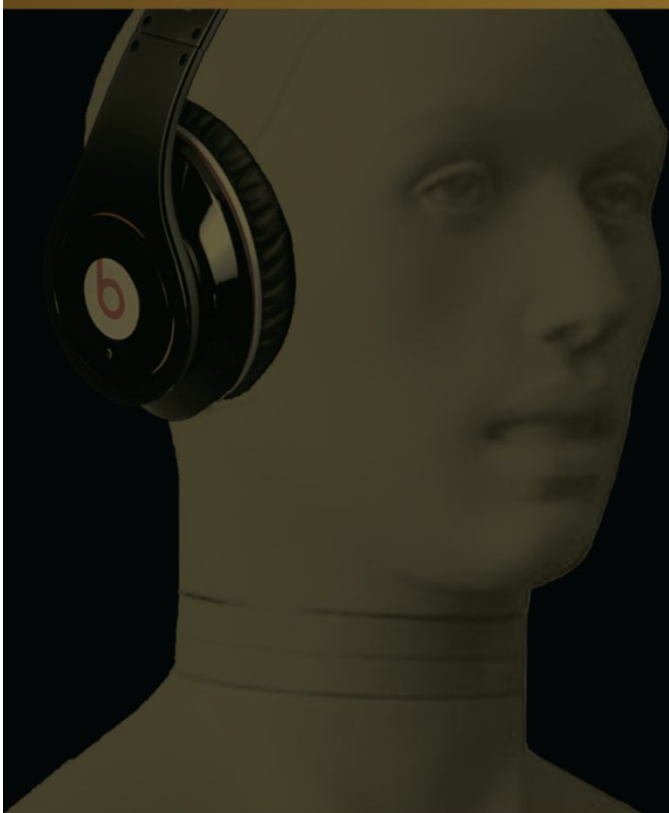
- Installing electrostatic speakers
- Using larger speakers (reduces Dd)
- Using multiple-speaker systems so that mid and high frequencies are not reproduced by a woofer (a crossover around 200 Hz would eliminate most Dd of mid and high frequencies, and theory predicts that much greater Dd of low frequencies is necessary for perceptibility),
- Using multiple woofers to increase cone area, and thus decrease excursion and cone velocity
- Using horns to increase the coupling efficiency of speaker to air, thus producing the effect of a larger cone, and reducing necessary excursion for a given SPL.

And one final note: The listening tests were not done in an anechoic environment. Thus, the listeners were presented with less difference between the “clean” and the “Dd” sound than would have been the case in an anechoic chamber, since the “Dd” signal would have reflected back to the listeners, when the listeners were beside the speaker. Although the reflected Dd would have been at a reduced level, this listener position should ideally have been a Dd-free zone, and wasn’t.

Engineering Consulting for the Audio Community

Celebrating
35 YEARS
1983-2018

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

USB Oscilloscopes

A Look at the LabNation SmartScope



By
Ethan Winer
(United States)

When Ethan Winer found himself in need of an oscilloscope, he turned to the Internet for advice and ended up selecting LabNation’s SmartScope. Described by its makers as “the oscilloscope of the new generation” the device, originally funded by a Kickstarter campaign, is targeted at makers and hobbyists, as well as professional engineers. SmartScope combines a dual-channel 30 MHz oscilloscope with a logic analyzer and an arbitrary waveform generator into an open architecture USB instrument that interfaces with most common platforms.

My desire to pursue a career in audio and music began as a teenager in the mid-1960s. I enjoyed playing the violin in elementary school, but my passion really took off when I started playing the electric guitar. After several years of practice I noticed that some days I could play a difficult passage perfectly, but not so well other days—especially when I wanted to show off for a friend! That drove my desire to get a tape recorder, and eventually it also drove my lifetime goal of learning all about audio and recording, developing a keen interest in analog circuit design along the way.

Knowing how to use audio gear effectively is great fun, but for me understanding how audio devices actually work is even more fun. I never got a degree or even went to school for electronics, but I worked as a technician with many excellent “real” engineers. I asked them endless questions, and learned a lot. By the 1970s, I had designed and built equalizers, compressors, analog synthesizers, and even an entire 16-channel recording and mixing console.

At one of my technician jobs, I had the good fortune to befriend Leo Taylor, who later was a senior tech at a nearby Hewlett-Packard (HP) repair facility. Leo had all the latest HP test gear, which we used to design several analog synths. Leo’s test bench included state-of-the-art oscilloscopes, signal generators, real-time audio analyzers, and even a distortion analyzer. But I never bought any

professional test gear myself. Between Leo and whatever electronics company for which I currently worked, I always had access to everything I needed. Eventually I developed other interests—by 1980, I built and was running a large commercial recording studio, then a software company, then I started playing the cello and composing classical music, plus other non-electronic projects.

Now, fast forward to the present when I started developing a new and unique type of audio testing device. Hopefully, that will be the subject of a future *audioXpress* article! I still have engineer friends with workshops and real test gear, but eventually I got tired of driving half an hour every time I wanted to troubleshoot the latest developments of my device.

I have a good digital multimeter, but it was time to buy an oscilloscope. I’ve done a lot of signal analysis just recording into Sound Forge—the audio editor program I use. You can zoom in to see the waveforms at any level of detail, and even show the spectrum as an FFT graph. You can also buy or build accessories that turn a USB sound card into an oscilloscope suitable for audio testing. But even with a sound card set to sample at 192 kHz, there’s no way to see if radio interference is getting into your circuit, or to tell if an op-amp is oscillating at megahertz frequencies.

Choosing an Oscilloscope

I've seen ads for USB oscilloscopes with megahertz bandwidths, so on my Facebook page, I asked for recommendations of two-channel USB oscilloscopes costing no more than a few hundred dollars. Many people posted links to their favorite models, and several others insisted that all USB oscilloscopes are awful and I'd never be happy unless I bought a real hardware oscilloscope. But the available space on my desk is barely more than one foot square, and I also wanted a signal generator. One of my local Facebook friends offered to lend me her LabNation SmartScope, and that turned out to be perfect for my needs. Besides doing everything I need and then some, it's only slightly larger than a deck of cards (see **Photo 1**). So I bought my own and have used it with great success for several months.

LabNation SmartScope

So what exactly do you give up with a USB oscilloscope compared to a standalone hardware model? As it turns out, not much! One limitation is the input A/D converter is only 8 bits, so at the highest sensitivity of 20 mV per major division you can see little steps when viewing small signals. (However, affordable digital oscilloscopes, even from "big name" companies such as Tektronix, are also only 8 bits.) Another limitation for me is that the documentation is online only, and not quite as complete or as detailed as I'd like. Though it's certainly adequate, and the company provides an online support forum where its tech staff answers questions promptly.

As far as I'm concerned the SmartScope's features totally outweigh its limitations, especially for only \$230. You get a real two-channel 30 MHz storage oscilloscope with capable and intuitive software. One big feature for me is that you see the waves on your computer screen, which in my case is a 24" LCD. Even at the \$1,000 price point, hardware oscilloscope screens are much smaller. The SmartScope also includes two signal generators—analog and digital—and an eight-channel logic analyzer (see **Photo 2**). However, the analog output is also only 8 bits, so it's not suitable for testing distortion in audio devices. All of the connecting wires needed are provided including a USB cable, two standard oscilloscope probes (1x or 10x switchable), plus connecting wires and "grabber" clips for the analog and digital signal generators and the logic analyzer.

The SmartScope Software

As for software, SmartScope is available in versions for Windows, OSX, Linux, Android, and iOS. The main screen is shown in **Figure 1**, and includes



Photo 1: The LabNation SmartScope ships with the two probes, a 24" USB A/Mini-B interface cable, a 10 conductor ribbon cable, nine grabbers, and cables for eight-channel digital signal analysis.

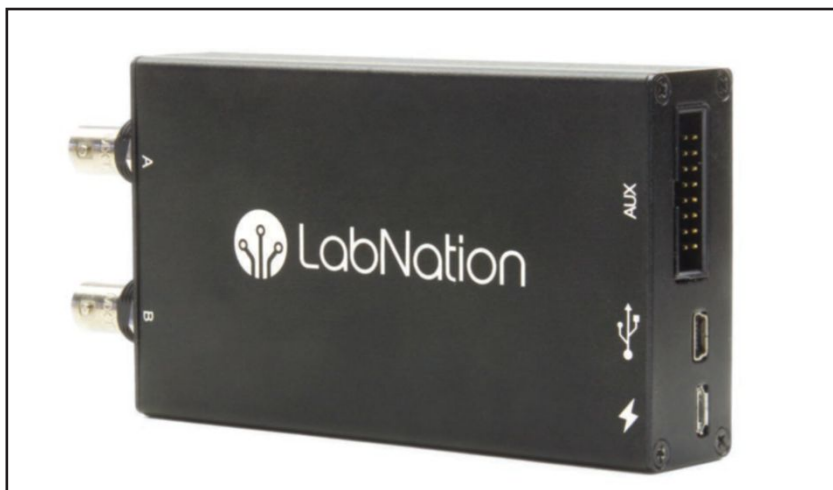


Photo 2: The non-BNC end of the SmartScope has a 16-pin auxiliary port, a Mini-B USB platform interface, and a Micro-B USB external power connector.



Figure 1: The bold legends at left in the SmartScope software expand to show the various options within each section.

extra features such as record to disk, Fast Fourier Transform (FFT) analysis, a frequency counter, and an RMS voltmeter. You can also place a pair of vertical cursors (not shown) some distance apart on the screen, and the software displays either the time span between them or the equivalent frequency for that period. Pressing Stop freezes the display and lets you zoom in on the displayed waves.



Figure 2: When you click the little ruler in the lower right, a pane opens letting you select which information to display in the Measurements Box at bottom. Here, I set it to show the frequency and RMS voltage for input Channel A, and the frequency for Channel B. But all of the other options shown are available for continuous monitoring.



Figure 3: For each input channel you can specify AC or DC coupling, trigger direction (up, down, or both), probe sensitivity, and other parameters.

About the Author

Ethan Winer has been an audio engineer and professional musician for more than 45 years. His “Cello Rondo” music video has received nearly 2 million views on YouTube and other websites, and his book *The Audio Expert* published by Focal Press, now in its second edition, is available at amazon.com and Ethan’s own website ethanwiner.com.

The software enables internal or external triggering in either direction (or both), and can store up to 4 million samples per channel in memory. Other options include setting the UI size, color scheme, and waveform trace thickness, to accommodate hardware platforms having large or small screens. The software also works in Demo mode so you can download it to see how it works before you buy the SmartScope. Though if you buy it and aren’t satisfied, the company offers a full refund within 30 days.

The SmartScope software is organized into sections along the left, and each section opens up when selected to show its available features and options. The Measurements Box at bottom can float on top of the main display, which is useful when using a tablet or smart phone having a small screen. But it can also be docked at the bottom as shown, or on the right edge. Another pane at the right opens to let you choose which measurements to display in the box.

Wave traces for the two channels are shown in different colors, and you can slide them up or down, or hide a channel when it’s not needed. The voltage legends at left reflect whichever channel is currently highlighted, in this case green for Channel A. Clicking anywhere in the blue area changes the voltage values at left to those for Channel B (see **Figure 2**). The software also supports touchscreens and recognizes gestures, in addition to a mouse, on platforms that support touching.

One of the warnings I heard from USB nay-sayers on Facebook is that I’d be unhappy with the latency. That means when an input signal changes, or you modify an oscilloscope setting, the change won’t show on the screen for half a second or so. But with the default buffer setting of 512K samples, the SmartScope responds as quickly as any hardware model I’ve used.

SmartScope calls this the Acquisition Depth, which is set in the System section at the bottom of the left pane. However, there’s enough sample memory to store up to 64 million samples for each channel. So when set to the maximum 64M size, there is some latency. But at that size you can freeze the display and zoom in afterward by a factor of 1,000 to see what happened leading up to an event in your analog or digital logic circuit that went by too quickly to view in real time.

I don’t have a touchscreen, but the software works beautifully with a mouse. With the mouse cursor over the main part of the screen, or over the time per division value at the bottom right (50 μs as shown in **Figure 3**), the scroll wheel zooms the horizontal time scale between 200 ms and 10 ns. If you set the Acquisition Depth to the maximum 4M

samples, the time can be set as slow as 2 seconds per division. And when you hover over either channel's voltage scaling, the vertical sensitivity for that channel changes over a range from 5 V down to 20 mV per major division. I found setting these parameters with a mouse to be faster and easier than handling the physical knobs on a "real" oscilloscope.

The Features

I have to admit I haven't yet tried the logic analyzer shown in **Figure 4** because my current project is purely analog. I didn't try the digital generator either because I don't have a suitable application, but I assume it works as the vendor claims. The analog signal generator shown in **Figure 5** certainly works as advertised, with one small and possibly expected limitation: Unlike a typical analog hardware generator, you can't vary the parameters in real time. Rather, you pick an output voltage, wave shape, and frequency, then click Upload, which sends the current settings to the output. If you change any of the parameters, the output doesn't automatically follow; you need to click Upload again before the new values take effect.

The analog Signal Generator offers all of the common analog waveforms, plus some combinations, and lets you set the output voltage, a DC offset, and frequency. The digital generator allows various signals such as Counter and Pulse, plus user-defined signals at either the 3 VDC or 5 VDC standard digital output levels.

Conclusion

One limitation of the signal generator is its waveform accuracy, most noticeable with sine waves. As you can see in **Figure 6**, when set for 1 kHz at 1 V output, most of the lower-order distortion artifacts are only 35 dB below the fundamental frequency. Though at around 3% THD, it's similar to inexpensive function generators. And it works perfectly well for basic testing such as frequency response.

There are a number of other useful SmartScope features including "operator math" that adds Channels 1 and 2 to show their combined voltage, or you can subtract one channel from the other to view their difference. You can also define your own probe values, and there's a Mixed Mode that displays one analog channel along with the logic analyzer on the same screen. But most of these features are "icing on the cake" so to speak, and go beyond what anyone could reasonably expect in an oscilloscope that costs only a few hundred dollars! So I'm totally satisfied, and this petite device now has a permanent place on my desk. 📺

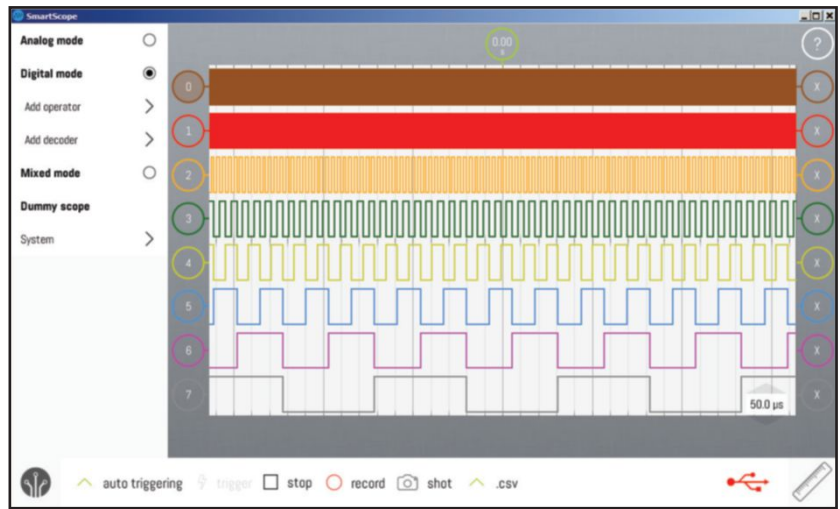


Figure 4: The Logic Analyzer can show up to eight digital input channels at once.

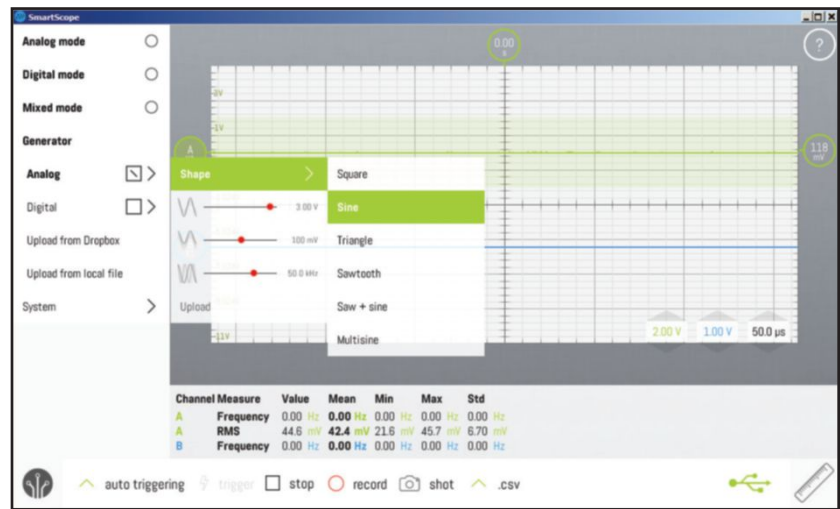


Figure 5: The analog Signal Generator is similar to a stand-alone function generator, and it contains all the expected features.

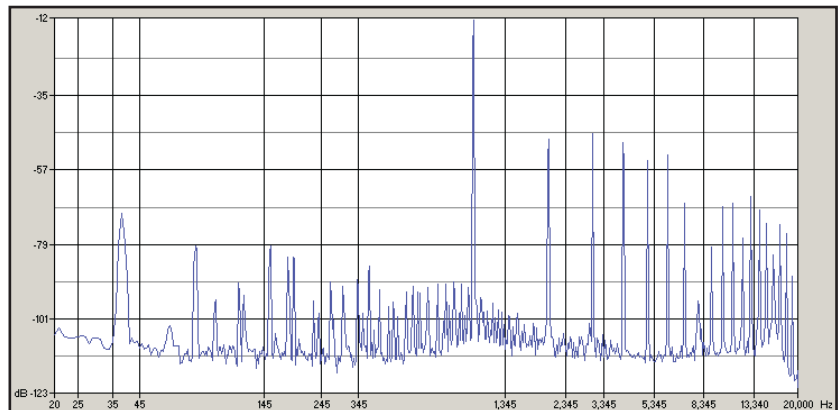


Figure 6: As you can see in this FFT of a 1 kHz sine wave, the analog signal generator isn't suitable for testing distortion because of its high harmonic content.

Resources

LabNation: www.lab-nation.com

User Manual: http://wiki.lab-nation.com/index.php/Main_Page



You Can DIY!—Speaker Feature

The Incredible Egg (Part 3)

The Proof Is in the Listening

By
Thomas Perazella
(United States)

In the last of his three-part series, Tom Perazella wraps up his latest bold speaker project, inspired by the Mach 17 speaker built in the 1990s by Waveform Research of Canada.

Tom Perazella's completed Egg Speakers are shown as they premiered at Capital Audio Fest 2017.
(Photo courtesy of Paul Elliot)

This three-way design features the latest generation of SB Acoustics' high-end Satori drivers and two Dayton 12" woofers in a separate sealed box base. As previously mentioned, for maximum flexibility, the crossovers were designed to be used either passively or actively. In the passive mode, the crossovers would use a 12 dB/octave Linkwitz-Riley configuration for all drivers with crossover points of 200 Hz and 2,500 Hz. Those points were chosen to provide minimum stress on the drivers and also to assure that the driver's radiation patterns were the same through the crossover region.

To assure that the proper component values could be chosen when designing the crossovers, I measured all the drivers for impedance across their operating ranges with the application LIMP, which is part of the ARTA measurement suite. This is a very important step because relying on published specs can lead to large errors in the actual vs. predicted crossover points. As an example, the 12" woofer used with the voice coils wired in series had a specified impedance of 8 Ω . The actual measured impedance

at the desired crossover point of 200 Hz averaged around 11.4 Ω for the four drivers. Picking component values for 8 Ω would have led to a very large shift in crossover frequency at the actual impedance of the driver at 200 Hz.

As previously mentioned, the woofer had a resonance at 2,100 Hz that was quite narrow and high. I was not sure that the crossover slope would completely eliminate the effect of that resonance even though the drive would be greatly reduced by the crossover. To assure that there would be no significant contribution to the overall sound from the woofer resonance, I decided to add a series tuned circuit that would essentially increase the circuit impedance to the woofers at that frequency to a very high value, further reducing drive beyond what the crossover would provide. The circuit combined an inductor paralleled by three capacitors to achieve the required capacitance needed. In theory, this type of series tuned circuit, if it had zero resistance, would have an infinitely high Q and infinite impedance at the tuning frequency.

In reality, the component impedance plus driver impedance result in a still high but not infinite impedance with a lower Q, spreading the effect out across a broader frequency well beyond where the woofers are operating.

As it turned out, with a sine wave sweep, there was a small amount of interference from the woofer resonance that could be detected when the tuned circuit was not used. When used, the effect disappeared. In addition with many types of music, there was no audible effect of the resonance with or without the tuned circuit, but since I had constructed the crossover with the tuned circuit I left it in place. **Figure 1** is the schematic for one channel of the crossover.

In the crossover, L1 and C1 are the low-pass elements for the woofers. L2 and C2, C3, and C4 are the resonant circuit to reduce drive to the woofers at 2,100 Hz.

L3 and C5 provide the low-pass function at 2,500 Hz for the midrange and L4, L5, C6, and C7 provide the high-pass function at 200 Hz for that driver. R1 and R2 are the components of the L pad to reduce the output of the midrange to match the woofer. Originally, I calculated that I needed about 2 dB of attenuation, but later listening and preliminary measurements confirmed that the actual value was closer to 3.2 dB so the L pads had to be reworked.

C8, C9, and L6 provide the high-pass function for the tweeter. The impedances of the two tweeters were slightly different at 3.62 Ω and 3.80 Ω at the crossover frequency. I knew that I would need considerable attenuation of the tweeters to match the woofers and midranges so I decided to add a different amount of series resistance to each tweeter and at the same time increase the total resistance to a level that would allow the use of a standard value inductor in the crossover.

R3 and R4 formed the L pad for the tweeter resulting in 8 dB of attenuation. To get the impedance of both tweeters the same, the series resistor R5 had two different values, 0.38 Ω and 0.20 Ω for the two tweeters, resulting in an impedance of 4 Ω for both.

The crossovers were built on three perforated crossover boards per channel from Parts Express. These boards are a convenient way to build crossovers of any configuration. One thing to note is that I used thick film 35 W non-inductive resistors in pseudo TO220 cases for some of the resistors. These resistors not only have excellent electrical characteristics but are easy to mount on heatsinks, having insulated tabs. **Photo 1** shows the woofer crossover and tuned circuit trap. **Photo 2** shows the two boards used for the midrange and tweeter crossovers and L pads.

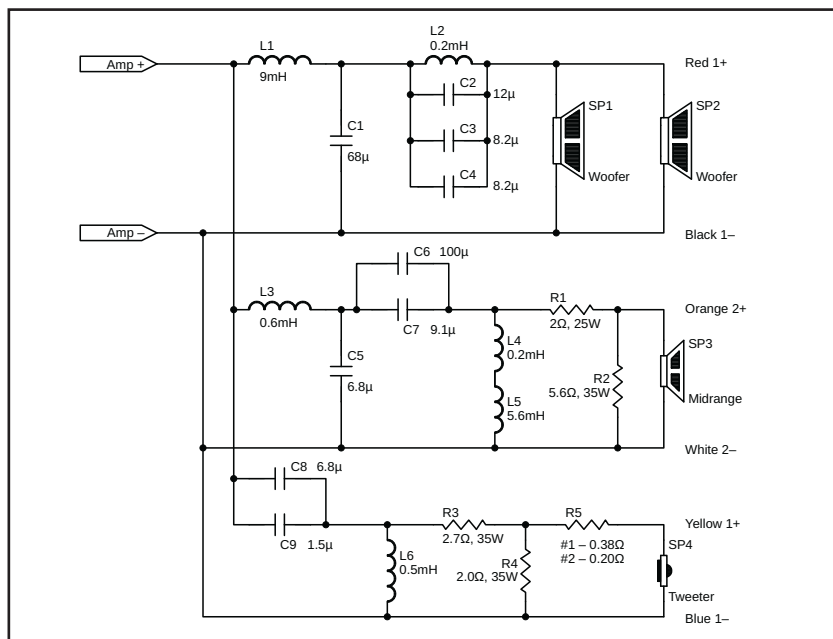


Figure 1: The crossover schematic

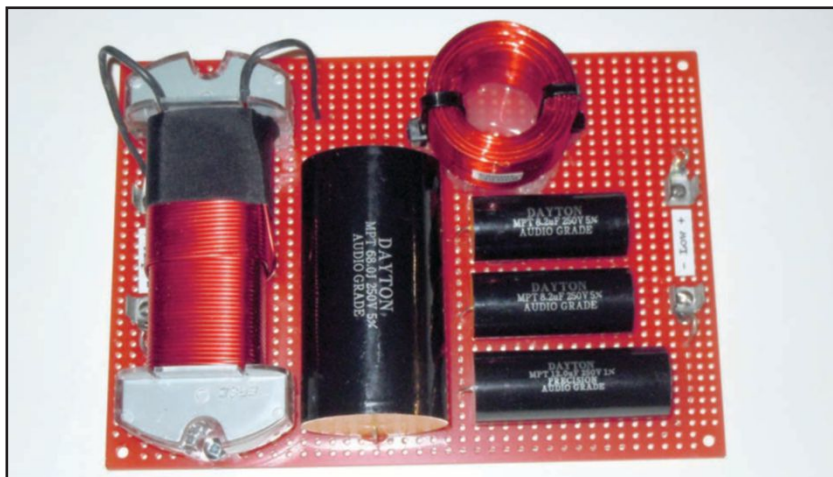


Photo 1: The woofer crossover with a resonant trap

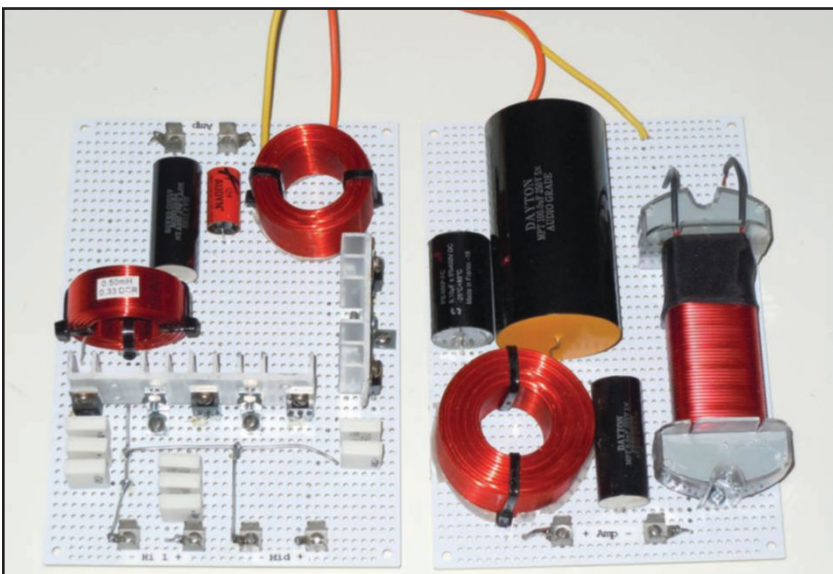


Photo 2: Mid and high crossovers with L pads

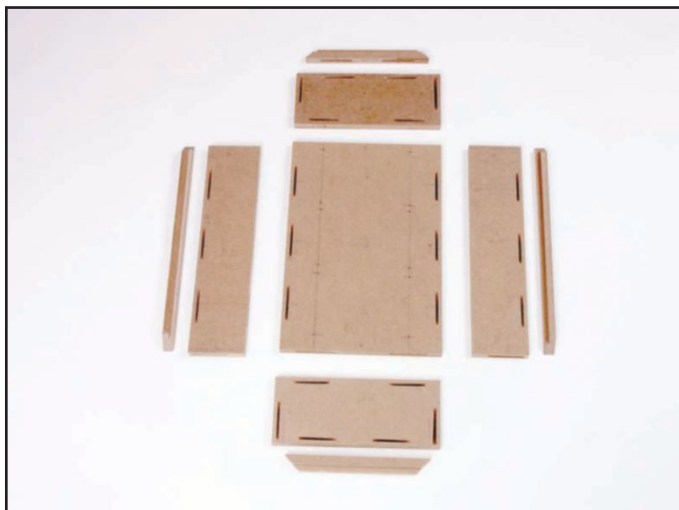


Photo 3: MDF parts for the crossover enclosure

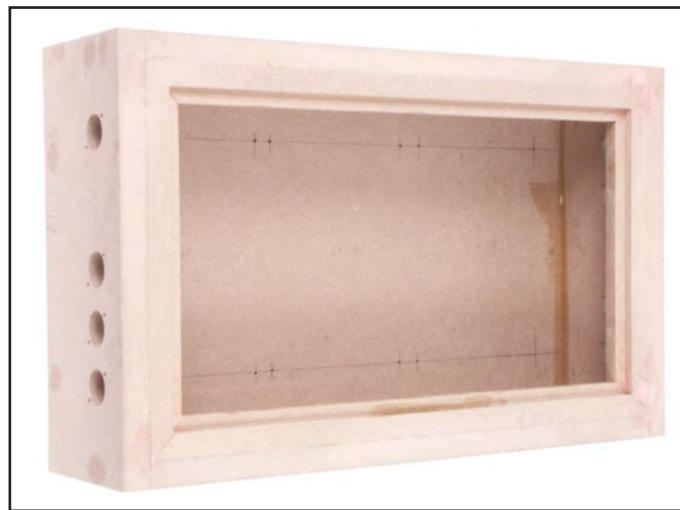


Photo 4: Bottom view of the completed crossover housing

Housings for the Crossovers

For this project, I wanted to have an external enclosure for the crossovers for several reasons. Having an external housing would make any part changes vastly easier. I have found in the past, regardless what method is used to predict the passive components, changes are always necessary. The second reason is that by making the housing for the crossovers with a clear acrylic bottom plate, the components could be easily seen, which was a plus when demonstrating the speakers at shows.

The rectangular housing was constructed from several pieces of 3/4" MDF and then glued together. The four pieces that formed the bottom support for the acrylic plate were cut with a recess so that the acrylic plate would be flush with the bottom (see **Photo 3**). After assembly, I cut the holes for the Speakon connectors and drilled pilot holes for the fastening screws. I routed the edges with an edge bit and sanded the housings (see **Photo 4**).

Preparation for Painting

With any wood project, the filling of any voids and multiple steps in sanding are very important. For this stage, I used the

Bondo wood filler mentioned previously to make sure there were no nicks in the wood. I did the sanding in the same sequence as described for the eggs (*audioXpress*, October 2018). I employed the same sanders and grits, resulting in very smooth surfaces.

Preparing MDF for veneering or painting with a textured paint such as DuraTex is relatively straight forward. One caution is that you should not use a water-based primer with MDF. Water can soften the surface and result in an uneven finish. Use a non-water-based primer such as Kilz. For this project, the preparation was especially important as an automotive finish was to be used so I left all the priming to the paint shop. These finishes are difficult to work with for at least two reasons. First, the final finish is very smooth and reflective, highlighting any irregularities in the base material. Second, they are very thin compared to regular paints with a reduced capability to fill voids and other irregularities.

Using an Specialty Painter

From the start of this project, it was clear to me that an outside vendor who had experience with automotive finishes would be needed as I had neither the experience nor the facilities to tackle this paint job. I was lucky in that my brother-in-law has a neighbor, Steve Sees, with an auto body shop who was willing to take on the project. This type of painting truly is an art as there are so many factors that can affect the final outcome especially when working with metallic paints. Steve had a lot of experience not only with repairing all types of regular cars and trucks but also custom hot rods with exotic finishes. Steve likes challenges and this project certainly fit that description. I'm sure he did not have a full understanding of the difficulty of working with these materials and shapes.

The paint I chose was that used on my wife's Buick Verano. It is from PPG and is called Carbon Black Metallic. The PPG number is 917964. This finish appealed to me because it is quite subtle. Most of the time it simply looks like a shiny black surface. Given the right lighting, the metallic flakes in the paint result in a very fine sparkle that has the appearance of both silver and cobalt

About the Author

Thomas Perazella is a retired Director of IT. He received a BS from the University of California, Berkeley campus. He is a Past President of the Rockville Chapter of the Izaak Walton League of America, one of the oldest national conservation organizations in the US and currently is the Treasurer. Audio has been his passion for more than 50 years and he is a member of the Audio Engineering Society, the Boston Audio Society, the Philadelphia Area Audio Group, the DC HiFi Group, and the DC Audio DIY Group. Thomas has written for *audioXpress* magazine and prior to that for its predecessor, *Speaker Builder*. In addition to audio, his interests include photography, cooking and competition pistol shooting. He has authored several articles in professional audio journals and taught commercial lighting at the Winona School of Photography. Recently, he received a patent on a cost-effective high-efficiency LED lighting system for commercial and residential buildings.



Photo 5: Steve Sees applying "icing"



Photo 6: Iced egg standing on the painting fixture

blue. The fine sparkle adds an elegant look to the speakers that would not be the same if large metallic flakes were present or if the finish were pure black.

The Painting Process

To assist Steve in the painting process, I made two fixtures to hold the eggs in a position high enough that all surfaces could be painted at one time. I also made fixtures to hold the housings for the crossovers.

Because of the size of the pyramids, I had to rent a trailer to get them to Steve for painting. Once there, I turned them over to him for final preparation and paint. Steve immediately identified two problems with the condition of the eggs and the pyramids. First, because of the differences in hardness of the MDF and glue joints, it was impossible to get an even surface when sanding. In addition, the MDF is very porous, which would have absorbed a lot of the paint. As a result, Steve felt the differences would be quite noticeable when painted and finished with clear coat. To correct that he would use a material that is known in the trade as "icing," a thin polyester filler, to achieve the necessary surface smoothness (see **Photo 5**).

Second, because I had done such a good job of sanding with very fine grits, Steve was concerned that the icing might not adhere properly to the surface of the wood since it was very slick. He, therefore, had to roughen up the surface before starting the work (see **Photo 6**).

The steps in the finishing process included roughing up the surfaces with a 120-grit sanding, icing where necessary, two coats of urethane primer, sanding after the second coat with 400 grit, three coats of the base-color coat containing the metallic, and finally three layers of clear coat. After the clear coat dried, a

wet sanding using 1500 grit was done followed by buffing with a polishing compound. This is quite a lengthy and laborious process that was far beyond my capabilities.

After talking with Steve about the process, I became aware of some of the techniques necessary to get a good finish with this type of paint. First, you can't just move the spray gun back and



Photo 7: Steve Sees painting the bass pyramid

Photo 8: The finished egg



forth across the surface but rather have to move it in various directions (see **Photo 7**). That way, the flakes of metal will not all line up in one direction and give a better reflective look from multiple viewing points. Second, the time in between the various steps is critical and depends on the temperature and the humidity as well as the different materials being used in the various steps. And, of course, prep is crucial.

When I went to pick up the finished enclosures, I realized just what a good job Steve had done. It was apparent he had spent a huge amount of time because the finish was like glass and the black was very deep. When light hit at the right angle, the fine metallic added a luster to the look. Once the drivers were installed, the black cones of the woofers, the white cones of the midranges, and the steely color of the beryllium tweeter domes were striking against the black metallic finish (see **Photo 8**).



Photo 9: Front view of the finished egg/pyramid

The Final Steps

I stuffed the pyramids with 5 lb of Acoustastuff each. I did not measure the exact amount of Acoustastuff used in the eggs but it was a loose fill. For the final installation of the drivers, I first attached the lead wires to the drivers and then fed the wires through the holes for the Speakon connectors. I then attached the drivers to the enclosures with particle board screws in the



Photo 10: Amplifier inputs on the front of the crossover



Photo 11: Speaker connections on the rear of the crossover

predrilled pilot holes. Next, I attached the wires to the connectors and inserted the connectors in the holes and screwed them in place. On the bottoms of each of the pyramids, I attached four large rubber feet. On the bottoms of the eggs, I glued a circular piece of felt the same size as the bottom in place.

Photographing the speakers was a challenge because of the shapes and finish. With shiny black surfaces you cannot just shine a light directly at the surface as it acts like a mirror and all you will get is a hot spot. Instead, you have to have white surfaces in front of the subject and light those surfaces. That way the reflection of the white surfaces is what you see in the object being photographed. That technique was used to tame the various surfaces of the enclosures and help define their shapes in the photos (see **Photo 9**).

Before I assembled the housings for the crossovers, I drilled pilot holes to mount the three crossover boards. I cut the acrylic bottom sheets to size and sanded the edges to smooth them. Then, I installed the Speakon connectors for the passive and active implementations. One side of the housing had a single connector for one amplifier in the passive configuration and three connectors for each of the low, mid, and high amplifiers when used in the active mode. I was lucky to find a company that made self-adhesive engraved labels with a gold colored surface. They were used to identify each of the connectors (see **Photo 10**).

The other side of the housing had four Speakon connectors. I used two to connect the speakers in the passive mode and two in the active mode. Even though there are two drivers in each egg, only one connector was needed as the egg connectors were four conductor and the cable between the crossover and egg had four conductors (see **Photo 11**).

In the crossover housing, the connections between the amplifier inputs and speaker outputs for the tri-amped mode are simple direct wire between the connector terminals. The wires in the passive mode go to the appropriate crossover board connections for both the amplifier and the speaker connectors (see **Photo 12**).

The First Audition

The only location I had for the first listen of the speakers was in my basement. That is a terrible location, as it has a recessed section that acts as a bass trap but that is the only place I had free for the test. I put the speakers in that location and they were rather close to the side and back walls. I expected significant problems.

To drive the speakers in the passive configuration, I used my Hewlett-Packard (HP) Envy laptop running

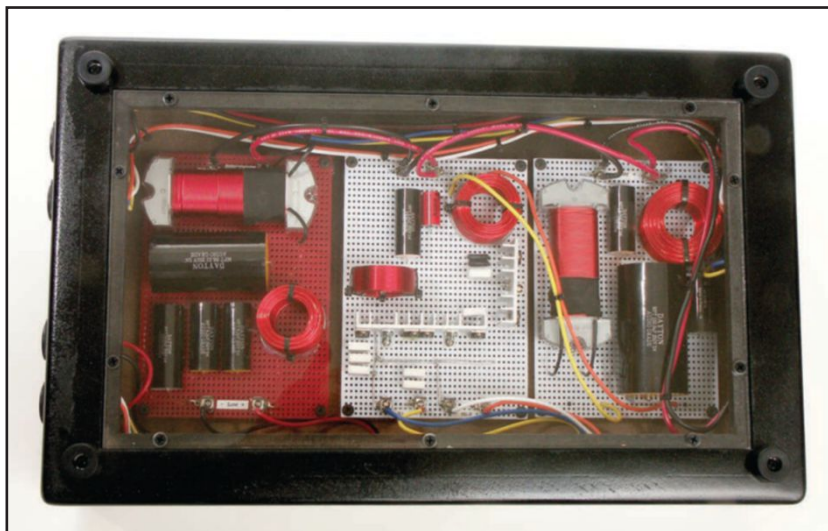


Photo 12: Bottom view of the completed crossover

JRiver Media Center 23 connected to a Leaf Audio Mini USB DAC that uses the XMOS XU208 receiver and the AK4490 DAC chip. All the music was on a Western Digital Elements USB 2TB drive. The amplifier was a Crown Studio reference to insure that regardless of speaker impedance or sensitivity the amp would not become a limiting factor.

To say that I was surprised with my first listen is an understatement. Even in that bad listening position, the sense of space and instrument position was outstanding. The highs were detailed while still very smooth. Bass ranged from somewhat weak to quite good depending on where I was in the room.

I did not have a lot of time to listen as the speakers were to be one of the center points of the



Photo 13: Speakers displayed in the DC HiFi Group room at Capital Audiofest

DC HiFi Group room at the Capital Audiofest show. So off to the show they went, which was no small feat in itself as the pyramids with the drivers installed weigh around 160 lbs each. As usual at these shows we had a small hotel room to display them. Again, even in the small room they were a big hit. We received comments like the best sound at the show, smooth yet detailed, better than my electrostats, and so forth. At times, there was standing room only and the number of people that came back several times, often bringing friends with them for further listening was very rewarding (see **Photo 13**).

Photo 14: Cracks in the egg's surface from the screw's expansion



Photo 15: EZ-Lok inserts



Photo 16: Installed EZ-Lok inserts



There is Always a Problem

Regardless of how many times you work on similar projects or follow the same procedures, there is inevitably a problem with a new project. In this case, the problem showed up after about a month. Over the years I have used the same-sized particle board screws in the same-sized pilot holes to mount drivers with good results. So why should that practice not work now? Well, I had never attached anything with screws into the ends of MDF. In the case of the eggs, since they were made with layers of MDF glued together, the pilot holes to mount the drivers had to go into the ends of the MDF slices. I was careful to not have any of the holes in the glue joints as I felt that might be a problem. At first, everything worked okay. However, over time, the edges of the MDF sheets separated slightly, resulting in slightly visible hairline cracks in the surface of the eggs (see **Photo 14**).

After much thought, I decide to change the mounting method of the drivers in the eggs. I had no problems with the pyramids as the mounting holes were not in the edges. I had previously used EZ-Lok inserts to enable the use of machine screws in wood. They come in several versions, one of which is for particle board (see **Photo 15**).

Simply inserting the EZ-Loks into enlarged holes and relying on the external threads of the fastener would not solve the problem as they could potentially also spread the MDF apart. Instead, I decided to drill the largest hole I could while having just enough material to hold the fastener in place. The drill I finally used was a letter "K" bit, which is 0.281", not the 6 mm recommended by the manufacturer. This would not provide enough holding power for the drivers but to secure the fasteners, I first put a small piece of masking tape on the bottom hole of each fastener and then coated both the inside of the hole and the outside of the fasteners with J-B Weld epoxy (see **Photo 16**).

I had tested this method on a piece of scrap MDF before going this route and found that once the epoxy hardened, if I used a washer under a screw that went into the fastener, I had to tighten the screw to the point that the washer significantly deformed before the epoxy joint was affected. Obviously, you would never use this much torque when fastening a driver or for that matter anything else. The drivers were re-installed and the small cracks filled with the appropriate automotive touch up paint from Buick. Problem solved!

What Is Next

During the show, and after a lot of listening, I felt that my first estimate on the levels of attenuation of the midrange and tweeter were off

by about 2 dB. I then changed the crossovers to add approximately 1.8 dB more attenuation in the midrange and tweeter L pads because that amount was attainable with standard value resistors. The sound was more neutral and I did a quick frequency and distortion test at 85 dB using ARTA and my ACO Pacific 7012 mic capsule. The response in that room was amazingly good and the distortion was in the range of good electronics (see **Figure 2**).

To properly test these speakers, I plan to move them to my living room. But first, I have to move my main system consisting of two BG RD75 planar magnetic speakers in custom dipole baffles, two dipole midbass speakers each using six 10" Peerless drivers, and my four 15" subs to the basement to make room for the Eggs. I will then test them again in the passive mode plus use two other devices for the active mode. The first will be the DEQX Express II crossover/DSP that I currently use in my main system and also the Legacy Wavelet processor used on the Legacy Valor speaker system.

It will be enlightening to hear what differences, if any, there will be with each of these three methods, but that is a project for another day. In the meantime, I will sit back and enjoy the Eggs. 🎧

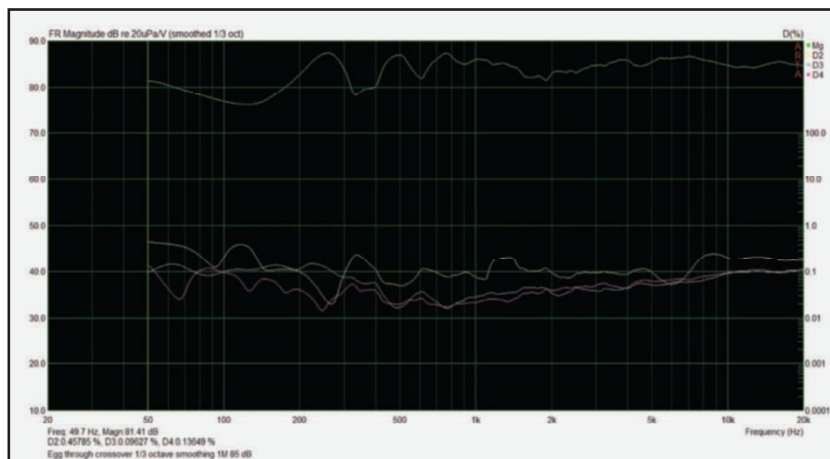


Figure 2: The frequency response and the harmonic distortion plot

Project Files

All references, parts lists, and other pertinent information is available on the *audioXpress* website.

To download the additional material and files, visit

<http://audioXpress.com/page/audioXpress-Supplementary-Material.html>.

Ribbon Loudspeakers
what you
need to
know.

Order your copy
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 1-800-572-7280

Use code "AX" on your web purchase to get a discount!

Reflective-Way, Two-Way System (Part 1)

Preparing for the Build

In this two-part article series, Peter Lehmann shares his design for a loudspeaker system with an integrated reflector for the tweeter.

By
Peter Lehmann
(United States)

Testing the response of the pair of drivers that I had selected for my loudspeaker system, in reproducing a tone burst, I found the woofer's response lagging behind that of the tweeter. So, I designed a loudspeaker system that solves this problem. Additionally, my design lends a spacious quality to reproduction by controlling the time of arrival of high-frequency first reflections off of the room's side walls from my loudspeaker system to the listener.

Relative Impulse Response of the Drivers

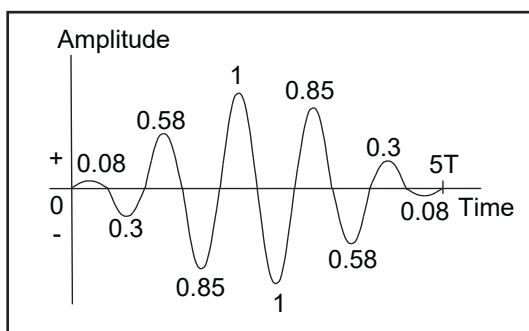
The cosine tone burst shown in **Figure 1** was my

test signal for measuring the lagging response of my pair of drivers. Applying the electrical tone burst across the input terminals of each driver enabled me to measure with an oscilloscope how long it took each driver to reproduce the peak amplitude of the fifth half-wave compared to the time of maximum positive amplitude of the applied (electrical) tone burst.

The tone burst shown in Figure 1 is a sine wave that is amplitude modulated following Siegfried Linkwitz. The amplitude of the nine other half-waves compared to that of the fifth half-wave equal to one are shown at the peak. As the tone burst consists of 10 half-waves, the duration of the tone burst is equal to five times the period (T) of the sine wave.

Figure 2 is a block diagram of my apparatus for measuring lagging reproduction of the drivers. The sine wave generator is at the input to the cosine burst generator. The output of the burst generator is connected to the input of a positive peak detector and the input of a power amplifier. The burst generator periodically produces a tone at one-second intervals. The output of the power amplifier is across the input terminals of the driver under test (DUT). The first and second channel inputs of

Figure 1: The graph shows amplitude vs. time of a cosine tone burst developed for frequency response testing of loudspeakers. My additional use of it is to test how long a driver takes to complete reproduction of the initial transient of the tone as a function of frequency.



the oscilloscope are connected, respectively, to the outputs of the peak detector and the microphone. The microphone is as close as possible to the driver's diaphragm without touching it (near-field).

Testing with the setup shown in Figure 2, for each trial frequency, the third step of the output of the positive peak detector is set to occur at a major 10-mm vertical division of the oscilloscope's screen. The time delay in reproducing the peak of the fifth half-wave of the tone burst is the horizontal distance on the oscilloscope between the selected vertical division and the location of the peak of the fifth half-wave of the Channel 2 trace.

Figure 3 is a graph showing delay in the reproduction of the fifth half-wave of the tone burst. The drivers I tested were the Newark woofer Stock No. 55-1190 and the Goldwood tweeter GT-525. Testing the woofer and the tweeter at a frequency equal to 200 Hz and 2 kHz, respectively, showed no time delay. Average delay of the woofer relative to that of the tweeter equaled $197 \mu\text{s}$ minus $38 \mu\text{s}$, or $159 \mu\text{s}$. Greater lagging response of the woofer, where a tone is about equally reproduced by both woofer and tweeter, results in distortion of the tone's initial transient.

The woofer's lagging response is exacerbated by mounting the tweeter and the woofer on the same perpendicular baffle, causing the woofer's voice coil to be behind that of the tweeter. In the case of my drivers, this results in the radiation center of the woofer 2.3" behind that of the tweeter.

Correcting the Woofer's Time Lag

Compensation for both the woofer's lagging response and its radiation center normally behind that of the tweeter could be accomplished simply by positioning the tweeter at a distance behind the front baffle and on top of the cabinet holding the woofer.

The average time delay of response of the woofer was calculated to be $159 \mu\text{s}$. This average delay times the speed of sound equal to 13,546" per second equals 2.2". Thus, mounting the tweeter on the top panel of the system cabinet, 4.5" behind the baffle to which the woofer is attached, could compensate for both defects. However, then the top panel diffracts some of the tweeter's sound.

Alternatively, compensation could be accomplished by causing an increase of the path of sound from the tweeter to the listener by the appropriate amount by reflection off of a nearby reflector. Given the tweeter's radiation center coincident with the vertical plane of the baffle to which the woofer is attached, reflecting the

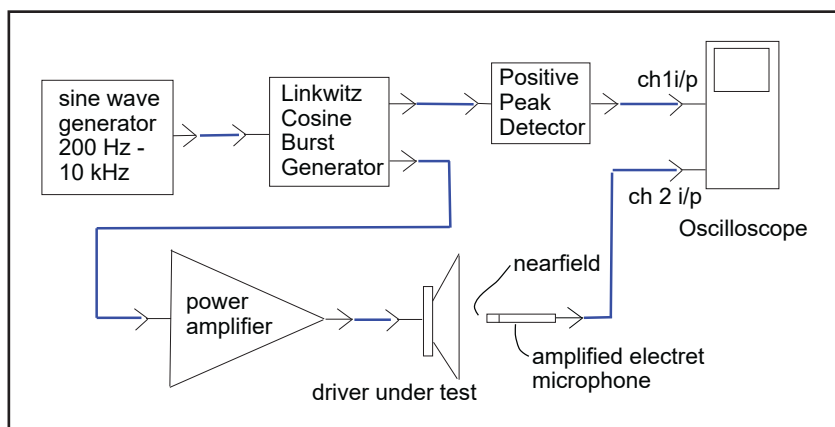


Figure 2: The arrangement of equipment shown was used to detect and measure any delay in sound generation of the fifth half-wave of the tone burst of Figure 1. The crossover frequency of my two-way system is 2 kHz. Thus, testing of the response of my system's woofer and tweeter is in the ranges of respectively 200 Hz to 2 kHz, and 2 kHz to 10 kHz.

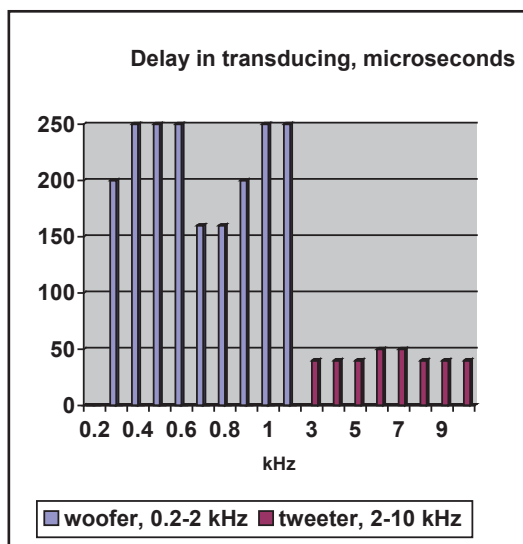


Figure 3: The graph shows that there is a significant delay in completing the initial transient of the tone burst shown in Figure 1 by the woofer relative to a smaller time delay by the tweeter. This data was collected on my drivers using the testing arrangement shown in Figure 2.

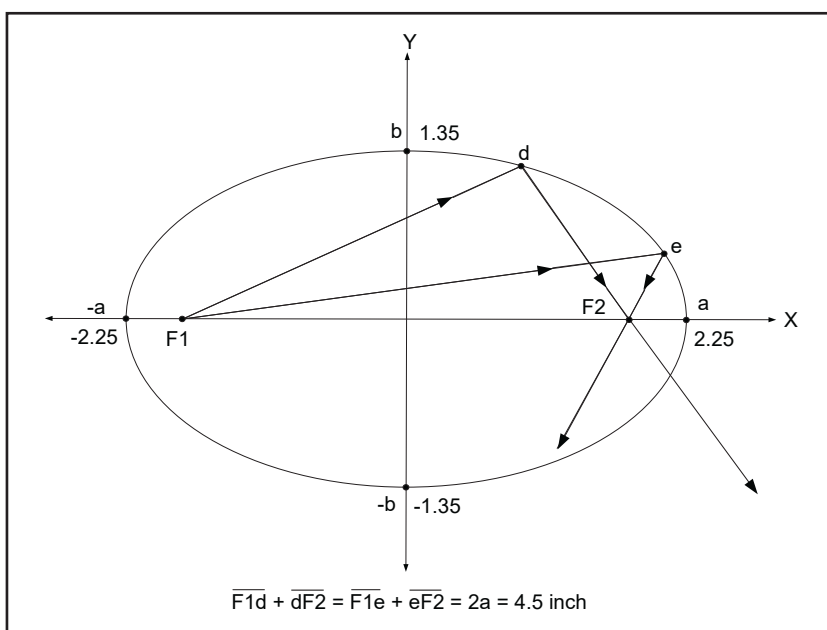


Figure 4: In a top view, the center of radiation of the tweeter is located at focal point F1 and reflected from the curvature of the ellipse in the first quadrant. For the ellipse shown, the length of a sound ray reflected to a listener is 4.5" longer than that traveling directly to the listener.

tweeter's sound to the listener should increase the path length by 4.5".

Figure 4 shows the first quadrant of an ellipse used as a reflecting surface to increase the path length of the sound from the tweeter to the listener. The curvature is in the horizontal plane and is flat in the vertical plane projecting out of the page (top view). The X-axis is coincident with the vertical plane of the front surface of the baffle holding the woofer. The tweeter's center of radiation is located at focal point F1. Focal point F2 is located directly above the radiation axis of the woofer.

The semi-major axis of the ellipse shown in Figure 4 has length equal to 2.25". The reflection of any ray emanating from focal point F1 passes through F2 and the total length of the ray from F1 to F2 equals 2a. The first and second sound rays with reflection points d and e, respectively, in the figure have identical length. As two times a equals 4.5", the reflector increases the distance that sound from the tweeter travels to the listener by about 4.5", relative to the path length of sound directly

from the tweeter to the listener.

The elliptical reflector shown in Figure 4 is advantageous in that the tweeter is positioned off to the left side of the reflecting curvature in the first quadrant, at point F1, and thus, does not obstruct most of the reflected sound. An incident ray originating at F1 at a shallow angle to the X-axis is reflected back toward, and diffracted by the body of, or the baffle for, the tweeter. Thus, not all of the curvature in the first quadrant can be used for reflecting.

Tweeter on a Flat Baffle in Listening Room

Figure 5a and Figure 5b, not to scale, illustrate the effects in a room of the horizontal angle of the tweeter's radiation axis and the size and the horizontal angle of the tweeter's flat baffle. In Figure 5a, the dome of the left channel tweeter pointing toward the far (right) sidewall is beneficial for two reasons. First, as opposed to being reflected off of the near (left) sidewall, this increases the length of ray 2

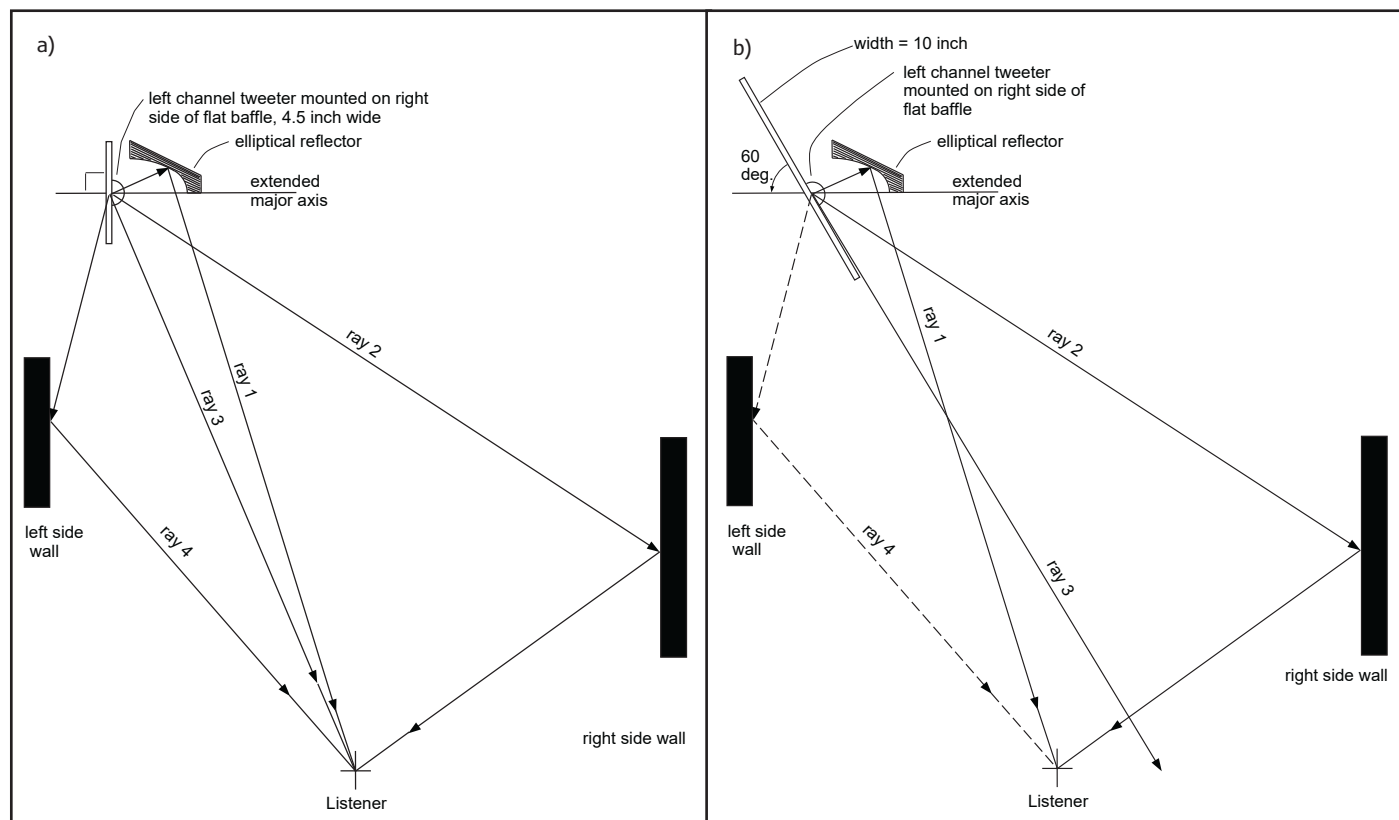


Figure 5a: Placing the left channel tweeter and nearby reflector in a room and tweeter radiating towards the far rather than the near wall increases the delay of arrival to the listener of ray 2 relative to that of ray 1. Ray 3 muddies delayed hearing of "direct" reproduction by the tweeter, relative to hearing sound from the woofer. Ray 4 of shorter length than Ray 2 and first arriving to the left ear of the listener may produce image shift and coloration. Figure 5b: The direction of radiation by the tweeter and width of the baffle for the tweeter from the arrangement shown in Figure 5a are modified so that the tweeter within its operating frequency range essentially radiates into half-space only and sound from it cannot arrive directly to the listener. As opposed to the arrangement shown in Figure 5a, ray 3 passes by the listener and level of ray 4 is substantially less than the level of rays 1 and 2.

relative to that of ray 1. If the first reflection ray 2 arrives to the listener only a few milliseconds after the arrival of sound reflected off of the nearby reflector, ray 1, then this produces unwanted image shift and coloration. However, if the difference of distance traveled by ray 2 minus that of ray 1 times the inverse of the speed of sound equals at least 10 ms, then image shift and coloration is replaced by a sense of spaciousness of reproduction. This is based on research conducted by Mike Barron in 1971 and cited by Floyd Toole (see Resources). Second, ray 1 and ray 2 arrive first at respectively the left and right ears of the listener. This results in a reduced perception of image shift and coloration compared to the effect of ray 1 and ray 2 first arriving to the same ear of the listener. Ray 1 and ray 2 would conversely first arrive at the same ear of the listener if the tweeter were obliquely facing the near (left) side wall.

The flat baffle of Figure 5a is perpendicular to the extended major axis (of the reflector's curvature). The major axis, in turn, is perpendicular to the sidewalls of the room. This allows sound to radiate directly from the tweeter to the listener shown as ray 3. Ray 3 muddies the time delay of arrival of sound from the tweeter to the listener by means of being first reflected off of the nearby reflector (ray 1).

Given the width of the baffle shown in Figure 5a equal to 4.5", the baffle step frequency of this baffle equals 1 kHz. This follows from an equation for calculating the step frequency in Hertz equal to 380 divided by the width of the baffle in units of feet. Also, examining the graph of diffraction loss vs. frequency of a 24" sphere (following Harry Olson) in an article about the baffle step, the graph indicates a threshold of radiating into full space from radiating into half-space occurring at a frequency that is about two octaves above the baffle step frequency.

Thus, the tweeter mounted on the 4.5" wide baffle shown in Figure 5a partially radiates into full-space for frequency less than about 4 kHz. The crossover frequency of my design equals 2 kHz. Thus, ray 4 or a first reflection off of the near (left) sidewall can occur in the range of 2 kHz to 4 kHz. The undesirable Ray 4 may produce image shift and coloration.

With Figure 5b, the left channel tweeter, the flat baffle, and the nearby reflector in the room is identical to that of Figure 5a, except that radiation of the tweeter is directed obliquely toward the rear and the right sidewall of the room and the width of the baffle is increased to 10". With the angle of the baffle to the major axis changed from 90° to 60°, ray 3 cannot arrive directly to the listener. The time

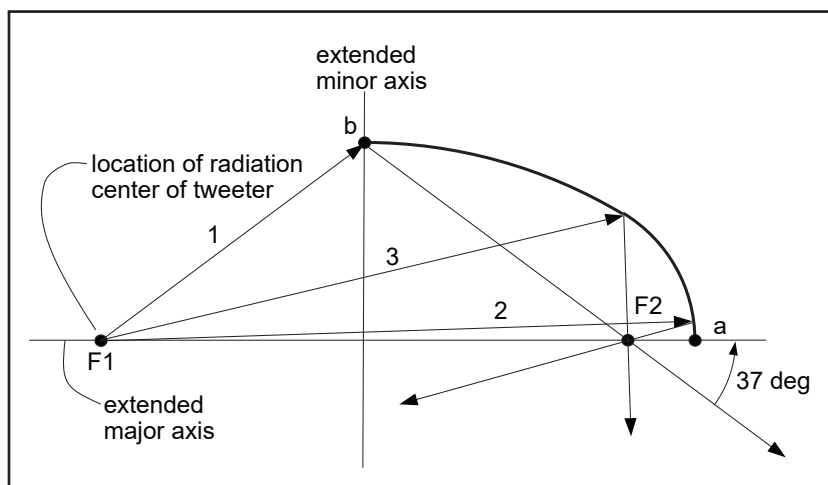


Figure 6: The first quadrant of the ellipse detailed in Figure 4 is shown reflecting three sound rays from the tweeter located at focal point F1 of the ellipse. Curvature in the second quadrant for reflection is not an option as then the angle of reflection to the major axis is too shallow. Reflections at locations close to the major axis are also at too shallow an angle.

delay of the "direct" sound (ray 1) from the tweeter to the listener is, therefore, preserved.

The width of the baffle shown in Figure 5b increased to 10", lowers the baffle step frequency (f_3) to 456 Hz. Two octaves above f_3 is about 2 kHz, which is also the tweeter's cut-off frequency. Thus, the baffle of 10" width essentially limits the tweeter to radiating into half-space only, and ray 4 representing the first reflection to the listener for full-space radiation of the tweeter and shown as a dashed line in Figure 5b, is greatly attenuated.

Transforming the Elliptical Curvature to Circular

Using the first quadrant of the ellipse shown in Figure 4 presents three difficulties. (1) Intensity loss by reflection is inversely proportional to surface area. (2) The range of angular reflection horizontally ideally is in a range of about 90°. (3) An elliptical curvature is difficult to accurately fabricate.

Figure 6 shows three rays reflected by the curvature of the first quadrant of the ellipse shown in Figure 4. Ray 1 reflected from the curvature at the minor axis is reflected at an angle of 37° to the major axis. All of the reflected rays should be directed to the listening area, so the angle of reflection of ray 1 is probably the shallowest

About the Author

While in high school, Peter Lehmann received some basic instruction in electronics. However, he did not return to involvement with electronics on his own time until several years after graduating with a four-year liberal arts degree from college in 1974. For several decades now, electronics has been a strong avocational interest of his, with a focus on audio and analog circuitry. Nevertheless, he does have several books on his shelf dealing with digital electronics.

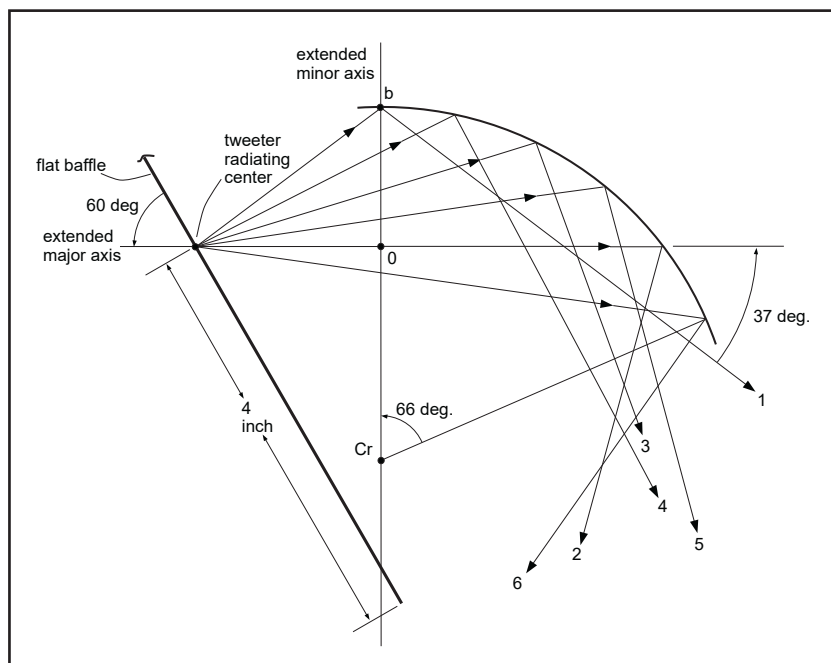


Figure 7: A reflector of circular curvature that closely approximates the gentler curvature of the ellipse shown in Figure 4 is substituted for the elliptical reflector shown in Figure 6. The substitution results in none of the reflected rays at a very shallow angle to the major axis and the baffle for the tweeter at a 60° angle to the major axis can be 4" long below the major axis without obstructing reflections off of the nearby reflector.

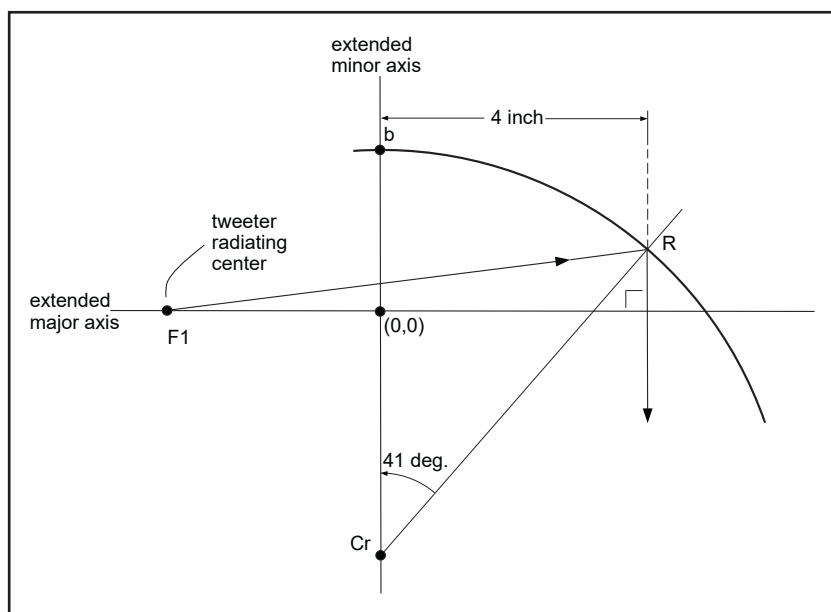


Figure 8: In the tweeter/reflector arrangement shown in Figure 7, the reflected ray that is perpendicular to the major axis is 4" distant laterally from the minor axis. In my loudspeaker system, the reflector is positioned above the woofer causing the perpendicular ray to be on the same vertical plane that the radiation axis of the woofer lies on.

desirable. Ray 2 is reflected at a shallow angle to the major axis and most likely would be obstructed by the baffle for the tweeter or restrict the extent to which the baffle could extend forward of the major axis. Ray 3 is reflected perpendicular to the major axis.

In drafting, one technique for approximating the curvature of an ellipse is called the four-center method. This method joins two circular arcs of larger radius to two circular arcs of a smaller radius. **Figure 7** is a to-scale drawing of reflecting sound from the tweeter by a reflector with a circular curvature. The curvature of the reflector shown in Figure 7 has the larger radius curvature of the four-center method applied to the ellipse shown in Figure 4.

The equation for calculating the radius of the larger radius arc closely equivalent to a gentler curvature of an ellipse is:

$$R_{LG} = \frac{\sqrt{a^2 + b^2} (a - b + \sqrt{a^2 + b^2})}{2b}$$

where:

R_{LG} = radius of the large equivalent circle

a = length of the semi-major axis of the approximated ellipse

b = length of the semi-minor axis of the approximated ellipse

The radius center of the large equivalent circle, Cr , lies on the extended minor axis below the major axis at a distance from the 0 point equal to R_{LG} minus b .

The angle of reflection of rays 1-6 shown in Figure 7 was solved graphically following the law of the angle of reflection equal to the angle of incidence. As is not the case for the elliptical reflector shown in Figure 6, in the arrangement shown in Figure 7, reflection in the fourth quadrant is still largely in a forward direction.

Figure 7 graphically demonstrates none of the reflected rays off of the reflector at an angle of less than 37° making the arrangement directional to the desired degree. The figure also shows graphically that the 4" length of the flat baffle below and at a 60° angle to the major axis does not obstruct ray 6 reflected at the shallowest angle towards the baffle.

Despite the cylindrical curvature of the reflector shown in Figure 7 extended beyond where it closely approximates the ellipse, the summation of the length of each incident ray from the tweeter to the curvature plus the length of the resultant reflected ray to the major axis is closely equal to 4.5".

Based on the configuration of the tweeter and

the reflector shown in Figure 7, **Figure 8** shows that the reflection at a 90° angle to the major axis occurs at a distance of 4" from the minor axis. This corresponds to the angle of the radius at the point of reflection at an angle to the minor axis equal to 41° . So that directionality of reflections off of the reflector is similar to directionality of radiation of the woofer, the reflector is positioned above the woofer with the minor axis of the reflector 4" to one side of the woofer's axis of radiation.

10 ms Delay of First Sidewall Reflection

In **Figure 9**, the degrees of arc of the reflector shown in Figure 7 equal to 66° has been extended on each side by 12° for an increase to 90° . The increase is done so that the 66° of arc between the points of reflection of ray 1 and ray 6 shown in Figure 7 occur a short distance in from the outside edges of the reflector.

In order to know how much toe-in my pair of loudspeaker systems should be given, I needed to know the angle of the median ray shown in Figure 9. The median ray is intermediate between first and second rays and is made the ray to be the first reflection arriving to the listener off of the far

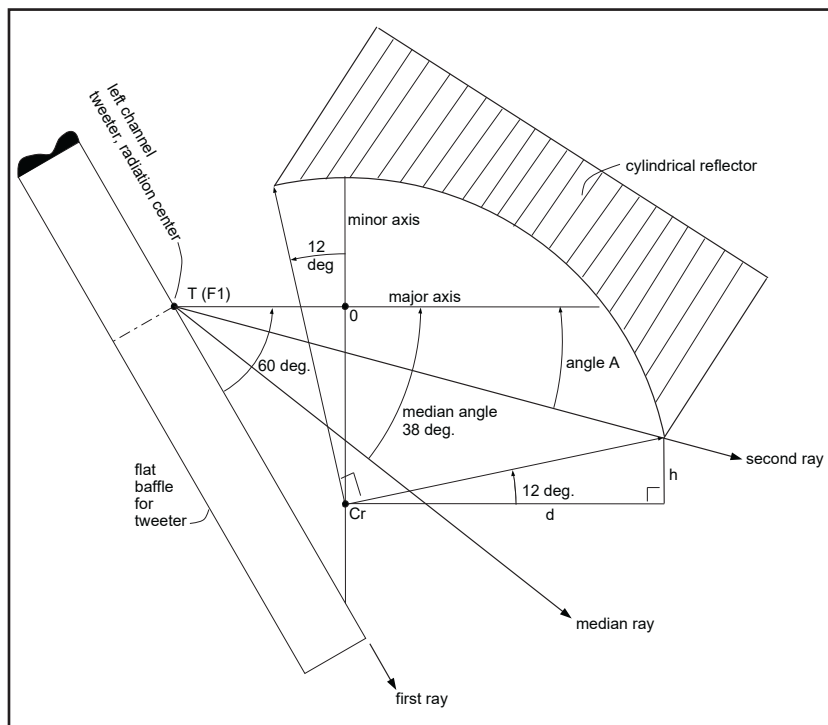


Figure 9: For the constructed reflector, the concavity of the reflective surface shown in Figure 7 is symmetrically increased to 90° . Thereby, ray 1 and ray 6 shown in Figure 7 reflect slightly inward of the edges of the constructed reflector. The median ray is in middle of forward sound rays emanated from the tweeter not reflected off of the nearby reflector.

A Comprehensive Course in Speaker Design

Get your copy today!

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

Electronic Crossovers:

Solid State, Tube,
Passive Line Level
6,12,18,24,32,48 dB/octave
Butterworth, Linkwitz-Riley, etc.

XM44 2-way crossover

Custom Speaker Crossovers:

Potcore Inductors, Toroidal Inductors
6,12,18,24 dB/octave

Marchand Electronics Inc.

(585) 423 0462 www.marchandelec.com

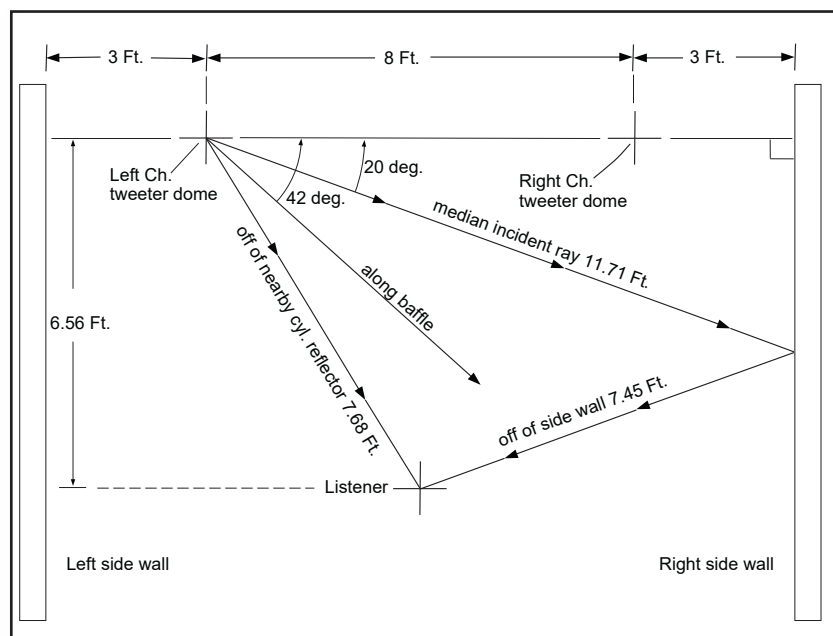


Figure 10: Left and right channel tweeters mounted on top of the cabinets holding the woofer of each of my loudspeaker systems are placed 8' apart in a 14' wide room. Toe-in of the left channel system cabinet at an 18° angle results in a 10-ms delay of arrival of the first reflection from the right sidewall of the room, allowing the room acoustics to add spaciousness of reproduction.

sidewall. The angle of the median ray equals one-half times the angle of the baffle plus one-half times angle A. Angle A can be calculated from the distance of the center of the tweeter from the minor axis, along the major axis, the radius of curvature of the reflector, and the lengths d and h. Angle A equals 15°. The angle of the median ray with respect to the major axis is calculated equal to 38°.

Figure 10 shows a pair of the left channel tweeter/reflector assembly shown in Figure 9

positioned in a room that is 14' wide. The right channel assembly is the mirror image of that of the left channel. The center of radiation of the left and right channel tweeters are laterally 8' apart and centered in the middle of the room. The radiation centers of the pair of tweeters lie on a line that is perpendicular to the sidewalls of the room. The listener is positioned laterally centered in the room and 6.6' back from the line joining the tweeters.

The median incident ray shown in Figure 10 emanating from the left channel tweeter is at a 20° angle to the line joining the pair of tweeters. This is accomplished as shown at **Figure 11** by toe-in of the front face of the left channel system-cabinet at an 18° angle to the line joining the radiation center of the tweeters and perpendicular to the side walls.

In the assembly of the pair of my loudspeaker systems, the extended major axis of each loudspeaker of tweeter/reflector lies on the vertical plane of the front baffle of the supporting cabinet. Figure 11 shows the right channel tweeter/reflector assembly as the mirror image of the left channel assembly.

The baffle shown in Figure 9 is at a 60° angle to the major axis of the assembly. The major axis of the assembly lies on the vertical plane of the exterior front panel of the supporting cabinet. Thus, toe-in of the cabinet at 18° results in the angle of the sound ray along the surface of the tweeter's baffle shown in Figure 10 at an angle of 42° to the line perpendicular to the side walls. This ray cannot arrive directly to the listener, negating any possibility of muddying the short delay of time of arrival of sound of the tweeter by that sound reflected off of its nearby reflector to the listener.

In Figure 10, the length of the sound ray off of the reflector near the left channel tweeter to the listener is very nearly 7.7' long. The median incident

Resources

E. Geddes, "The Geddes Loudspeaker System Design Philosophy," www.gedlee.com/Papers/Philosophy.pdf.

S. Linkwitz, "Shaped tone-burst generator," Linkwitz Lab,
www.linkwitzlab.com/sys_test.htm.

W. Luzzader, *Fundamentals of Engineering Drawing*, Prentice-Hall, Inc., Engelwood Cliffs, New Jersey, 1977.

J. Murphy, "Loudspeaker Diffraction Loss and Compensation," Topic No. 2, True Audio, February 2017, trueaudio.com/st_diff1.htm.

H. F. Olson, "Direct Radiator Loudspeaker Enclosures," *Journal of the Audio Engineering Society*, Volume 17, No. 1, October, 1969

J. Rindel, "Reflection of Sound from Finite-Size Plane and Curved Surfaces," Technical University of Denmark, 2005.

“Tables of random-incidence absorption coefficients, α ”
cds.cern.ch/record/1251519/files/978-3-540-48830-9_BookBackMatter.pdf

F. Toole, *Sound Reproduction*, Focal Press, New York & London, 2008.

Sources

Newark 8" Woofer, Part Number 55-1190
Newark element 14 | www.newark.com

Goldwood GT-525 silk dome tweeter
Parts Express www.parts-express.com

DAP 4-lbs carton of plaster of Paris
Lowe's Home Improvement | www.lowes.com

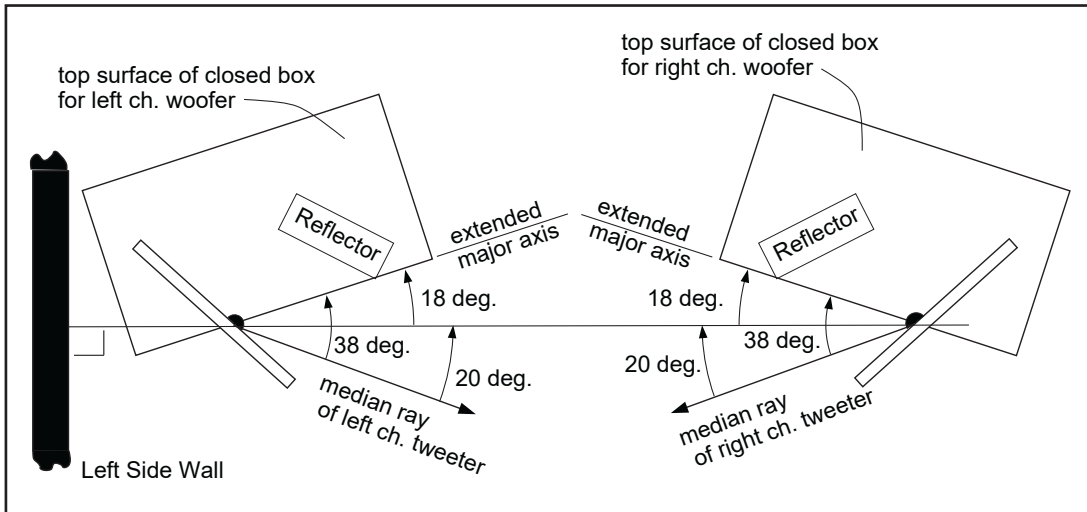


Figure 11: Given the left channel assembly of tweeter and integrated reflector shown in Figure 9 and the major axis of that arrangement lies along the top front edge of the supporting cabinet, then toe-in of that cabinet at 18° to a line perpendicular to the left sidewall results in the median ray of sound from the tweeter at a 20° angle to that line. The arrangement of the right channel tweeter/reflector and the median sound ray emanating from the tweeter of that assembly is the mirror image of the left channel arrangement.

ray and that ray reflected off of the right sidewall to the listener are, respectively, 11.7' and 7.5' long; the sum of these lengths equals 19.2'. Thus, the median ray reflected off of the right sidewall has traveled a distance difference of 19.2' minus 7.7' equals 11.5'. The speed of sound at 70°F equals 1.13' per millisecond. Furthermore, the first reflection off of the sidewall arrives to the listener 10 ms

after the direct sound (off of the nearby reflector) has arrived. The delay of 10 ms allows the room to impart spaciousness to the reproduction.

In the second part of this two-part article series, I describe the construction and the theoretical reflectivity of a nearby reflector as well as how to build the towers. Then, I put the reflective-way, two-way loudspeaker system to the test. 

For Over 35 Years

ACOustics begins with ACO™



Measurement
Microphones
Polarized and Electret

Community/Industrial
Noise Monitors/Alarms
Indoors and Out



Windscreens
3 to 18 inch & Custom
Weather/UV Resistant

Microphone Systems
Standalone
IEPE Powered
Phantom Power
Custom

ACO Pacific, Inc
www.acopacific.com sales@acopacific.com
 Tel: +1-650-595-8588

Personalize Your Sound Environment



AEQ ONE

AS SEEN ON KICKSTARTER
ASYMMETRIC EQ HEADPHONE AMP





We also design and develop
an extensive line of tactile audio
transducers and marine speakers.

Vidsonix Design Works
 508 Performance Road, Mooresville, NC 28115
sales@vidsonix.com

More info
at: **VIDSONIX.com, asymmetricEQ.com**

©2001~2018 VIDSONIX DESIGN WORKS. All Rights Reserved.

Back EMF Phase Relationships in Moving-Coil Loudspeakers (Part 4)

Back EMF Reactive Impedances and Energy Losses

In this article, Andy Lewis expands upon the Back EMF concepts hinted at in the previous three articles in this series. The theories can be applied to virtually all areas of loudspeaker design, starting with two filters: Simultaneous Electrical and Newtonian Systems.



By
Andy Lewis
(Acme Sound, LLC)

In the first installment of this series, which ran in the May 2018 issue of *audioXpress*, we identified and quantified Back EMF at and around the resonant frequency of a moving-coil loudspeaker, and modeled it using a simple LCR filter, as shown in **Figure 1**.

In Figure 1, the 3.5 Ω resistor represents the voice coil's DC resistance. The RCL parallel filter models the Back EMF, as previously described. A moving-coil loudspeaker, however, is not composed of one resonant filter, but of two. It is not a coincidence that these two resonant filters share the same frequency of resonance. It was observed in the first article in this series (*audioXpress*, May 2018) that:

$$LC = MC_{ms} \quad (1)$$

where:

L = inductor in LCR model
C = capacitor in LCR model
M = mass of loudspeaker moving assembly
 C_{ms} = loudspeaker's compliance

We will examine this convergence.

What are Electrical and Newtonian Filters?

Because the previous articles in this series were about Back EMF specifically, they overlooked the primary resonant filter, which is made up of the loudspeaker's moving mass and compliance. We will call this resonant filter the Newtonian Filter. These physical characteristics of the driver determine its free-air resonance, according to this common equation:

$$f_s = \frac{1}{2\pi\sqrt{(MC_{ms})}} \quad (2)$$

where:

f_s = frequency of resonance

M = moving mass

C_{ms} = compliance

Because the loudspeaker's resonant frequency is determined by these Newtonian properties of mass and compliance, the resonant frequency is, as well. The filter created by Back EMF is modeled using a resistor, a capacitor, and an inductor (see Figure 1). We will refer to this LCR filter as the Electrical Filter (Back EMF). The resonant frequency of this filter is necessarily the same as, and determined by, the characteristics of this Newtonian filter, as in Equation 2. For the purposes at hand, the terms "Electrical Filter" and "Back EMF" can be considered interchangeable. The equation relating capacitance, inductance, and resonant frequency of an electrical filter is:

$$f_s = \frac{1}{2\pi\sqrt{LC}} \quad (3)$$

where:

f_s = frequency of resonance

L = inductance

C = capacitance

The two equations, 2 and 3, are obviously analogous. This is because the reactive Newtonian quantities are an analog to the electrical reactive components making up the LCR filter. We will examine the intimate relationship between these two filters.

An electrical LCR filter has both a capacitance and an inductance. A loudspeaker's physical structure has both a physically inductive moving mass and physically capacitive spring compliance. We call the moving mass inductive because, like an inductor, it has a high reactance at high frequencies, in that it's difficult to accelerate at high frequencies. We call the loudspeaker's spring capacitive because, like a capacitor, it has a high reactance at low frequencies. Both of these physical components store energy, and return it to the system, just as do the electrically-reactive components making up the Electrical Filter.

The Electrical Filter

We have seen that the impedance of a loudspeaker can be modeled using passive components. From the point of view of an AC voltage, the speaker exhibits inductive and capacitive

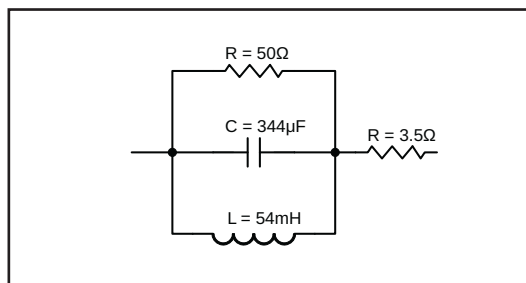


Figure 1: Specific values of the LCR model of a Fisher woofer

reactances and resistance, just as the LCR model does, and the impedance curves are nearly identical (see **Figure 2**)

Capacitors and inductors are devices which store energy and release it back to the system. This energy is stored in the form of concentrated electric charge in capacitors, and in the form of external magnetic fields in and around inductors. As the frequency applied to a loudspeaker driver is increased, the impedance of the unit increases below resonance, just at the impedance of the LCR model does. (For more detailed explanations see the previous articles in this multi-article series, which ran in *audioXpress* May, June, and July 2018.) This rising impedance is, by definition, inductive. Similarly, above resonance, the impedance is decreasing, which is characteristic of a capacitance.

As we have seen, Back EMF behaves as if it were an electrical filter, even in the absence of these reactive and resistive passive components. In the LCR model, energy is dissipated as heat by the series resistor representing the DC resistance of the voice coil and by the resistive component of the LCR model. At any frequency, the energy

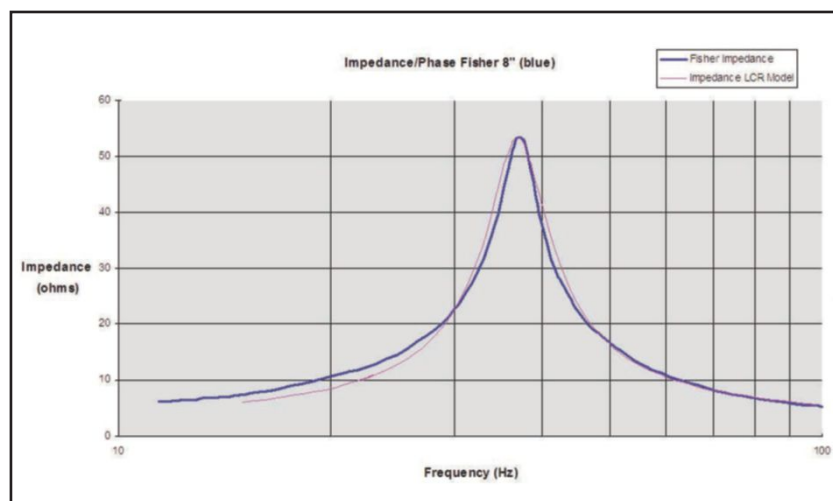


Figure 2: This graph shows the impedances of the loudspeaker and of the LCR model.

lost to heat is equal to the RMS current squared times the total DC resistance:

$$E = I^2 R \quad (4), \text{ also used below}$$

where:

E = energy lost to heat

I = RMS current

R = DC Resistance

The Newtonian Filter

In the LCR model, energy storage and retrieval are to and from the electrostatic and the magnetic fields of capacitors and inductors. The components of the physical loudspeaker assembly also store and release energy, but through a different mechanism—energy is stored in and released from a different type of reactive filter.

This filter and its storage of energy involves quantities we associate with Isaac Newton and his Laws of Motion: mass, velocity, and spring characteristics. In the moving-coil loudspeaker, energy is stored in, and released from, the moving mass and the spring compliance. As a speaker cone moves back and forth, kinetic energy is stored in, and returned from, the moving mass. Similarly, potential energy is stored in, and returned from, the surround and the spider, as a speaker's moving assembly oscillates.

As this activity is occurring, some energy is lost to frictional heat. Frictional losses can be seen as resistive components, because they dissipate heat, just as a resistor does. More on this later. The moving mass of a loudspeaker's cone and the spring force of the compliance result in a physical resonance, just as a capacitor and inductor form an electrical one.

Just as the electrical filter is made up of a reactive capacitor and a reactive inductor, the Newtonian filter is made up of a reactive mass, and a reactive compliance (with frictional resistance). Just as an electrical inductance impedes high frequencies in the electrical filter, the inductive nature of the loudspeaker's moving mass impedes the cone's high-frequency acceleration. This can be seen as a physical inductance.

Similarly, the loudspeaker's compliance

impedes the cone's low-frequency acceleration, just as a capacitor impedes low frequencies in the electrical filter. This can be seen as a physical capacitance. As energy is stored in, and returned from, the mass and the compliance, it results in impedance characteristic essentially identical to that of the electrical filter made up of a capacitor, an inductor, and a resistor. This Newtonian filter has a resonant frequency. In the world of loudspeakers, this is referred to as f_s . The equation relating mass, compliance, and resonant frequency of this physical filter is shown in Equation 2.

Intersection of Two Filters

Because Electrical and Newtonian filters share a common resonant frequency, they have a mathematical similarity:

$$f_s = \frac{1}{2\pi\sqrt{(LC)}} = \frac{1}{2\pi\sqrt{(MC_{ms})}} \quad (5)$$

and algebraically that:

$$\frac{1}{\sqrt{(LC)}} = \frac{1}{\sqrt{(MC_{ms})}} \quad (6)$$

And ultimately:

$$LC = MC_{ms} \quad (7)$$

This quantity is expressed in "seconds squared." The resonant frequencies of the two filters coincide. But because f_s is determined by the physical quantities of mass and compliance, f_s is not dependent on the electrical filter, but only the Newtonian one. Obviously, then, as you add mass to the cone, the effective capacitance and the inductance of the electrical filter change to equal the lower resonant frequency of the altered Newtonian filter.

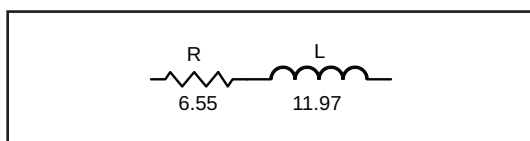
Similarly, when you install the driver in a sealed enclosure, the electrical filter would have to change effective component values to model the decreased compliance, and the increase in Newtonian resonant frequency, in turn.

Observation on Reactive Parallel Electrical Filters as Pertains to Back EMF

We have previously shown that the Back EMF of the Fisher woofer can be modeled by using a parallel filter made up of a 50 Ω resistor, a 344 μF capacitor, and a 54 mH inductor. The DC resistance of the voice coil is 3.5 Ω .

At f_1 , 23 Hz, the inductive reactance of the parallel LCR filter in is 11.97 Ω , and the effective

Figure 3: This is the RL model of the Fisher woofer at 23 Hz.



DC resistance is 6.55 Ω . Because, at this frequency, the impedance is inductive, it can be modeled by a simple resistor and inductor, as in **Figure 3**.

Yes, you read correctly. At 23 Hz, the impedance of the two components in Figure 3 is the same as that of the more complex filter in Figure 1, and as that of the Fisher woofer itself! At this frequency, this inductive reactance represents an inductance of an inductor of 82.8 mH. What is interesting is that while the actual inductor in the LCR filter is 54 mH (see Figure 1), the apparent inductance at that frequency is much greater.

This is due to the bizarre nature of parallel reactive filters, in which currents add as vectors. Therefore, because the currents in the capacitor and in the inductor are traveling in opposite directions at any instant, the phase angle between their impedances is 180°. The currents are exactly out of phase with one another. Therefore, below resonance, the capacitor in the LCR model has an effective negative-capacitance, which is equivalent to a positive inductance, and therefore adds to the net inductance of the filter. In other words, the opposite current in the capacitor offsets the current in the inductor, resulting in lower net

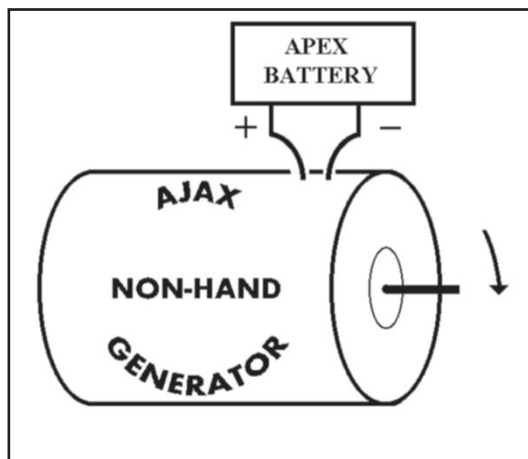


Figure 4: A "Generator" driven by an external voltage is a motor.

current. Less current equals higher impedance (Ohm's law). Therefore, yes, it is possible for the filter's effective inductance to be greater than the actual inductance of the inductor, even though it's completely counter-intuitive. Because of this, the effective inductance of the LCR filter below resonance will always be greater than the inductance of the physical inductor itself.

Similarly, because of the "mirror image" nature of capacitors vs. inductors, the capacitance of the

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 Circuit Cell Design by Barry McClure

cc-webshop.com

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

LCR filter above resonance will always be less than that of the actual capacitor in the LCR model. In this case, at $f_2 = 59.3$ Hz, the capacitance due to the 11.97Ω capacitive reactance is $224 \mu\text{F}$, for the same reason: Opposite parallel currents result in less net current, which translates to higher impedance.

Energy Lost to Back EMF: Another Look at the DC Motor

In order to examine how energy is lost to Back EMF in a loudspeaker, it is helpful to return to the model of the DC rotary motor, that we used as in the first article of this multi-part series (*audioXpress*, May 2018). We will use the same motor we used as our previous example and will drive this motor with no external load. The only physical resistance to its rotation will be in the friction of its bearings, which warm slightly as heat is generated by this friction.

The total resistance of the DC motor is equal to the internal resistance of the motor plus the effective resistance of the Back EMF generated as the motor turns (see article 1 of series). Back EMF manifests itself as, and can be expressed and measured as, a resistance. When this resistance is added to the motor's inherent resistance, the sum equals the total resistance of the working motor, as shown in **Figure 5**.

$$R_1 + R_2 = R_t \quad (8)$$

Add Friction or Load to the DC Motor: Increased Load Results in Decreased Back EMF

When there is no physical load on the motor, it turns very quickly for a given DC voltage. Because voltage is proportional to velocity, Back EMF is at its maximum, as is total resistance, in turn. Because of this high resistance, current flow is very low. Consequently, very little energy is

dissipated as heat by the wire windings, or the bearings. In the case of a low-friction motor with no load, very little heat is generated, as very little energy is lost.

With no load, the Back EMF has a greater resistance than the windings themselves, and more energy is lost to the Back EMF than to the heat dissipated by the windings (I^2R). The heat dissipated by the Back EMF component of the motor's resistance is in the friction of the bearings. The DC Motor, when not presented with an external load, is a device that dissipates very little heat because of its maximized resistance and low current flow.

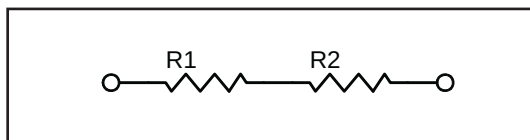
If the motor is called upon to drive an external load, such as a fan, the external load will make the motor harder to turn. Internal friction has the same effect. Any frictional loss will result in heat, either within the motor itself, or externally, such as in the air being stirred up by the fan (this observation is relevant to loudspeakers). When the friction increases, the Newtonian motor force driving the coils is opposed by this friction. This decreases the net force on the rotating armature, for a given current flow, decreasing its velocity. Because Back EMF is proportional to velocity, less velocity equals less Back EMF, resulting in less resistance at the terminals. Therefore, Back EMF becomes a smaller component of total electrical resistance.

As frictional force increases to infinite (enough, in other words, to cause all activity to cease), Back EMF becomes zero, and the total resistance becomes equal to the resistance of the windings themselves. The result is maximum current flow, heat dissipation in turn, and risk of fire. That's why loading a motor down with a greater load than it can handle can burn it up; because of the loss of Back EMF.

Under normal circumstances, with the DC motor driving a reasonable load, the heat energy dissipated is equal to that lost to current through the resistance of the coils, the heat dissipated in the bearings by the resistance of the Back EMF, plus the heat dissipated by the load. The all-encompassing term "load," would include internal friction, and whatever external entity is being driven by the motor (e.g., a wheel, a propeller, or a lawn mower). The motor does Newtonian work on these loads, and heat is dissipated externally as a consequence, to various losses.

To summarize: For the DC Motor, when the load is minimized, Back EMF is at its maximum, and DC current is at its minimum. As frictional losses increase (including externally) they are manifested in a decreased Back EMF component

Figure 5: The DC motor is represented as two resistors.



About the Author

Andy Lewis lives in Englewood, CO, plays drums, and produces "Acme Sound Low-B" low-coloration loudspeakers for bass players. He is also involved in Special Olympics and other disability activities with his son, Collin.

of the motor's total resistance. Current is increased proportionally, as described by Ohm's Law:

$$I = E/R \quad (9)$$

But while current is inversely proportional to total resistance, energy dissipated (power) is proportional to the square of the current.

$$E = I^2R$$

where:

E = energy

I = current

R = resistance

Ironically then, energy lost to friction, reflected in decreased Back EMF, actually increases total energy lost to that Back EMF, because lower resistance increases current. To summarize, all energy losses not associated with the I^2R energy lost to coil resistance is manifested in the resistive component of the Back EMF ($R_1 + R_2 = R_L$).

The resistance of the Back EMF decreases with

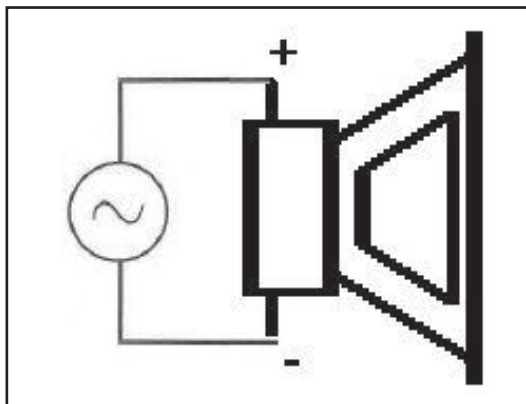


Figure 6: In the moving-coil loudspeaker the AC linear generator is driven by external voltage.

increased load or internal friction, even as total energy lost to this friction increases. All of these things are true of the moving-coil loudspeaker, which we referred to as the "AC Linear Motor," in previous articles in this series.

How Does This Translate to the Loudspeaker?

Figure 6 shows the moving-coil loudspeaker. To a voltage source, just as with the DC Motor, all energy losses are resistive and seen as the resistive

Spiral Airtm
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

**Ready, willing
and
Avel**

**offering an extensive range of ready-to-go
toroidal transformers
to please the ear, but won't take you for a ride.**

Avel Lindberg Inc.
47 South End Plaza, New Milford, CT 06776
p: 860.355.4711 / f: 860.354.8597
sales@avellindberg.com • www.avellindberg.com

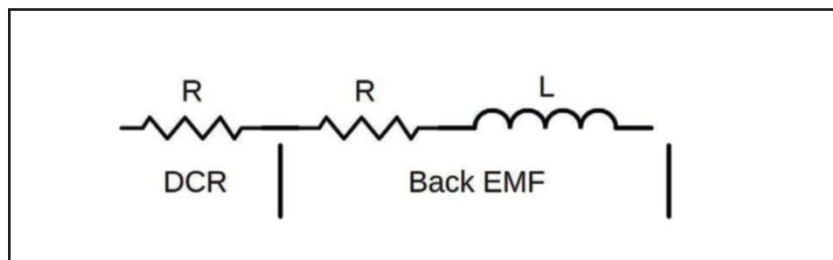


Figure 7: R_{DCR} and $R_{Back\ EMF}$ are shown in series with a reactive component of Back EMF.

component of the total impedance. This applies to the DC Motor, the moving-coil loudspeaker, and the LCR model of the loudspeaker's impedance. All energy (heat) dissipated by these systems is reflected in the resistive component(s) of their impedances.

The total resistance (not impedance) of the moving-coil loudspeaker is the sum of its voice-coil resistance and the resistive component of the Back EMF. To an amplifier, there is no difference between the resistive component of Back EMF and an actual resistor. Therefore, the energy not lost to voice coil DC resistance is necessarily dissipated by the resistive component of Back EMF, just as with the DC Motor, or any two resistances in series, as shown in **Figure 7**. Therefore:

$$E_t = E_{DCR} + E_{Back\ EMF} \quad (10)$$

and

$$E_{Back\ EMF} = E_t - E_{DCR} \quad (11)$$

where:

E_t = total energy dissipated

E_{DCR} = energy lost to voice coil resistance

$E_{Back\ EMF}$ = energy lost to the resistive component of Back EMF

Where Does Energy Lost to Back EMF Go?

Energy can neither be created nor destroyed, but can be converted to another form. In electrical motors it is converted to heat. In the case of the moving-coil loudspeaker, some of this heat is dissipated in the room, as your favorite music excites the air with sound waves. In the loudspeaker, as with the DC Motor, energy is dissipated as heat through frictional losses. Mechanical losses involve friction in various forms. Energy is lost in the surround, for example, and even to "wind" in the vicinity of the driver.

Some of these energy losses are audible, such as air being pushed through the gap, or through a vent in the dustcap. Air leaks in the enclosure behave similarly, and most are audible. If damping materials are used in an enclosure, energy is lost to this absorptive material, and is dissipated as heat. The reproduction of music warms the air in the room ever so slightly. All of these losses are manifested in the resistive component of the impedance.

It is possible to analyze the activity through a current cycle at any frequency to see when, and to what extent, the Newtonian force from the voice coil is opposed by the motion of the cone. At these moments in the current cycle, Newtonian work is done, the cone is slowed, and energy is dissipated. In the specific case of energy lost to this Newtonian work, the heat is dissipated in the voice coil by the resistive component of Back EMF. Just as an increased external load causes the wire windings of the DC Motor to get hotter, the work done during various parts of the current cycle cause the current through a loudspeaker's voice-coil to increase, and heat is dissipated, in turn.

What the DC Motor and the AC Loudspeaker have in common is that these energy losses all result in an increased physical load to the AC motor. This load decreases velocity, for a given input voltage, resulting in decreased Back EMF. The impedance seen by the voltage-source is decreased as Newtonian work is increased. Just as with the DC Motor, the impedance drops, resulting in increased current, and more heat dissipated into the environment.

Differences Between DC Motor and Loudspeaker

There is no energy storage in the bearings of a rotary motor, and no energy returned to the system. The impedance of the DC Motor is purely resistive, and much less complicated, as described in previous articles in this series. As an alternating device, the loudspeaker driver is

References

A. Lewis, "Back EMF Phase Relationships in Moving Coil Loudspeakers (Part 3): "Slip" Creates Inductive, Capacitive, or Resistive Back EMF," *audioXpress*, July 2018.

—, "Back EMF Phase Relationships in Moving Coil Loudspeakers (Part 2): Thought Experiments, *audioXpress*, June 2018.

—, "Back EMF Phase Relationships in Moving Coil Loudspeakers (Part 1): Identifying, Quantifying, and Modeling Back EMF, *audioXpress*, May 2018.

The "Must Have" reference for
loudspeaker engineering professionals.

Home, Car, or Home Theater!

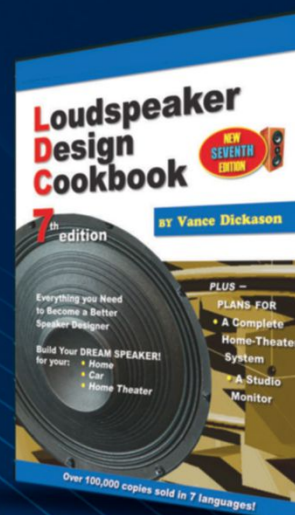
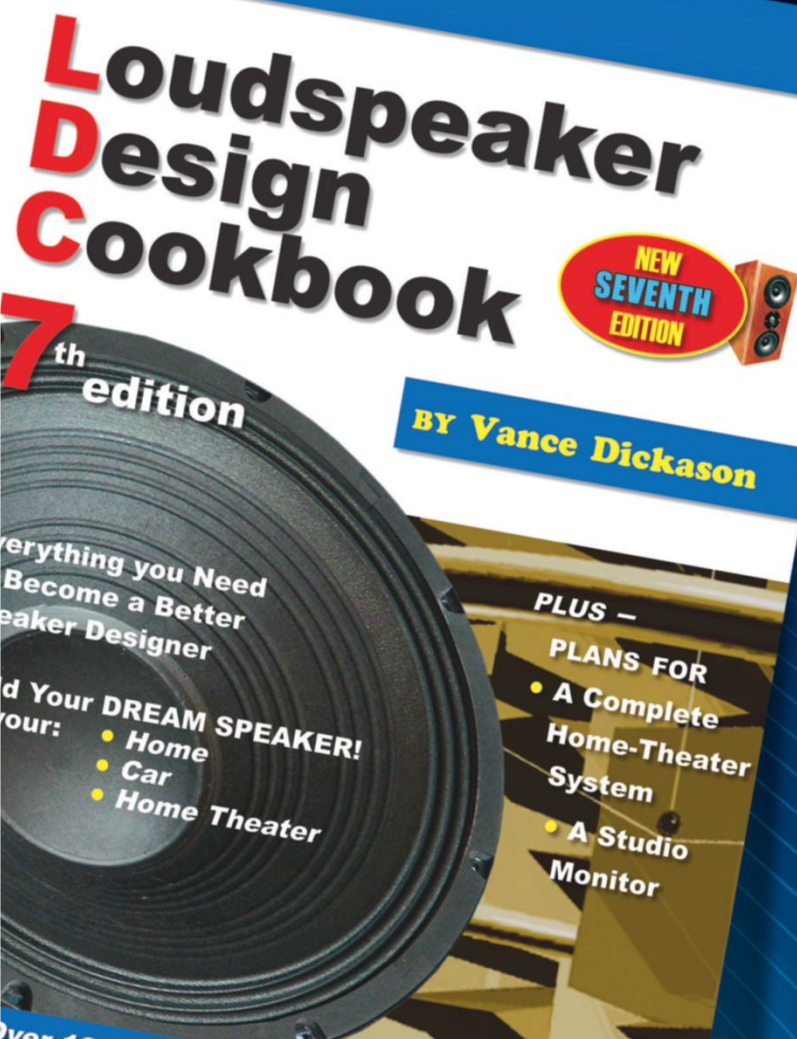
Back and better than ever, this 7th edition provides everything you need to become a better speaker designer. If you still have a 3rd, 4th, 5th or even the 6th edition of the *Loudspeaker Design Cookbook*, you are missing out on a tremendous amount of new and important information!

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.



	f_1	f_s	f_2
Frequency	23 Hz	36.93 Hz	59.3 Hz
Z_t	13.64 Ω	53.5 Ω	13.64 Ω
Z_t Phase (RAD)	1.07	0	-1.07
Z_t Complex Impedance (ohms)	6.55 + j11.97	53.5 + j0	6.55 - j11.97
EMF Complex Impedance (ohms)	3.05 + j11.97	53.50 + j0	3.05 - j11.97
EMF Magnitude	12.35 Ω	50 Ω	12.35 Ω
EMF Phase Angle (RAD)	1.32	0	-1.32
R_t	6.55 Ω	53.5 Ω	6.55 Ω
$R_{\text{Back EMF}}$	3.05 Ω	50 Ω	3.05 Ω
Voltage Applied	2.83 V	2.83 V	2.83 V
Current ($I = E/Z_t$)	0.207 A	0.053 A	0.207 A
Total Heat Dissipated I^2R	0.282 W	0.15 W	0.282 W
Dissipated by DCR	0.151 W	0.01 W	0.151 W
Dissipated by EMF	0.131 W	0.14 W	0.131 W

Table 1: These methods are used to isolate the resistive component of Back EMF at three frequencies and calculate the heat dissipated by Back EMF at these frequencies.

different from the DC Motor, in that the mass and compliance of the moving-coil loudspeaker have the ability to store and return energy. This results in the reactive component of Back EMF, and the high impedance peak we observe at resonance.

Electrical energy loss occurs when input current does Newtonian work on the cone, accelerating it in one direction or the other, or on the suspension, which alternately works with, or against the motor force. As with the DC Motor, an increased physical load will increase its I^2R loss as (electrical) resistance decreases and current increases, as was explained.

As with the DC Motor, there is work being done as the motor force is physically resisted variously by the inertia of the mass, the spring behavior of the suspension, and the various frictional losses. One of these frictional losses, thankfully, is the heat dissipated into the room as sound is produced by the speaker. (In an unpublished appendix, I have followed the moving assembly through its current cycle in each of the three modes of Back EMF generation, to illustrate how and when this Newtonian work is being done. Because there are three modes of Back EMF generation, as described in the first article of this series, three analyses are required to explore how and when this work is done by the loudspeaker in each of the three modes.)

Calculating Energy Lost to Back EMF

We have identified the loss of energy to the resistive component of Back EMF as a result of both frictional losses and Newtonian work. It can be interesting to see how much energy is lost to Back EMF at various frequencies. I have calculated these losses as they occur in our Fisher woofer. (Note that this woofer is in free-air. If the driver were actually mounted in an enclosure, there would be myriad other losses introduced. As was stated, each of these losses would ultimately be evident in the resistive component of the impedance.)

Just as with any current flowing through a resistance, the energy dissipated is equal to the resistance multiplied by the square of the current (reactive components return energy, as opposed to dissipating it):

$$E = I^2 R_{\text{EMF}} \quad (12)$$

where:

E = dissipated heat in watts

I = current in amperes

R_{EMF} = resistive component of Back EMF in ohms

The chart shown in **Table 1** uses the methods outlined in the previous articles in this series to isolate the resistive component of Back EMF at three frequencies, and calculates the heat dissipated by Back EMF at these frequencies. We arbitrarily use an input voltage of 2.83 V.

Interestingly, although the impedance of the speaker varies widely (from 13.64 Ω to 53.5 Ω), the energy dissipated by Back EMF is minimally changed! The difference between 0.140 W and 0.131 W is about 9.4%, or 0.277 dB, if you prefer. As the resistance of the Back EMF increases, the attendant decrease in current effectively offsets it, such that the power dissipated by the resistive component of Back EMF remains nearly constant.

In Closing

I hope you have enjoyed this series on Back EMF. As I stated up-front, I really don't see anything in this article series that would directly benefit a loudspeaker designer. I had just never seen anybody examine this activity. The important lesson to be taken from this article series applies not just to Back EMF, but to virtually all areas of loudspeaker design. There are many areas of loudspeaker design that involve frequency-dependant (complex) arithmetic. Back EMF is a filter. Bass enclosures are filters. Crossover networks are filters. Moving-coil drivers are filters.

An understanding of, and ability to perform, calculations using complex numbers can empower a designer in many ways. I developed my skills using complex impedance information in an effort to design accurate passive crossovers using fewer components, for my own selfish ends. My efforts were successful, and my passive crossovers are very good.

I never would have noticed how Back EMF behaves had it not been for this work on passive crossovers. An understanding of complex impedances and reactive filters enabled me to figure out how Back EMF behaves, and increased my understanding in other areas, as well. Time permitting, I intend to share some of the techniques I've developed to build crossovers in future articles. 

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

Why We Use "RMS" Current

Most people are aware that "RMS" stands for "root-mean-square." But that's not particularly illuminating. Here's why RMS quantities are so practical to use in the real world.

Ohm's Law is fundamentally about DC quantities: voltages, currents, and resistances (E, I, and R):

$$\begin{aligned} I &= E/R \\ E &= IR \\ R &= E/I \end{aligned}$$

Derivations of Ohm's Law pertaining to power or energy/time (E) are:

$$\begin{aligned} E &= I^2R \\ E &= E^2/R \end{aligned}$$

When dealing with AC, we could express quantities, such as voltage, in terms of peak voltage, peak-to-peak voltage, average voltage, or RMS voltage. So why do we so often use RMS when discussing AC quantities?

The answer is that when we are using RMS quantities, which are equal to 0.707 of peak, the simple equations of Ohm's law can be used for AC, just as they are used for DC. If we were to represent the same activity using, for example, peak voltages, the Ohm's law equations could no longer be used to describe the relationships between voltage, current, resistance, and power, without using correcting factors. Using RMS quantities makes our lives easier, which is why we like to express these quantities using RMS. However, it should be noted that this only applies to the resistive component of reactive impedances.

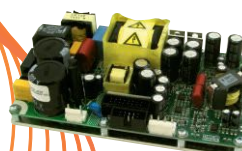
hypex NCxxxMP series

overview

The NCxxxMP amplifier module incorporates a low power standby power supply (meets 2013 ERP Lot 6 0.5W requirements), a highly efficient switch mode power supply and a high-performance Class D amplifier in one compact and easily applicable power brick.



NC122MP



NC250MP



NC252MP



NC500MP



NC502MP

Add-on Module NC100HF
The NC100HF is a dedicated tweeter-amplifier which fits the NCxxxMP series.



Highlights

- High efficiency
- Universal mains operation
- Flat, fully load-independent frequency response
- Low output impedance
- Very low, frequency independent THD
- Very low noise

Features

- One or two channel amplifier
- 5W standby SMPS
- Advanced over current protection
- External controlled operation
- Auto-switching (115/230V)
- Low weight
- Compact

Applications

- Monitor loudspeakers for recording and mastering studios
- Audiophile power amplifiers for professional and consumer use.
- Public address systems
- Active loudspeakers

ncore
audio power conversion

hypex electronics
www.hypex.nl

Screen-Grid Biasing

We compare several methods of setting screen-grid voltage in power amplifiers to see their distortion performance.

By
Richard Honeycutt
(United States)

Most tube enthusiasts were introduced to tubes in the order in which they were invented: diodes, triodes, tetrodes, pentodes, and then—maybe—pentagrid converters and other less common types. We learned how the control grid bias affects the plate current. We learned how control-grid-to-plate capacitance can introduce negative feedback at high frequencies, decreasing the gain at those frequencies. We came to understand that the screen grid is grounded from an AC standpoint, and thus, acts as a shield between the control grid and plate, greatly decreasing the control-grid-to-plate capacitance. The positive bias on the screen grid accelerates electrons moving from the control grid to the plate, reducing the dependence of plate current upon plate voltage.

Tetrodes and Pentodes

Even though we may never have used an actual tetrode, we might remember reading about the “kink” in a tetrode’s plate characteristic curve that acts as a negative resistance and can cause undesired oscillation. The kink was caused by electrons that were dislodged from the plate when other electrons struck the plate. These “secondary emission” electrons formed a space charge between the plate and the screen grid, and if the instantaneous plate voltage dropped below the screen grid voltage, electrons would be preferentially drawn to the screen grid, increasing screen grid current and decreasing plate current. “Unkinking” the tetrode was accomplished by adding the suppressor grid. Being biased at cathode potential, the suppressor grid repelled space-charge electrons back to the plate, eliminating the negative-resistance region in the plate characteristic.

However, many aficionados of tube audio have never worked with a tetrode, although some people choose to refer to beam-power tubes as tetrodes. We are most familiar with triodes, and to some extent, pentodes. Our understanding of pentodes may be limited to knowing that they can have a higher voltage gain than triodes, especially at high frequencies, and that they are often used as power amplifiers. And we know that the suppressor grid (or beam-confining electrodes in a beam power tube) is usually connected to the cathode.

Methods to Set Screen Grid Bias

We may also be a little shaky on our understanding of the effect of screen grid bias. **Figure 1** shows how little effect the plate voltage of a pentode has on the plate current, except at very low plate voltages. **Figure 2** shows the input characteristic (relation of plate current to control grid bias) for a 6AU6 small-

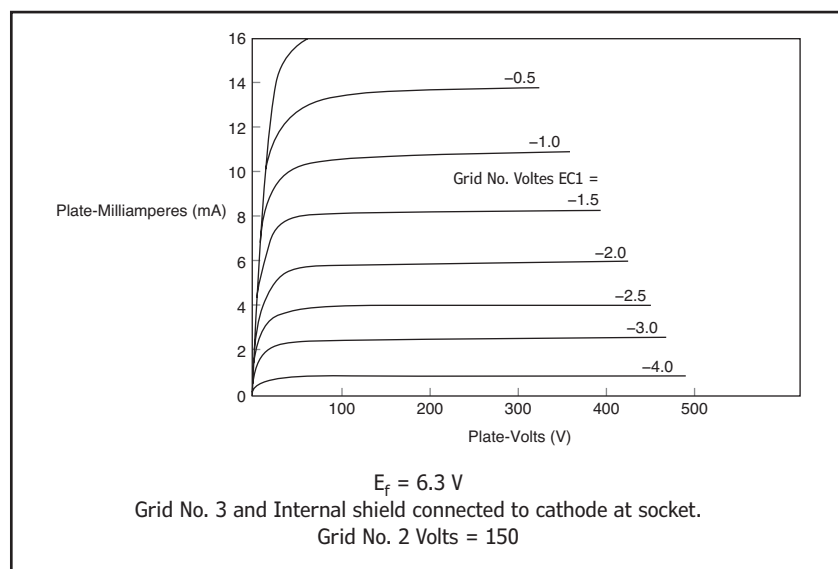


Figure 1: This is the plate characteristic of the 6AU6 small-signal pentode for 150 V on the screen grid.

signal pentode. The plate voltage is 250 V, but as I just pointed out, plate voltage does not have much influence on plate current in a pentode, as long as plate voltage is above about 50 V. Notice that the three sets of curves (representing screen grid voltages of 50 V, 100 V, and 150 V) have very similar shapes. The plate current (I_b) and screen grid current (I_{c2}) for a given control grid bias increase significantly as the screen grid voltage is increased. Above a plate current of about 4 mA, the transconductance:

$$\frac{\Delta I_b}{\Delta V_{g1}}$$

is fairly linear. In an audio amplifier, the screen grid voltage is usually selected to provide the desired gain. **Table 1** will give you an idea of typical values.

So far, we have concentrated on the DC bias of the screen grid. Usually, the screen grid voltage is derived from the B+ through a screen-dropping resistor, and the variations in screen-grid current—resulting from signal conduction in the pentode—are leveled out by a capacitor connected from screen grid to ground. If such a capacitor is not used, or if the capacitor opens, an AC signal will appear on the screen grid, and this signal will be out of phase with the control-grid signal, producing negative feedback. Deliberate application of a DC control voltage to the screen grid is sometimes used to create a Voltage Controlled Amplifier (VCA). VCAs are useful in limiters, compressors, and expanders.

Figure 3 shows the described screen-grid biasing circuit, which is the most common approach in small-signal amplifiers. **Figure 4** shows a voltage-divider bias approach. The voltage divider is designed so that the screen-grid current is only a small fraction of the total voltage-divider current; thus, variations in screen-grid current make little difference in the screen-grid voltage. It is possible to use a potentiometer for the voltage divider, permitting adjustment of screen-grid voltage and, hence, of stage gain.

In power amplifiers using pentodes or beam power tubes, the screen-grid bias can be supplied in any of three ways. **Figure 5** shows the screen grids biased through dropping resistors with the usual decoupling capacitor to ground. With this circuit operating at high output powers, the high B+ voltage may droop somewhat, causing the screen grid voltage to do the same. The resulting decreased gain can add some slight compression of the output signal. There could also be a slight increase in distortion in the output stage.

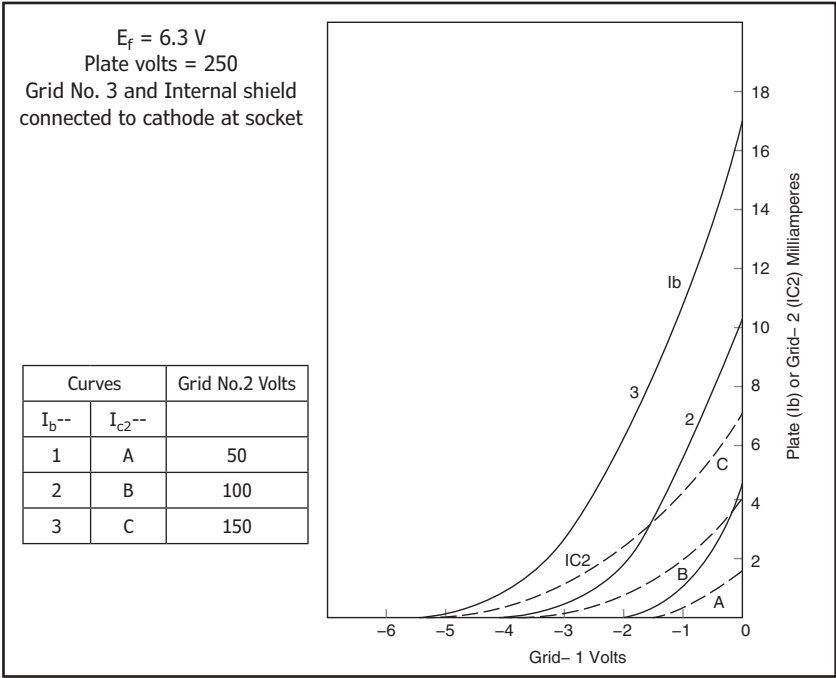


Figure 2: Here are the effects of screen grid voltage on input characteristic of a 6AU6 pentode.

Pentode Connection			
Plate Supply Voltage	100 V	250 V	250 V
Grid No. 3 and Internal Shield	Connected to cathode at socket		
Grid No. 2 Supply Voltage	100 V	125 V	150 V
Cathode Resistor	150 Ω	100 Ω	68 Ω
Plate Resistance (approximate)	0.5 MΩ	1.5 MΩ	1 MΩ
Transconductance	3900 μS	4500 μS	5200 μS
Plate Current	5 mA	7.6 mA	10.6 mA
Grid No. 2 Current	2.1 mA	3 mA	4.3 mA
Grid No. 1 Voltage (approximate) for plate μa = 10	-4.2 V	-5.5 V	-6.5 V

Table 1: Typical voltages and currents for a 6AU6 pentode in an amplifier, from the RCA datasheet for the 6AU6

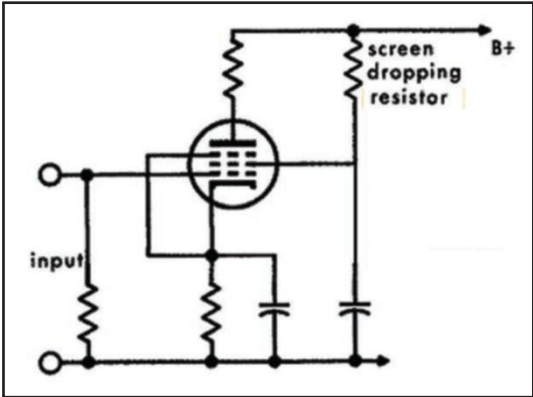


Figure 3: This screen grid is biased by a screen-dropping resistor and a decoupling capacitor.

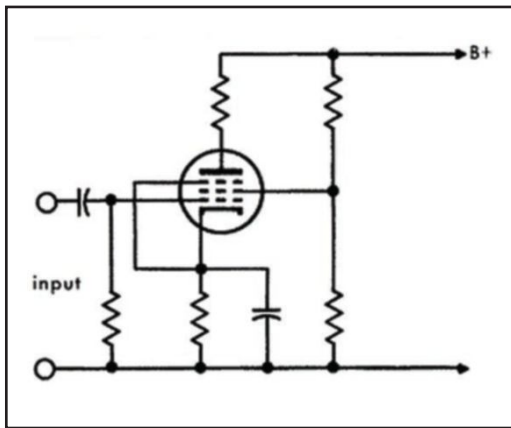


Figure 4: This image shows a voltage-divider screen bias.

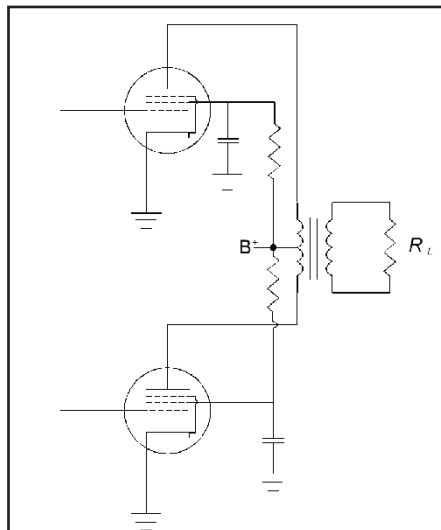


Figure 5: These screen grids are biased through dropping resistors.

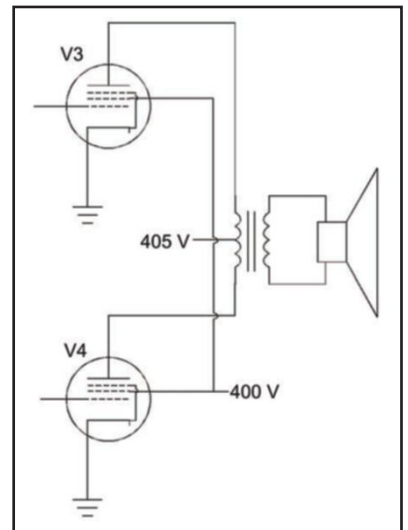


Figure 6: The screen grids are biased from a separate section of the power supply.

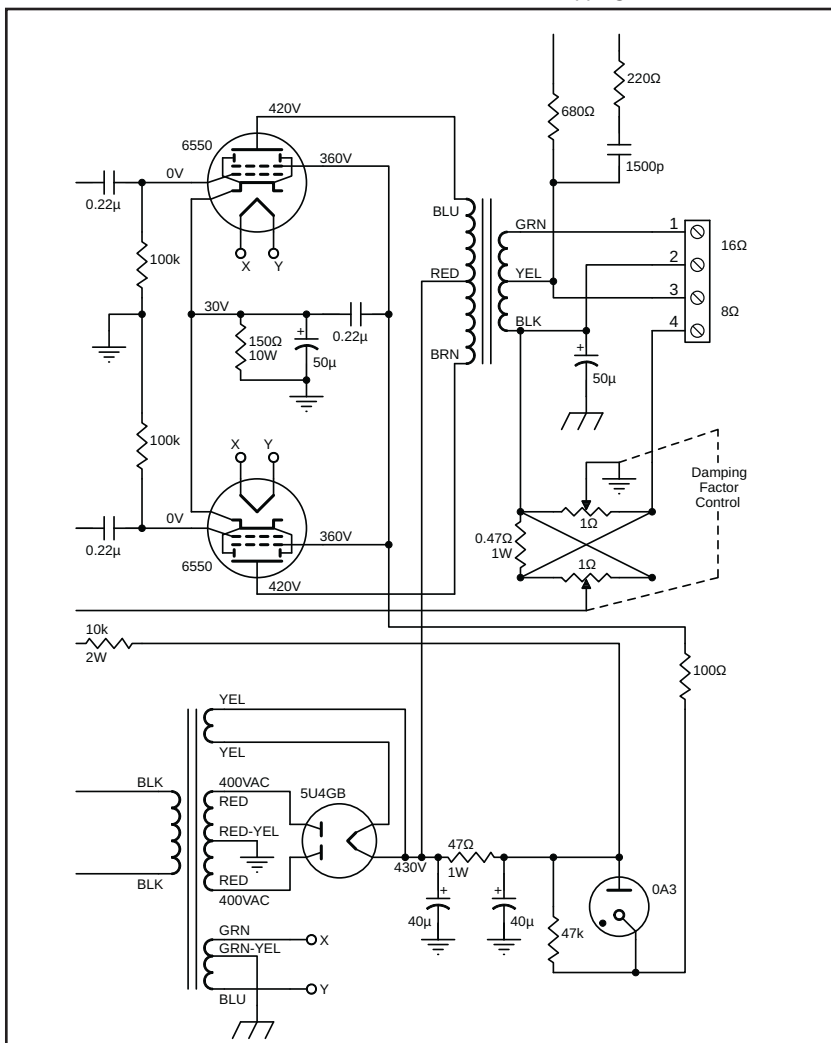


Figure 7: The screen grids are biased via a VR tube dropping from high B+. (The tube at lower right is an OA3.)

Figure 6 shows the screen grids biased by a separate section of the power supply. Although the screen grid current will increase at high output powers, the resulting droop in screen-grid voltage is not likely to be as great as in the previous circuit, so theoretically, any signal compression or added distortion would be less. A variant of this screen-grid biasing method uses a regulated screen-grid supply.

Figure 7 shows a somewhat unusual method of screen-grid biasing in which a voltage-regulator tube is used to bias the screen grids at a fixed voltage (in this case, 70 V) below the high B+. This approach was used in the Altec A340A power amplifier.

Figure 8 shows the method of screen-grid biasing used in Williamson and ultra-linear amplifiers: the screen grids are fed from a tap on the output transformer primary. This drives the screen grids with a signal that is out of phase with the control-grid signal. In other words, it provides negative feedback. Any droop in high B+ due to increased plate current at high output powers will also lower screen-grid bias, but the negative feedback helps to eliminate any effects on gain or distortion.

Figures 9–11 show the output waves for LTSpice simulations of the circuits shown in Figures 5–7, respectively, with the levels set to barely produce clipping. As you can see, there is little difference in the results based on the type of screen-grid biasing circuit used. The circuit of Figure 5 (biasing via a dropping resistor from high B+) is a bit more asymmetrical than either of the others, indicating a higher proportion of even-order harmonics compared to odd-order harmonics. In amplifiers where the output stage is driven into clipping before other

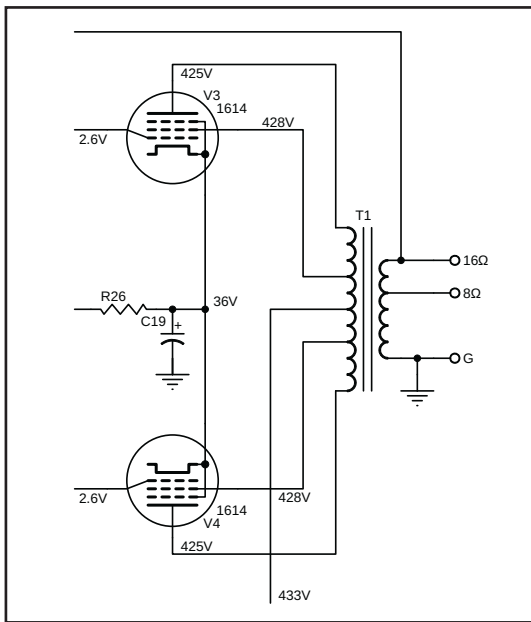


Figure 8: The screen grids are biased from output transformer taps.

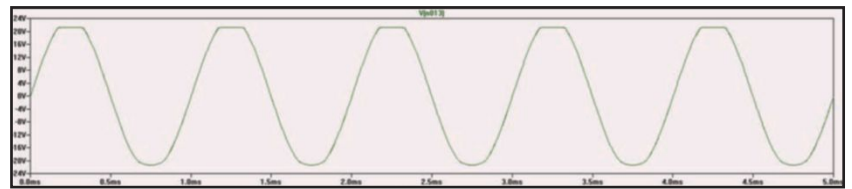


Figure 9: This is the output wave for the circuit shown in Figure 5.

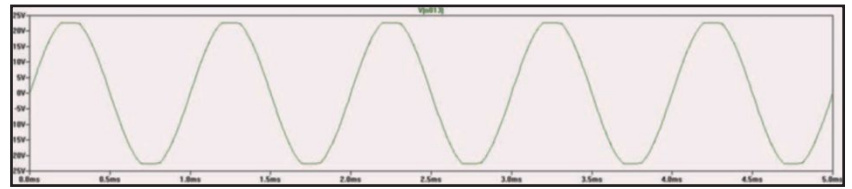


Figure 10: This is the output wave for the circuit shown in Figure 6.

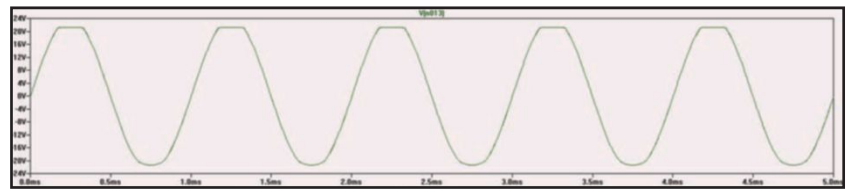


Figure 11: This is the output wave for the circuit shown in Figure 7.

stages (such as input or phase-splitter stages), this could make a slight difference in the amplifier sound when the amp is clipping.

The circuit shown in Figure 7 has a slightly higher gain than either of the other two, as you can see from the fact that both the positive and the negative peaks are a bit more noticeably clipped than those of the other circuits. Since the maximum output amplitude is set by the B+ voltage, there is no significant difference in the output power among the three circuits.

The negative feedback in the circuit shown in Figure 8 decreases the likelihood of screen-grid bias variations affecting the gain or distortion, so I did not look at the output wave of this circuit.

Comparing the Methods of Screen-Grid Biasing

Another way of comparing the effects of the different methods of deriving the screen-grid bias is to use Fast Fourier Transform (FFT) analysis to look at the frequency components of the output waves. **Figure 12** compares the FFT plots for the circuits shown in Figure 5 (blue plot), Figure 6 (green plot), and Figure 7 (red plot). The FFTs reveal one significant difference and one minor difference. The amplitudes of the even-order harmonics are much lower in the circuit of Figure 6 (the one using a separate power-supply section for screen-grid bias) than in either of the other circuits. The minor difference is the higher level of non-harmonically-related signal in the blue and red plots. These signal components could include intermodulation products of any order. If the FFTs had been made from actual outputs from real amplifiers, the non-harmonically-related signals could

also have included power-supply hum and various kinds of noise. However, such extraneous signals do not show up in circuits simulated by LTSpice or other circuit simulators.

In multi-tone testing, such signal components have been called “spectral contamination,” and their appearance in relation to the harmonic peaks is sometimes called “the dirt between the toes.” Spectral contamination is associated with lower definition in an audio signal.

It would be interesting to physically test a hollow-state power amplifier using each of the three kinds of screen-grid bias circuits in turn, to see whether the differences shown in Figure 12 really show up. If they do, then the circuit shown in Figure 6 may be a viable alternative to the Williamson or ultra-linear circuit for really clean amplification. 

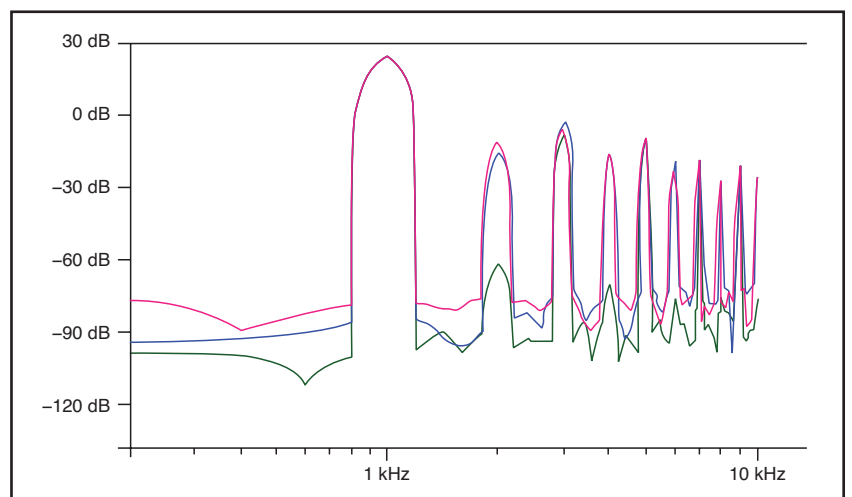


Figure 12: Here is the FFT analysis of the output waves.

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

November 2–4, 2018

Capital Audiofest

**Hilton Hotel at Twinbrook Metro,
1750 Rockville Pike, Rockville, MD
www.capitalaudiofest.com**



Following the 2016 edition with the largest visitor turnout ever during three days, the promoters decided to keep the show dates in November at the same venue, the Rockville Hilton Hotel at Twinbrook. The show benefited from the hotel's renovations, allowing for larger listening rooms and an expanded Marketplace and CanMania dedicated areas with 50% more space. The Rockville hotel has a major Metro line right out the back door, connecting from/to the center of Washington DC with no transfers.

November 3–4, 2018

CanJam Shanghai Headphone Festival

**Shanghai Marriott Hotel City Center,
Shanghai, China
www.canjamglobal.com/shanghai2018**



CanJam is the world's premier headphone audio show with annual events in New York City, Singapore, Los Angeles, London, Denver, and Shanghai and is produced by Head-Fi.org, the world's largest online audio community. CanJam is the headphone audio industry's leading platform for exhibiting brands to expand their product distribution channels, interact with their customers and industry peers, and engage with the international audio, lifestyle, and technology press.

The inaugural CanJam Shanghai features more than 100 leading brands in more than 20,000 sq ft. of exhibition space, the Shanghai Headphone Festival will showcase the latest products and innovations in headphones and personal audio electronics in all price ranges — from \$50 earbuds to the world's finest headphone audio systems at \$50,000.

November 9–11, 2018

New York Audio Show

**Park Lane Hotel, Central Park South,
New York**

www.chestergroup.org/newyorkaudioshow/2018



The New York Audio Show will return for its third year to Manhattan on November 9–11 2018! The New York Audio Show will once again take place at the luxurious Park Lane Hotel on Central Park South, one of New York's most prestigious locations.

Check out the latest in high end audio technology from loudspeakers created to perfection, headphones with the perfect fit and sound to high-resolution digital audio. This show enables visitors to compare hundreds of top products in one prime location and engage with the available engineers, designers, and product specialists.

November 13–16

electronica

**Messe München, Munich, Germany
www.electronica.de**



There's one word to describe electronica: massive! That's the main reason why it only happens every two years. For everyone involved in electronics, from R&D to production, this is it. There are Asian shows that are probably valuable for sourcing components and manufacturing, but the Munich show is where visitors stand at the cutting-edge of technology, and are able to find "the entire world of electronics in one place," as the show's slogan states. electronica showcases the complete range of technologies, broken down into sectors, from Automotive and Displays to Electromechanics/System periphery, Electronic design (ED/EDA), Embedded systems, manufacturing services,

semiconductors, PCBs and other circuit carriers, Test and measurement, Micro- and nanosystems, Passive components, Sensors, Services, Power supplies, System components/Assemblies and subsystems to Wireless.

A show within a show, with more than 450 exhibitors in the semiconductor manufacturing, printed and organic electronics sectors SEMICON Europa is the leading European microelectronics exhibition and conference for industry and research in the segments of flexible hybrid electronics, smart manufacturing, medical technology and power electronics.

More than 3,000 companies from more than 50 countries in more than 180,000 square meters of space will be at electronica 2018 to display the future with their products and solutions. Along with the increase in show size, electronica 2018 will also hold four conferences, 16 forums, and the new hall C6 will focus on Live Demonstrations showing the industry's products in their respective application.

November 12–18

Live Design International (LDI) Show

**Las Vegas Convention Center,
Las Vegas, NV**

Exhibition: November 16–18

www.ldishow.com



Since 1988, Live Design International (LDI) has been a leading trade show and conference for live design professionals from all around the globe. LDI hosts over 13,000 attendees from more than 80 countries, working in theaters, concerts, clubs, theme parks, and houses of worship, as well as a wide range of international live and broadcast venues. More than 350 companies exhibit, mostly stage and lightning vendors, but also many professional audio companies, providing live demos of cutting-edge gear.

November 14–16, 2018

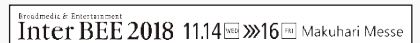
Inter BEE

Inter BEE 2018 |

54th International Broadcast Equipment Exhibition

**Makuhari Messe, 2-1, Nakase, Mihama-ku,
Chiba City, Chiba Prefecture Japan**

www.inter-bee.com



Promoted by the Japan Electronics and Information Technology Industries Association (JEITA), and run by the Japan Electronics Show Association (JESA), with the support of the Japanese government, all major broadcasting organizations and associations in Japan, and certified by the United States Department of Commerce, the International Broadcast Equipment Exhibition (Inter BEE) is always an important event for the professional audio and video industries. The event traditionally focuses on television technologies and is chosen by all major Japanese companies for key technology and product presentations—sometimes well ahead of any other show. Inter BEE is an important opportunity to promote solutions in the Japanese market with a large impact across Asia. The location, at the Makuhari Messe, is highly convenient and exhibitors recognize a very high level of visitors, including from remote areas of the world. Professional audio equipment companies represent close to 100 exhibitors, including all major microphone manufacturers.

For the first time, Inter BEE 2018 will be using Halls 1-8 at Makuhari Messe and attendance is set to go even higher this year, with more than 1,000 exhibitors. In anticipation of the 2020 Tokyo Olympics, the show will reflect important industry's efforts to enhance broadcasting services with 4K and 8K production and transmission. This year's exhibition will also feature demos of large-format speakers (X-Speaker), headphones (X-Headphone), and microphones (X-Microphones) at the Inter BEE Experience.



ELECTRONIC TRANSFORMERS

- Original Equipment
- Replacement
- Amateur

TRIAD MAGNETICS, INCORPORATED

460 HARLEY KNOX BLVD., PERRIS, CA 92571

www.TriadMagnetics.com

THE ULTIMATE^{IN} AFFORDABLE SIGNAL PROCESSING



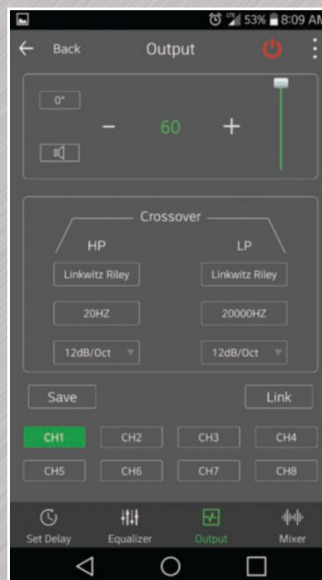
The Dayton Audio DSP-408 is a feature-rich 4 input, 8 output digital signal processor that was designed to give you the tools to contour your audio system to fit your tastes. The DSP-408 allows the user to choose the application of their device. With two methods of powering the DSP, this processor is the perfect addition to your home or car audio system.

#230-500

Windows Program



Mobile App



The most unique feature of the DSP-408 is the ability to operate in a vehicle or home audio system right out of the box. This adaptation allows users to utilize a single model of DSP for all their audio processing needs.

The ability to control this DSP is provided via a windows application or via Bluetooth connection to an Android or iOS device when paired with the Dayton Audio DSP-BT4.0 (#230-502). The volume and preset selection for the DSP can also be controlled by the optional wired remote (#230-504).

For many individuals, the pursuit of perfect sound quality is a never-ending quest. When utilizing all the features the DSP-408 offers, the ability to manipulate and improve the sound quality of your audio system is simply astonishing. Connect to your DSP-408 and get to work!

Visit parts-express.com.
Custom OEM inquiries welcome.

For optimal performance, check out these accessories (sold separately).

**Dayton Audio
DSP-BT4.0 Bluetooth
Data and Streaming
USB Interface for DSP-408**



#230-502

**Dayton Audio
DSP-RC Wired
Remote Control
for DSP-408**



#230-504

**SAVE \$10
off \$100 order!**

Code: AXP4Q8
Expires 11/30/2018

DAYTONAUDIO
Your path to great sound.™

Distributed By:
**PARTS
EXPRESS**
AUDIO COMPONENTS & SOLUTIONS

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