INNOVATIONS IN AUDIO • AUDIO ELECTRONICS • THE BEST IN DIY AUDIO

www.audioxpress.com

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Focus on Acoustics

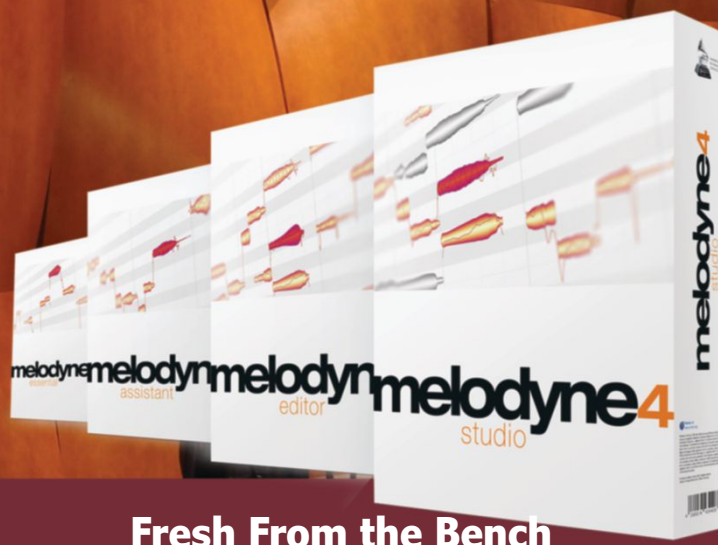
Acoustic Solutions Round-Up 2017
First-Approach Solutions

Audio Praxis
Sound Diffusion: The Movement Toward Omnipresence
By Jim DeGrandis

Audio Praxis
Early Reflections
By Ethan Winer

Sound Engineering
JOCAVI Acoustic Shells Enhancing the Stage
By João Martins

R&D Stories
Low-Frequency Absorption Using Membranes
By John Calder



Show Report
Munich High End 2017
By Ward Maas

Sound Control
Acoustical Principles for Microphone Placement
By Richard Honeycutt

Fresh From the Bench
Celemony Melodyne 4
By Fernando Rodrigues

Fresh From the Bench
Primacoustic Recoil Stabilizers
By Gary Galo

$9.00US $10.00CAN
08
7 25274 84697 9

AUGUST 2017

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

www.partsconnexion.com

Sales: order@partsconnexion.com

Inquires: info@partsconnexion.com

Toll-Free - US/Canada

905-631-5777 **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

August 2017 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., at 111 Founders Plaza, Suite 904, East Hartford, CT 06108, US Periodical Postage paid at East Hartford, CT, and additional offices.

Head Office:
KCK Media Corp.
111 Founders Plaza, Suite 904
East Hartford, CT 06108, US
Phone: 860-289-0800
Fax: 888-980-1303

Subscription Management:
audioXpress
P.O. Box 462256
Escondido, CA 92046, US
Phone: 800-269-6301
E-mail: audioxpress@pcspublink.com
Internet: www.audioxpress.com

Postmaster: send address changes to:
audioXpress
P.O. Box 462256
Escondido, CA 92046, 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
111 Founders Plaza, Suite 904
East Hartford, CT 06108, 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. 2017
Printed in the US



Transcendental Acoustics?

Acoustics are part of this magazine's mission. Acoustic science is where everything starts, determining how sound and audio systems behave in practice.

As Richard Honeycutt wrote in an article for our monthly column Sound Control, dedicated to acoustics, "One of the most convenient properties of physics is also sometimes one of the most confusing. This property is often called "complementarity," and it is the property according to which a complete understanding of certain phenomena can only be achieved by using very different models, with very different mathematical formalisms." In that article, "A Complementarity Principle in Acoustics," published in our December 2016 edition, Honeycutt reminds us that the study of acoustical phenomena can sometimes be measured as waves and rays... or particles, since a popular electro-acoustical software such as EASE (Enhanced Acoustic Simulator for Engineers—www.afmg.eu) refers to the elements used for "ray-tracing" as "particles." And as stated in Physics manuals, "Sound is a mechanical wave that results from the back and forth vibration of the particles of the medium through which the sound wave is moving."

So, how exactly should we consider sound behavior?

This is only one of the fascinating aspects of acoustics science and the reason why I always immensely enjoy meeting with the minds behind some of the most brilliant acoustic solutions and installations, and finding out how they study, design, simulate, and measure acoustic solutions. As it becomes obvious, every acoustician follows his or her own approach and applies established methods to deal with phenomena, described as echoes, reflections, refraction, diffraction, interferences, standing waves, pressure waves, response peaks, nulls, reverberation, modal ringing modes, vibrations, compressions, rarefactions, and so forth. It all depends on what the perspective toward controlling the propagation of sound might be. And to deal with those phenomena, all acousticians seem to depend upon the same basic set of tools, applying absorption and diffusion.

Significantly, there is a fascinating amount of research happening worldwide in this area, and great progress is to be expected when those projects combine the understanding of acoustics with new approaches in electro-acoustics—which isn't only used to generate and amplify sound, but can also cancel it—combined with the exponentially better digital signal processing resources that we have available.

While I was working on this edition and reading about ongoing research, I learned about an announcement from the Department of Energy's Lawrence Berkeley National Laboratory (Berkeley Lab, University of California—www.lbl.gov), where an "elegantly simple experiment with floating particles self-assembling in response to sound waves has provided a new framework for studying how seemingly lifelike behaviors emerge in response to external forces." Basically, scientists describe the findings of their research as "the essence of life." Apparently, scattering particles excited by sound waves create "collective patterns" that emulate life itself, and "could form the basis for developing intelligent networks that perform simple non-algorithmic computation," the researchers say.

Could acoustics, therefore, also be an expression of a new dogma? Could it be that acousticians are dealing with something that transcends them? As large as life?

João 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 | circuit cellar | LIS Loudspeaker Industry Sourcebook | Voice Coil

SUPPORTING COMPANIES

ACO Pacific, Inc.	49	IsoAcoustics, Inc.	47
Acoustic Geometry	19	Jensen Transformers, Inc.	27
Acoustics First Corp.	40	JOCAVI, Lda.	43
All Electronics Corp.	49	KAB Electro-Acoustics	43
Avel Lindberg, Inc.	55	Menlo Scientific, Ltd.	67
COMSOL, Inc.	9	OPPO Digital, Inc.	13
Front Panel Express, LLC	34	Parts ConneXion	2
G.R.A.S. Sound & Vibration	33	Parts Express International, Inc.	68
Hammond Manufacturing, Ltd.	3	Primacoustic	27
Hypex Electronics B.V.	35	The Loudspeaker Sourcing Show	21

NOT A SUPPORTING COMPANY YET?

Contact Peter Wostrel (audioxpress@smmarketing.us, Phone 978-281-7708, Fax 978-281-7706) to reserve your own space for the next edition of our magazine.

Highlight your company by sponsoring an article or column.

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.



Contents

audioxpress







Acoustics Special Features

36 Acoustic Solutions Round-Up 2017
By João Martins
Read about innovative "first-approach" acoustic treatment solutions.

41 Sound Diffusion: The Movement Toward Omnipresence
By Jim DeGrandis
This article discusses recent developments in diffusion, diffusor design and use, the science around diffusion simulation, and contributions for an upcoming standard.

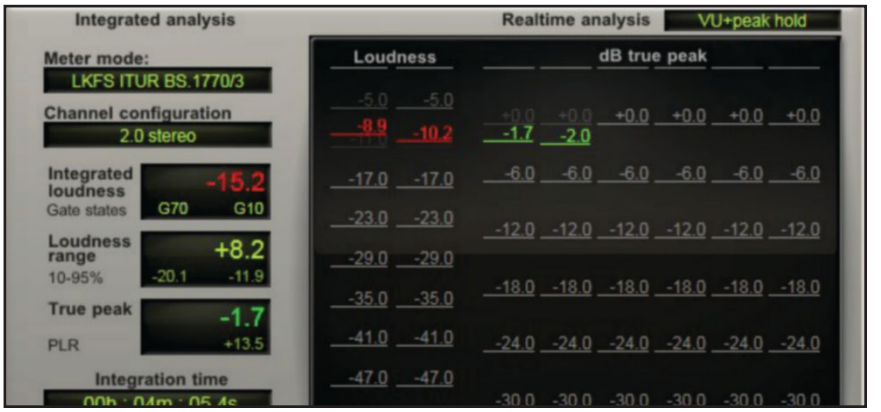
44 Early Reflections
By Ethan Winer
This article explains what "early reflections" are and why they must be avoided if your goal is accurate sound reproduction.

50 New Research on Low-Frequency Absorption Using Membranes
By John Calder
The article describes the problem of destructive room modes in critical-listening and sound recording environments, and the difficulties associated with testing low-frequency absorbers, which are most often used to mitigate room modes.

56 JOCAVI AS Acoustic Shells
Enhancing the Stage
By João Martins
Using different materials and unique geometries, JOCAVI's acoustic shells are meant to project the non-amplified original sound from the stage toward the audience, while also enhancing the performers' awareness of stage sound.







Features

14 Munich High End 2017
A Vital Showcase for Trends and Businesses
By Ward Maas
The 36th edition of the Munich High End show, held May 18-21, 2017 at MOC, drew people from all over the world to explore products and trends in the audio industry.

22 Celemony Melodyne 4
A Different Way to Look at Audio
By Fernando Rodrigues
Celemony Melodyne 4 software uses its Direct Note Access (DNA) technology to analyze audio to determine the main frequency content and display those notes so users can manipulate them individually, changing pitches and durations at will.

28 Primacoustic Recoil Stabilizers
A Solid Speaker Base
By Gary Galo
Primacoustic has attempted to solve the problem of loudspeaker stability and isolation with its Recoil Stabilizers range.

Columns

IT'S ABOUT THE SOUND
8 Loudness Revisited (Part 3)
Software Tools
By Ron Tipton

SOUND CONTROL
30 Acoustics and Microphone Placement
By Richard Honeycutt

HOLLOW-STATE ELECTRONICS
62 Williamson's Amplifier Games
By Richard Honeycutt

Departments

4 From the Editor's Desk
5 Client Index
66 Article Updates
66 Industry Calendar

Websites

audioxpress.com
voicecoilmagazine.com
cc-webshop.com
loudspeakerindustrysourcebook.com

@audioxp_editor audioxpresscommunity
linkedin.com/company/audioxpress

Loudness Revisited (Part 3)

Software Tools

This article series continues to look at ways to understand, measure, and use loudness control. We all want to avoid the need to constantly adjust the volume control and loudness normalization is the answer. In this article, we look at the practical experience with some available software tools.

By
Ron Tipton
(United States)

When it comes to loudness control, free loudness meters are useful but be aware of the old saying that you usually get what you pay for. I like the TB_EBULoudness_v3(2014) meter from Toneboosters because of its nice screen layout and because it holds all the peak values after the music stops playing. Available for both Windows and Mac, it's not free but it is reasonably priced at €19.95 (about $22). For Windows, it's a VST plug-in that I use in my Reaper DAW. The screenshot in **Photo 1** shows all the peak values neatly held at the end of the measurement.

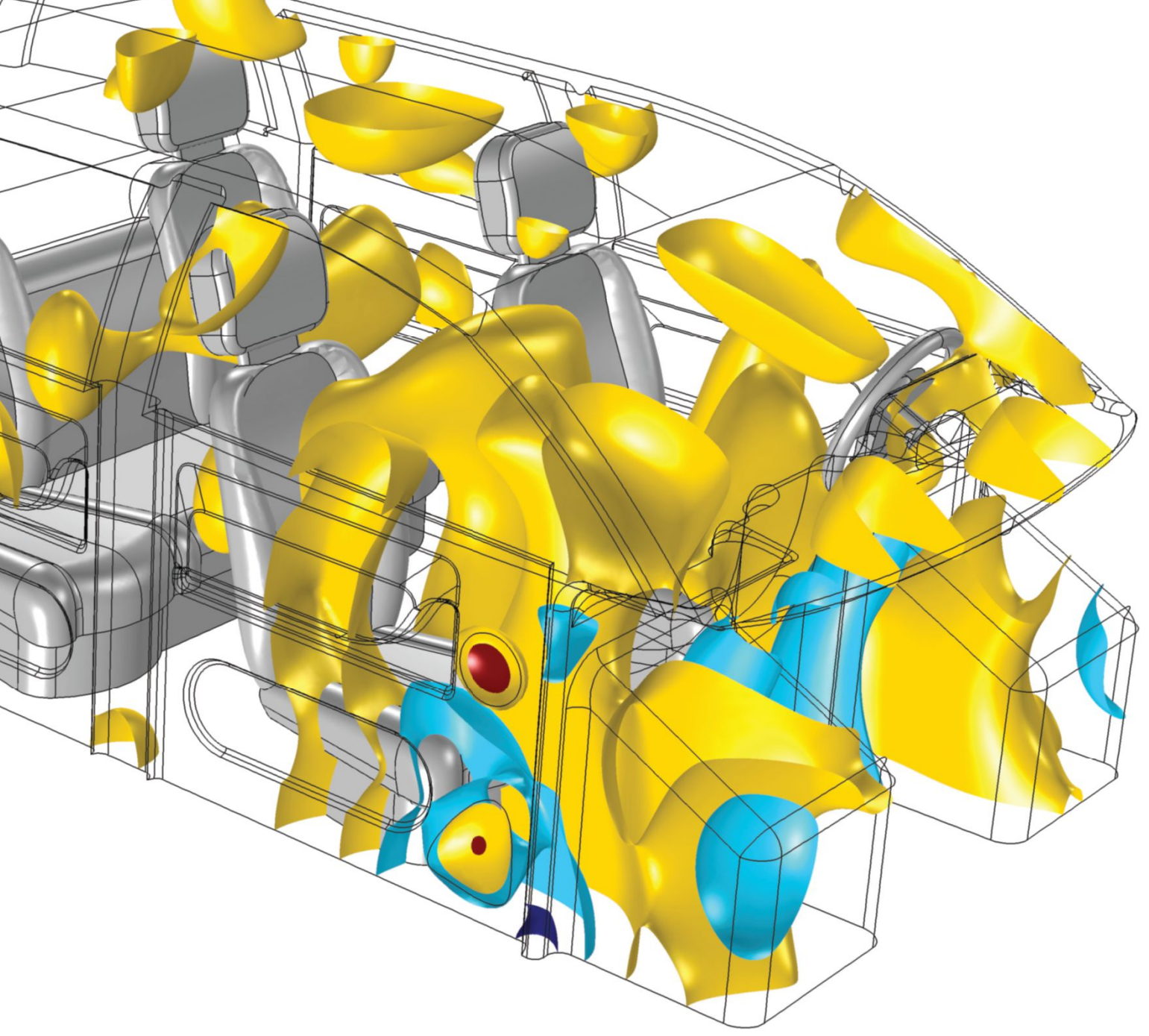


Photo 1: The Toneboosters loudness meter captures and holds the peak values, making it easy to write them down. The numbers to save are the Integrated loudness, True peak and SLk, the maximum Short-term, "S." The Meter mode is selected by clicking in the mode window and selecting from the drop-down menu.

Photo 2 shows the drop-down Meter Mode menu. There are several choices, but LKFS ITUR BS.1770/3 is rapidly becoming a standard and its readings are the same as LUFS EBU R128 (2014). The added gating in versions 3 and 4, which exclude the quietest 10% and loudest 5%, improve the measurement agreement with subjective listening tests.

Loudness of Different Sources

As I mentioned in "Loudness Revisited: Part 1" (*audioXpress*, June 2017), my goal in this article series is to look at normalizing music tracks taken from a variety of sources. This is exactly the situation you would have if you created a playlist without regard to the sources. I picked tracks at random from my music folder making sure I selected one track from each source. I used the



VERIFY AND OPTIMIZE YOUR DESIGNS

with COMSOL Multiphysics®

The evolution of computational tools for numerical simulation of physics-based systems has reached a major milestone.

Surpass design challenges with ease using COMSOL Multiphysics®. Work with its powerful mathematical modeling tools and solver technology to deliver accurate and comprehensive simulation results.

Develop custom applications using the Application Builder and deploy them within your organization and to customers worldwide with a local installation of COMSOL Server™.

Benefit from the power of multiphysics today, request a live demo at **comsol.com**

© Copyright 2017 COMSOL. COMSOL, the COMSOL logo, COMSOL Multiphysics, Capture the Concept, COMSOL Desktop, COMSOL Server, and LiveLink are either registered trademarks or trademarks of COMSOL AB. All other trademarks are the property of their respective owners, and COMSOL AB and its subsidiaries and products are not affiliated with, endorsed by, sponsored by, or supported by those trademark owners. For a list of such trademark owners, see www.comsol.com/trademarks.

COMSOL

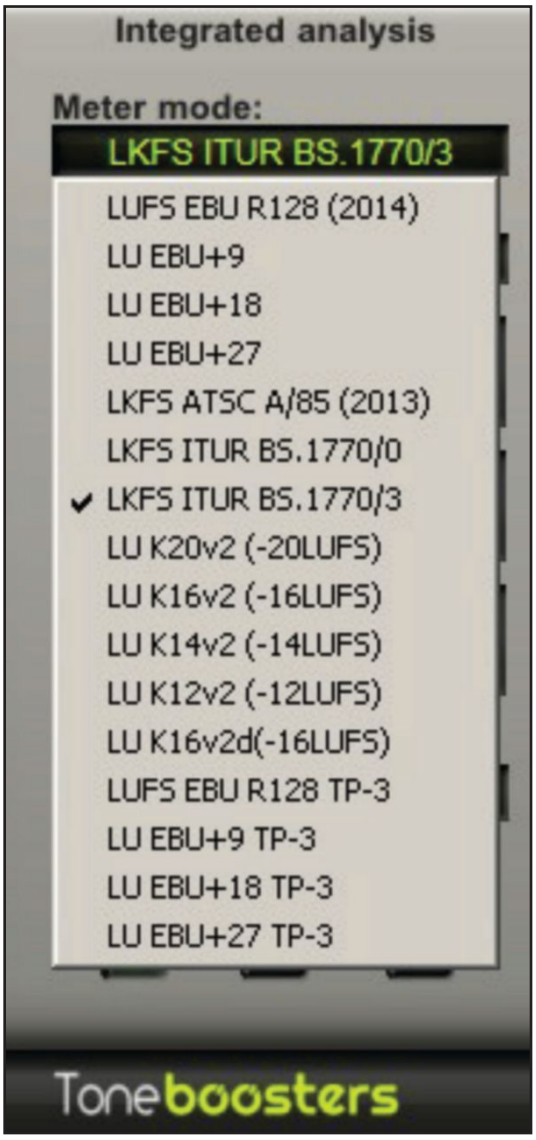


Photo 2: This screenshot shows the Meter mode drop-down menu. I used LKFS ITUR BS.1770/S because of its gating to exclude the quietest 10% and loudest 5%. However, LKFS EBU R128 (2014) displays the same measured numbers.

Toneboosters analyzer to find the loudness and true peak numbers shown in **Table 1**. The table would be too cluttered to have all the music source information included so I put the details in the Sources section at the end of the article.

Using the equation: Normalization (decibels) = Target – Integrated ("I"), and using the data from Table 1, the Integrated values range from -14.5 to -21.1 LUFS—all larger than -23.0 LUFS. I calculated the normalizing values for the 10 tracks to a target of -23.0 LUFS, as shown in **Table 2**. Normalizing to -16 LUFS did not work without allowing compression because the DVD Sound Track gave a true peak of +1.2 dB.

Next, I used the free Audacity audio editor to normalize each track. Using Amplify in the Effects menu, I entered each normalization value (in decibels) and pressed OK. I saved the normalized files by adding a "-N" to the original file name to keep things organized. Measuring each of the normalized files with the Toneboosters meter shows them all at -23.0 LUFS, as expected.

I created a 10-track playlist in both VLC Media Player (see **Photo 3**) and Foobar 2000. Foobar seems more straightforward to me, but both performed well. A listening test was pleasing, all tracks seemed equally loud so volume control adjustment was not needed. But, I wondered how normalization using the maximum short-term numbers would sound.

Normalization Using Max "S"

Looking again at Table 1, we see all the numbers in the "S" column are larger than -16 LUFS. Therefore, normalization to a -16 LUFS target will result in attenuation in all cases which will not increase any of the true peaks, see **Table 3**.

I created another 10-track playlist and also one with 20 tracks using the arrangement: Track-1-I, Track-1-S, Track-2-I, Track-2-S, and so forth. It was difficult to hear any difference but I think I had a slight preference for the Max S normalization. This

Source	Integrated "I"	Max Short-term "S"	True Peak
78 RPM record	-15.8 LUFS	-11.9 LUFS	-1.7 dB
Reel-to-reel tape	-14.7 LUFS	-10.4 LUFS	-2.5 dB
Audio cassette	-15.4 LUFS	-11 LUFS	-1.4 dB
33-1/3 RPM record	-15.2 LUFS	-10.5 LUFS	-1.7 dB
Purchased CD	-14.8 LUFS	-10.6 LUFS	-0.3 dB
VHS movie sound track	-14.5 LUFS	-12.4 LUFS	-3.9 dB
DVD sound track	-18.7 LUFS	-11.9 LUFS	-1.5 dB
iTunes movie sound track	-17.1 LUFS	-10.7 LUFS	-3.5 dB
Deezer capture	-16.1 LUFS	-9.8 LUFS	-1.5 dB
Tidal capture	-21.1 LUFS	-14.4 LUFS	-6.8 dB

Table 1: These are my 10 source tracks with their "I," "S," and True peak numbers as measured by the Tonebusters loudness analyzer. Note that all "I" values are greater than -23 LUFS. This means all tracks will be attenuated (negative amplification) when normalized to -23 LUFS and the true peaks will also be attenuated.

Source	Normalization to -23 LUFS using "I"	Calculated New True Peak	Measured "I"	Measured True Peak
78 RPM record	-7.2 dB	-8.2 dB	-23.0 LUFS	-8.2 dB
Reel-to-reel tape	-8.3 dB	-10.8 dB	-23 LUFS	-10.8 dB
Audio cassette	-7.6 dB	-9 dB	-23 LUFS	-9 dB
33-1/3 RPM record	-7.8 dB	-9.5 dB	-23 LUFS	-9.5 dB
Purchased CD	-8.2 dB	-8.5 dB	-23 LUFS	-8.5 dB
VHS movie sound track	-8.5 dB	-12.4 dB	-23 LUFS	-12.4 dB
DVD sound track	-4.3 dB	-5.8 dB	-23 LUFS	-5.8 dB
iTunes movie sound track	-5.9 dB	-9.4 dB	-23 LUFS	-9.4 dB
Deezer capture	-6.9 dB	-8.4 dB	-23 LUFS	-8.4 dB
Tidal capture	-1.9 dB	-8.7 dB	-23 LUFS	-8.7 dB

Table 2: Column two lists the Effects > Amplify numbers I used in the Audacity audio editor to normalize the tracks to -23 LUFS. After normalization, I re-measured with the expected results in Column four and Column five.

may just be my bias because Max S seems more logical to me.

Free, or nearly free, software works well for normalization but the measurements are done in real-time. That is, each track is played so it takes generally three to five minutes, plus some time to write the numbers down, close the file, and load the next one. This is fine for a short playlist but it would be very time consuming for perhaps 100 or more tracks. Fortunately, it can be done faster, but it's not free.

L-Normalizer

L-Normalizer, software for Windows and Mac, performs the track measurement and displays the results in 15 to 25 seconds depending on the track length. I licensed the Windows version. The cost is $219, which is worth it if you have a lot of tracks to process. The Windows software "bundle" contains a VST plug-in and a stand-alone version. I've been using the stand-alone, although it is a bit "odd."

When the installation program is run, the VST plug-in is put where your DAW can find it, but the stand-alone is left in the C:\program files\beDSP folder as L-Normalizer.exe. It's convenient to right-click it and put a short-cut icon on your desktop.

The initial screen has the usual File selection in the Menu bar, but it's for saving the rendered (normalized) file. To load the file to be processed, drag and drop it to the initial screen. A progress bar appears and the measurement will start. The normalized results will be added to the screen in the "Source" column (see **Photo 4**).

As we know from the previous discussion of normalization, you can't arbitrarily set the target and true peak without allowing compression—L-Normalizer will do the compression if it needs to, in order to meet your target values.

Source	Normalization to -16 LUFS using Max "S"	Calculated New True Peak
78 RPM record	-4.1 dB	-5.1 dB
Reel-to-reel tape	-5.6 dB	-8.1 dB
Audio cassette	-5 dB	-6.4 dB
33-1/3 RPM record	-5.8 dB	-7.5 dB
Purchased CD	-5.4 dB	-5.7 dB
VHS movie sound track	-3.6 dB	-7.5 dB
DVD sound track	-4.1 dB	-5.6 dB
iTunes movie sound track	-5.3 dB	-8.8 dB
Deezer capture	-6.2 dB	-7.7 dB
Tidal capture	-1.6 dB	-8.4 dB

Table 3: I used the Column two numbers to normalize the source tracks to -16 LUFS using the Maximum Short-term, "S," measurements. Again, all tracks were attenuated with true peaks well below 0 dB full scale. My listening tests revealed a slight preference for this normalization as compared to using the Integrated loudness.

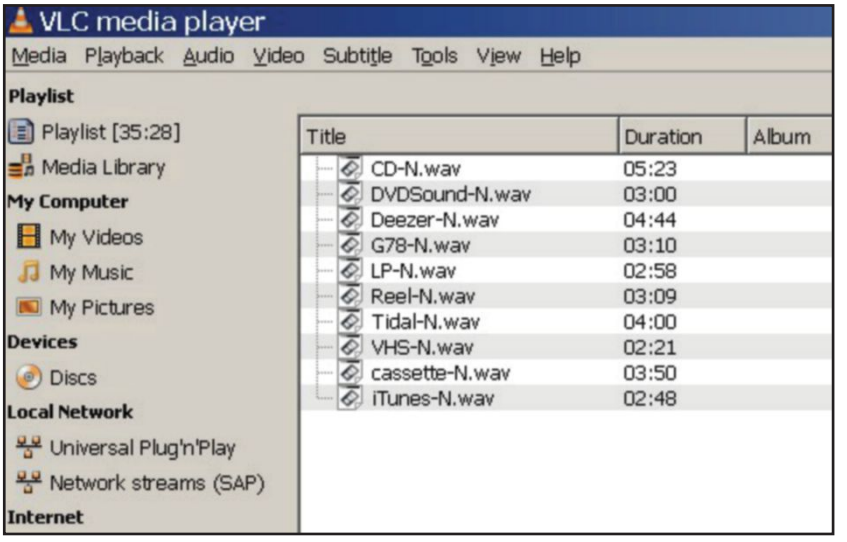


Photo 3: My VLC media player playlist for the tracks normalized to -23 LUFS lists my 10 normalized tracks. Listening was very pleasant without having to make any volume control adjustments.

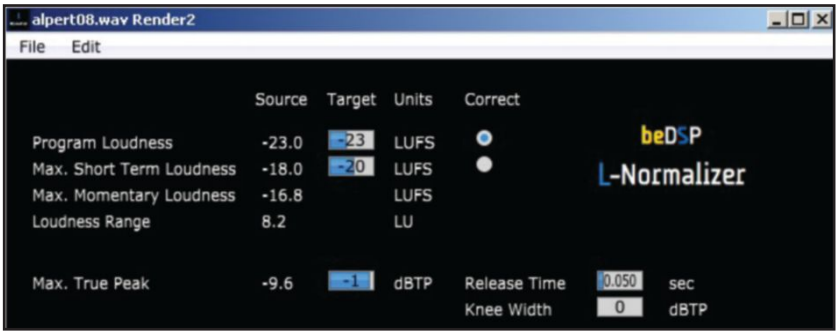


Photo 4: The L-Normalizer screen after analysis shows the measurements in the Source column. To save the normalized track, click File in the Main menu bar, then Render and then Save. The next screen asks for the saving location folder and file name.

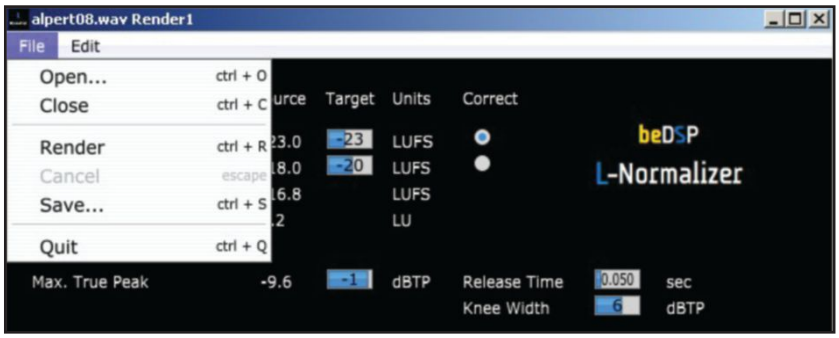


Photo 5: This screenshot displays the File drop-down menu. In addition to Render and Save, you can Open a saved file for another measurement, Close an opened file or Quit the program.

The "Correct" radio button was set to Program Loudness ("I"), but Maximum Short Term could have been selected instead. A True Peak of -9.6 dB means no compression would be used and these numbers agree with my previous results for the LP track (Herb Alpert). L-Normalizer is compliant with international loudness standards including ITU BS1770/3.

To save the normalized track, click File > Render and then Save (see **Photo 5**). Choose a file name and folder and you are done. Other normalizations can be done by changing the targets, rendering and saving.

LCn Loudness Correct

This is another loudness measurement and normalization program for Windows and Mac. It includes a native plug-in for the Pro Tools DAW and a stand-alone. The license fee is $299 for the stereo version (LC2n) and $699 for the 5.1 (LC6n). A 14-day free trail is available and an iLok account is needed for the license key. It does have batch processing which may be useful for multiple files so you may want to check it out.

User Manuals for all these programs are included in the Supplementary Material, located on the audioXpress website (www.audioxpress.com).

Music Sources

78 RPM Record
Movin' Around, Erroll Garner Trio, Black & White Record Co No.16A, released 1945. Digitized to 44.1 ksps, 16-bits in 2015.

0.25" Reel-to-Real Magnetic Tape, 7.5 ips
You'll Never Walk Alone, sung by Roy Hamilton. From the musical Carousel by Oacar Hammerstein II. Recorded from a radio broadcast c.1952. Digitized to 44.1 ksps, 16-bits in 2005.

Audio Cassette
"Stardust," sung by Willie Nelson, *Willie Nelson: His Greatest Hits*, CBS Special Product for the Reader's Digest, KRB733/A1. Digitized to 44.1 ksps, 16 bits, 2001.

33-1/3 RPM (LP) Record
"Let It Be Me" (Track 8), Herb Alpert and the Tijuana Brass, *The Lonely Bull*, A&M Records, 101. Digitized to 44.1 ksps, 16 bits, 2012.

Purchased CD
"Night Train," composed and played by Ray Montford, *Many Roads*, Softail Records, Canada, 2003. Digitized to 44.1 ksps, 16 bits, 2005.

Video cassette (VHS) Sound track
Opening song of the Third Act, sung by Nelson Eddy (in French). From the opera *Marta* by Friedrich von Flowtow. This is an opera segment from the 1943 musical version of *Phantom of the Opera* by Gaston Laroux. Digitized to 44.1 ksps, 16 bits, 2011.

DVD Sound track
Memory from the musical *CATS* by Andrew Lloyd Weber. "Memory" is sung by the cat named Grizabella but as to who was playing that part in the Broadway production when the DVD was captured is open to question.

iTunes Video Download (Sound track)
"One Day I'll Fly Away," duet sung by Satine (Nicole Kidman) and Christian (Ewan McGregor). From the *Moulin Rouge* movie sound track, 2001.

Deezer Streaming capture
"Stella by Starlight," Miles Davis, **1958 Miles**, released in 1974 by CBS/Sony. 44.1 ksps, 16 bits.

Tidal Streaming capture
"Old Man River,"
The Great Broadway Collection, Volume 1, London Theater Orchestra and Singers. 44.1 ksps, 16 bits.

Project Files

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

Sources

Audacity audio editor
Audacity | www.audacityteam.org

L-Normalizer
beDSP | bedsp.net/l-normalizer

Reaper DAW
Cockos, Inc. | www.reaper.fm

foobar2000
foobar2000 | www.foobar2000/org

LCn Loudness Correct
tc electronic | www.tcelectronic.com/lcn

TB_EBULoudness_v3(2014) meter
Toneboosters | www.toneboosters.com/tb-ebuloudness

VLC Media Player
VideoLAN Organization | www.videolan.org/vlc/index.html

oppo



Get More From Your Music.

Sonica DAC works on a fairly basic principle— maximize your audio from just about any source. Get wired with USB, or stream from local servers and smartphones over a slew of wireless connectivity options. Stream your favorite TIDAL or Spotify playlists, or play your downloaded tracks from USB drives. However you choose to enjoy digital music, Sonica DAC's clean power, fully balanced design, and high-performance ES9038PRO DAC makes it all sound its very best.





Sonica DAC is available for $799 on oppodigital.com or from select retailers and installers across the country.

OPPO Digital, Inc. | www.oppodigital.com | Menlo Park, CA

@oppodigital

Munich High End 2017

A Vital Showcase for Trends and Businesses

By
Ward Maas
(Netherlands)

The 36th consecutive staging of the Munich High End show was held from May 18-21, 2017, on the MOC premises. The exhibition halls and rooms encircling the atriums featured 538 exhibitors from 44 countries and the flow of visitors was constant, as people from all over the world came to explore products and trends in the audio industry.

Stefan Dreischärf, Managing Director of the High End Society, dispelled any lingering rumors by saying the show will remain in the MOC for the coming years. He added that there are even expansion plans and ongoing renovations in the complex given that the only dilemma for the promoters is where to accommodate growing requests for more exhibits.



To quickly sum up the facts of the High End 2017, which I think is the world's best-organized consumer audio show:

- The number of exhibitors increased 4%. Approximately, 63% of these were international. On the international level, the following countries were most strongly represented: United States, United Kingdom, Italy, France, Switzerland, Denmark, China, Japan, the Netherlands, Canada, and Taiwan.
- Increasing visitor numbers, totaling 21,412 visitors (+10%).
- 8,002 trade visitors from 78 countries (+13%).

So as usual, standing at the entrance, I felt overwhelmed with all the people and products competing for my attention. It took some time to step in and start the search for products that were

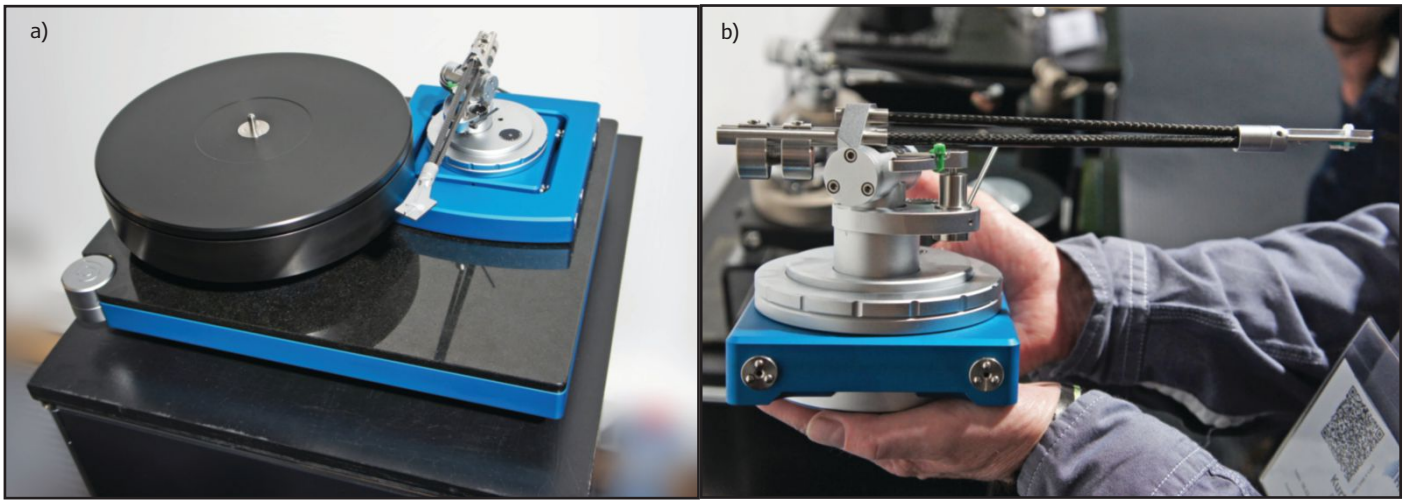


Photo 1: Swiss high precision De Baer turntable (a) and the De Baer arm construction (b)

different, special, drew attention, or simply brought a smile on my face.

Turntables

Well, the turntable is back, but this year they flooded the place, from small to extremely large, from understated finishes to loud color schemes, and from affordable to extremely expensive and sophisticated.

As an example, the De Baer Saphir turntable (www.debaer.ch) was the ultimate expression of Swiss design and craftsmanship. Even if you do not need or want a turntable, you have to be impressed by this well-thought-out piece of Swiss engineering (see **Photo 1**). The platter and the base for the tonearm are connected to the chassis through hardened steel balls. This results in maximum stiffness with minimal contact area. The tonearm is mounted



Photo 2: The Pro-Ject Essential series turntable is very affordable and contains some special features.

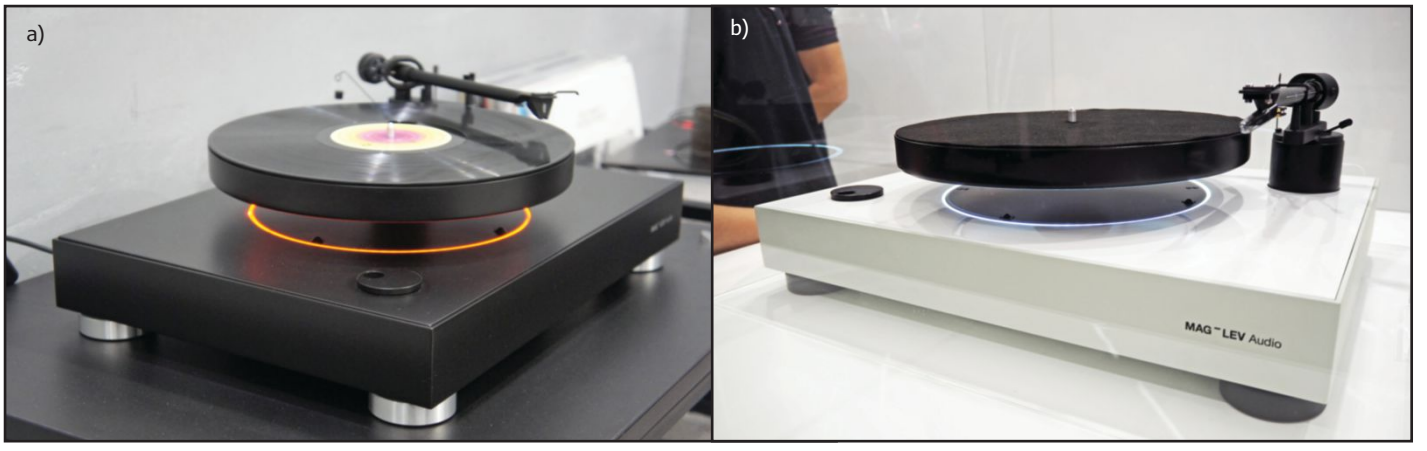


Photo 3: The Mag Lev floating platter is held in the air by magnetic levitation (a) and comes in various colors including white (b).

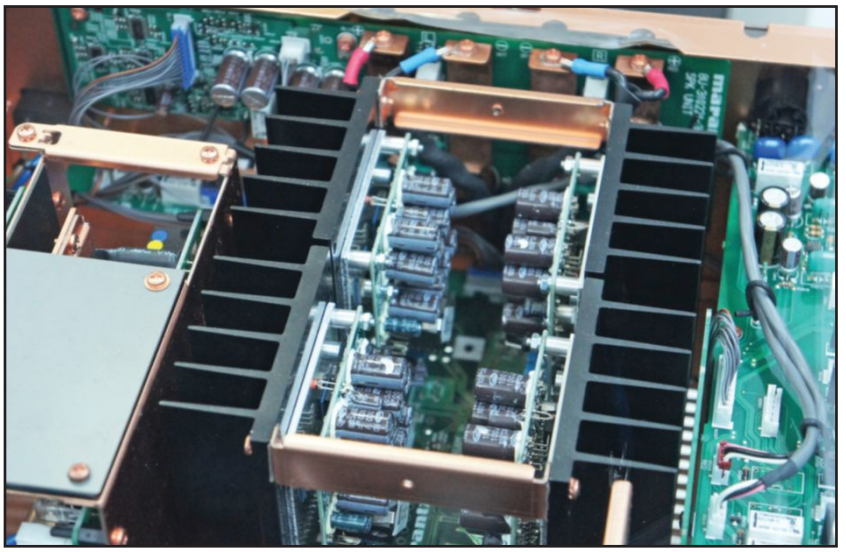


Photo 4: The Marantz PM-10 has a Class-D under the bonnet.



Photo 5: The Audio Note Kagura tube amplifier is a beauty.



Photo 6: The Hybrid Pathos InPol Heritage is great for tube enthusiasts.

with magnets and high-precision ball bearings. The preloaded carbon tubes counteract the oscillation of the pickup. Stunning... But do not ask for the price if you cannot pay it.

Austrian turntable manufacturer Pro-Ject (http://project-audio.com/) presented a turntable series that price-wise is almost the opposite of the De Baer turntable. A very affordable series with some very special features.

The Essential III series consists of a normal turntable with a built-in quality preamp, or an optical output, or with Bluetooth, or a USB output and a built-in 24/192 D/A conversion (see **Photo 2**). They come in three colors and have a price ranging from €325 to €450 (which converts to approximately $360 to $500).

I also have to mention the Mag-Lev Audio turntable (https://maglevaudio.com) with a free-floating platter held in the air by magnetic levitation. The orange light below the platter gives it something of a sci-fi look. When not in use the platter lands softly on raising feet. Impressive! I still need to find out the specs, but it looks so cool (see **Photo 3**).

Amplifiers

As for amplifiers, did someone mention that Class-D was on the rise? Check out the Marantz PM-10 with Hypex units under the bonnet (see **Photo 4**). Yes, there were also the large dinosaur amps around. Some companies clearly state that they will never would use Class-D technology. Some of them for very sound (!) reasons. But reading a review of the PM-10 in a German magazine where the Marantz (www.us.marantz.com/us/pages/home.aspx) was described as having a slightly "warm" sound, I wonder how many of the big amps we will see next year. We know what happened to the dinosaurs.

And for the tube enthusiasts, there was lots to see as well. Now, the Audio Note Kagura (www.audionote.co.jp/en) is simply a beauty (see **Photo 5**). The hybrid Pathos InPol Heritage (www.pathosacoustics.com) is special as well (see **Photo 6**).

Loudspeakers

Loudspeakers were everywhere. And, there are a lot of developments happening. The trend of active speakers with a lot of "extras" continued. The KEF LS50W wireless (http://uk.kef.com/) is a good example of this trend. For mid/high, it uses an A/B amplifier. For the bass, a Class-D. The LS50W comes with a lot of features such as dual band Wi-Fi, Bluetooth, 24/192 converters, and a subwoofer output. That makes it a very interesting alternative to a passive system with all the extra hardware needed (see **Photo 7**).



Photo 7: The KEF LS50W comes with several great features.

Fink Audio Consulting (www.finkteam.com) is one of the most successful European design companies for loudspeaker contract design. Many award-winning loudspeakers have their origins there. Under the brand name Fink-Team, they decided to make a no-compromise loudspeaker completely to their own liking.

The result is the WM-4, a 135 Kg, three-way system with in-house developed bass and midrange drivers and a Mundorf AMT as a tweeter. The in-house joke is that WM stands for "Washing Machine." That might be true as the prototype was called "Miele" after the German washing machine manufacturer. At €65.000 (which converts to approximately $70,000), this WM-4 is a loudspeaker you can both look and listen to with awe (see **Photo 8**).

Another interesting new loudspeaker series is the Wolf von Langa range of speakers (www.wolfvonlanga.com). They are field coil loudspeakers where the loudspeaker's magnetic field is generated by the current going through an extra coil on the loudspeaker (see **Photo 9**). A difficult but very rewarding approach if you get it right. The new series ranges from about €30.000 to €100.000 (which converts to approximately $34,000 to $115,000).

Färber Acoustics (www.faerber-acoustics.com) is a company that does not shy away from unusual loudspeaker designs. Several years ago, the company was at the show with the all marble Equilibrium speakers equipped with a Karlson split tweeter. Now it has a series of… well let's say… tree trunk speakers. They are called Trunkquility (see



Photo 8: Fink-Team's WM-4 is a three-way system with in-house developed bass and midrange drivers and a Mundorf AM as a tweeter.



Photo 9: Wolf von Langa introduced a new range of field coil loudspeakers.



Photo 10: The Färber Trunkquility line is made from specially hollowed out tree trunks.



Photo 11: Schanks motional feedback drive unit features a sensor-controlled woofer.



Photo 12: The Pro-Ject Pre Box S2 Digital is very small.

About the Author

Ward Maas is the owner of Pilgham Audio. He studied electronics, marketing, and amplifier design. During his career in consumer electronics, Ward worked in areas ranging from CD standardization to radio and television to personal GPS navigation. Ward has worked on an extreme low-noise magnetic cartridge preamplifier and several special amplifier products. As the CTO of "Witchworld," a theme park near Amsterdam, he also works with animatronics. He lives in Almere, Netherlands, with his wife and son.

Photo 10) with specially hollowed out tree trunks, nicely polished or specially treated bark. Not to forget, acoustically tuned. Yes, they sound very nice.

I also want to mention the German company Schanks (www.schanksaudio.com). It is concentrating on home-theater systems with all the DSP stuff and active speakers. The remarkable thing is that it uses a sensor-controlled woofer. The promise is an accurate deep bass from a small speaker. After the Grimm Audio subwoofer, it seems that Motional feedback is back again (see **Photo 11**).

I already mentioned the Austrian company Pro-Ject while discussing turntables. For years, it has had a product line that is sold under the family name BOX DESIGN (http://project-audio.com). According to the company's philosophy that "less is more" these are handy products fulfilling simple needs. A small amplifier, a phono preamp, but also a CD player, a tube phono amp, and so forth—literally a myriad of small and very small products that offer great value for money. This according to another of the company's philosophy that you should only pay for what you need.

This year, the company took it to the next level with the introduction of the S2 line and one product really stands out. The Pre Box S2 Digital (see **Photo 12**) is a 32-bit 768 kHz, DSD512 D/A converter, with hardware MQA decoding. Inputs for USB, coax or optical and besides RCA outputs, a Headphone amplifier. And all that at a price tag of €349 ($390). No you did not read that wrong €349. Let's just say, I want one.

Accessories

There were also many companies offering accessories in all shapes and uses. First, I would like to mention the new IsoAcoustics (www.isoacoustics.com) Gaia series isolation feet for speakers and subwoofers (see **Photo 13**). They come in three sizes (depending on the weight they have to carry) and all the material to set up your system (nuts and bolts, metric, imperial). Just what you need to quiet down your loudspeakers. Some say rocket science is involved here. Just ask the IsoAcoustics crew at a show near you.

Due to the "mains pollution" by switch mode power supplies or just by LED lamps, there is also a growing interest in all sort of mains cleaning equipment. All come with stories of how bad this is. You could go completely berserk with the battery-fed mains regenerators that could have military use (and a bit more). Others tried to convince us that the anodized red mains plug would solve our problems, while the green one would solve our neighbor's problems. I guess that is a growing market for the years to come.



Photo 13: The new IsoAcoustics Gaia series isolation feet come in three sizes.

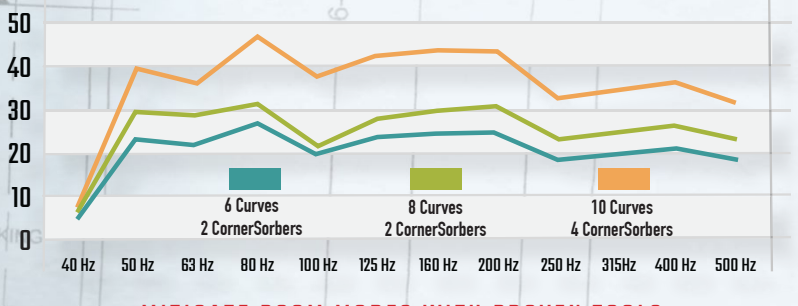


Photo 14: Hans M Strassner gives a talk on mains pollution.

Technology Stage

The Technology Stage offered lectures on all kinds of subjects regarding audio. One of the lectures was about Mains connection, Mains pollution, and filtering. Hans M. Strassner is a veteran on that subject who quietly explained what happens where and what you can do. What might (!) work and what certainly will not (see **Photo 14**). Of course, he is offering the remedies for sale.

Also on the Technology Stage, there was an interesting lecture by Bob Stuart who took the time to explain the underlying aspects of MQA. Unfortunately, it was not easy for the general audience to follow. It seems that the time smearing aspects of A/D and D/A converting have been neglected too much in the past years. However, I am curious to follow how the entire MQA story will evolve in the coming years as we get more MQA-encoded examples to compare.

PROVEN
Our Bass Absorbers WORK from 45Hz to over 200Hz.
YOU'VE GOT DESTRUCTIVE ROOM MODES and the only fix is bass absorbers ("bass traps") – they're critical to mitigating room modes. We've published accurate lab-tested results for our low-frequency absorbers – the CornerSorber™ and the Curve Diffusor™ both feature efficient MLV membranes in close-parallel placements. We also tested CornerSorbers and Curves in the same setup configurations you'll use in your room – so we *know* they work.
Here's the Proof:
COMBINED ABSORPTION, SABINS: CURVES WITH CORNERSORBERS


MITIGATE ROOM MODES WITH PROVEN TOOLS.
For our White Paper, and NWAA Labs Test Reports for Curves and CornerSorbers visit:
www.AcousticGeometry.com


ACOUSTIC GEOMETRY™
BY ACOUSTICAL SURFACES
www.AcousticGeometry.com



Photo 15: The Bugatti Veyron comes equipped with a Burmester sound system.

High End on Wheels

Of course, there is always a "High End on Wheels" section on the show. This year's "piece de resistance" was the Bugatti Veyron equipped with a Burmester sound system. Very impressive! Of course, you might run out of Autobahn while listening to your music. Still, it was no mean feat to get a sound system in that car (see **Photo 15**).

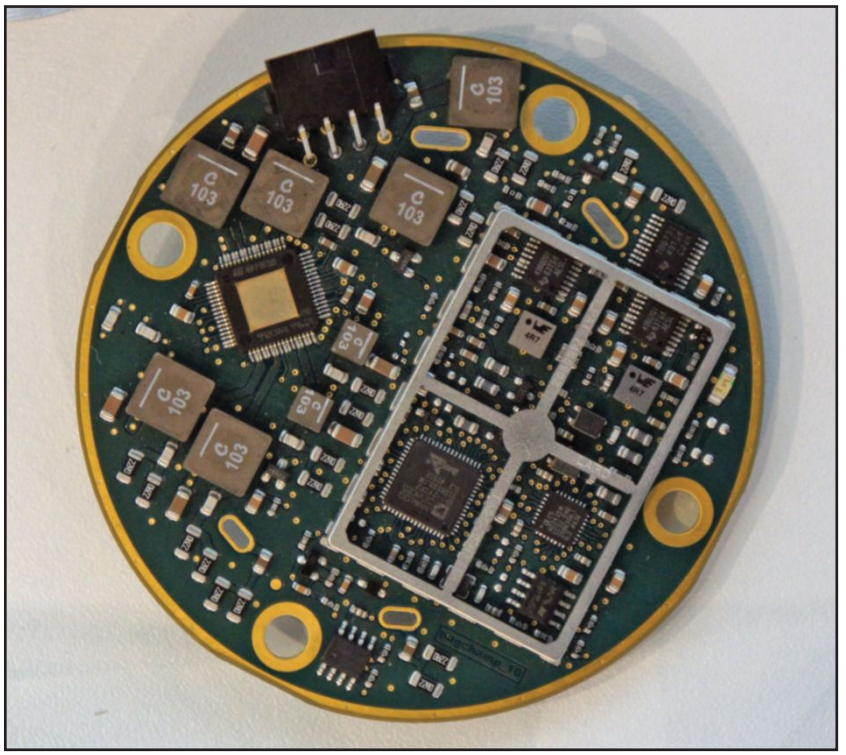


Photo 16: This 3D plus in-car audio system represents the future, according to Paragon.

More for the common man was the concept introduced by Paragon (www.paragon.ag). This company presented a 3D+ in-car audio system, with integrated amplification, DSP environment and communication hardware, all on a very compact board and using a digital communication bus, which can be nicely integrated with your car management system (see **Photo 16**).

Beyond High End

The number of potential exhibitors for the High End show is growing so much that the High End Society is already looking at expanding the exhibition grounds to nearby buildings. It's not a secret that a parallel exhibition is being promoted in a nearby hotel by certain companies that don't wish to be affiliated with the High End Society. Unfortunately, due to fact that some manufacturers are not finding the quality demonstrations that they want, they end up aligning themselves with such initiatives, which are not covered in this report. Hopefully, the expansion will put an end to this.

But there are also other companies that simply want to show their new products or ideas during the annual event, exclusively to the trade, using nearby hotels for that purpose. And sometimes those companies allow the press to have an exclusive first look at such novelties.

That's how we had the opportunity to find Electrocompaniet from Norway (www.electrocompaniet.no), showing its newest product range at a beautiful Bavarian hotel in Munich. Next year, the company intends to return to the MOC and show its new products to the world. All I can say for now is that I am very impressed with its wireless active speaker and streamer, the EC Living Tana.

In another hotel (invitation only), we found Zoltàn Bay from Bayz Audio (www.bayz-audio.eu), demonstrating his carefully designed and very special audio system. The Bay Radial Speaker is a cylindrical membrane surface capable of changing in diameter. It is indeed an omnidirectional speaker and, it seems that due to the large surface area there is less energy concentration in a single direction. The polar diagram is a nice circle at 2 kHz and still in very good shape at 20 kHz. Distortion, pulse behavior, waterfall diagram are all impressive. The sound is one of the best I have ever heard. A bit like an electrostatic speaker. We also recommend a look at the system's amplifiers if you can! Brilliant.

Well after tons of impressions in a short time I was really exhausted. I was glad that on the Sunday the show closed at the new earlier time of 4 PM. Time for a High End Bavarian Beer!


THE
LOUDSPEAKER
SOURCING SHOW
CONSUMER ☼ COMMERCIAL ☼ AUDIOPHILE
MUSICAL INSTRUMENT ☼ PROFESSIONAL


☼ MEET DIRECTLY WITH THE BEST SUPPLIERS
☼ FIND ALL OF YOUR LOUDSPEAKER SOLUTIONS
☼ TECHNICAL REPRESENTATIVES AVAILABLE
☼ CHINESE BUSINESS LEGAL SERVICE ADVICE
☼ HEADPHONE AND MICROPHONE SUPPLIERS
OCT 16, 17 & 18, 2017 NANSHA GRAND HOTEL
NANSHA, CHINA
www.loudspeakersourcingshow.com

Celemony Melodyne 4

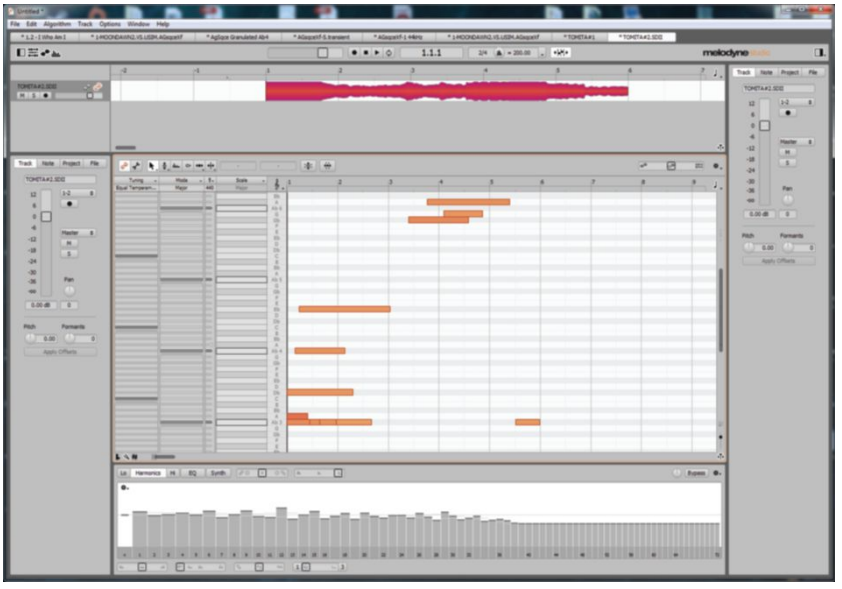
A Different Way to Look at Audio



Celemony (the company) and Melodyne (the software) are currently associated with two important technologies and acronyms familiar to Digital Audio Workstation (DAW) users: ARA and DNA. ARA stands for Audio Random Access, while DNA, in this case, stands for Direct Note Access.

By
Fernando Rodrigues
(Portugal)

Direct Note Access (DNA) is the main technology behind Melodyne, and what makes it unique. Basically, it analyzes the audio to determine the main frequency content (notes) in it and displays those notes in "piano roll" style, where we can individually select them and manipulate pitches and durations at will. Initially, it only worked in monophonic files, but the algorithm was later optimized to also enable note access on polyphonic material.



One of the most powerful tools in Melodyne is the possibility to manipulate the audio spectrum by altering its harmonics. Here we can see a portion of the audio (zoomed out enough to not see the envelopes) and a note selected. In the bottom of the window, we have the different partials, displayed as bars, to manipulate at our disposal. In the upper part, we can see the audio contour.

Audio Random Access (ARA) technology was developed later, jointly with PreSonus. It was first introduced in PreSonus' Studio One Pro v2 DAW software. What it does, is to free the user from analyzing a full file (more on that later) for advanced processing, enabling audio to be randomly accessed, based on user selection. ARA enables Melodyne to work faster as a plug-in, inside any DAW. After PreSonus, ARA gained support from Cakewalk, with Sonar X3, followed by Tracktion, with Tracktion 5. Currently, besides those three DAW solutions, also Samplitude and Mixcraft support ARA.

While these technologies are indeed important, they are not the main focus for this article. Of course, we will be discussing them while we review the features offered by the latest version of Celemony's flagship product—Melodyne Studio 4.

Melodyne was introduced in 2001, creating the possibility of working with audio in the same way we were used to working with MIDI: This meant dragging notes up and down to change pitch, or shrinking/enlarging them to change duration. Working like this with audio was like a dream. Celemony—a company from Munich, Germany,

founded by Peter Neubäcker, Prof. Dr. Hildegard Sourgens, and Carsten Gehle—did it. Now, 17 years later, Melodyne Studio has entered its fourth incarnation (in fact, it's already in Version 4.1, which saw improvements in ARA implementation).

What's New and What's Hot

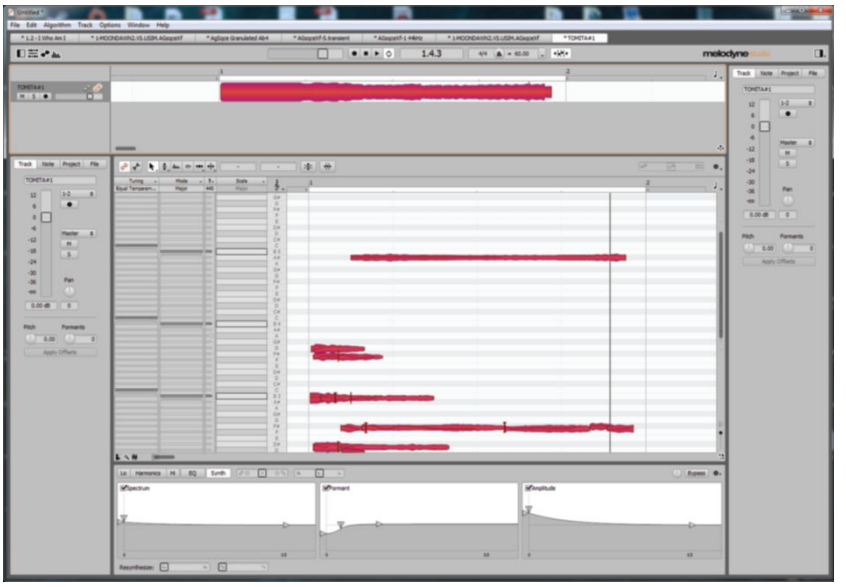
Almost everything that's new is hot but some things in the new Melodyne Studio (the top version of Melodyne) are hotter. First, Melodyne now works almost as an audio editor, in the way we are used to with the likes of Sound Forge or Wavelab. We say almost, because the way we see things is not common. While discussing the new features, we will mention some that were newly added in Version 4.1, a recent, free upgrade to all 4.0 users.

First, the Main Editing Window. This one received a very important improvement. Besides working with ARA, it now enables the processing of several audio tracks simultaneously, almost as within a DAW. This is important, because now we can work with as many tracks as we want (if we have the processing power to do so).

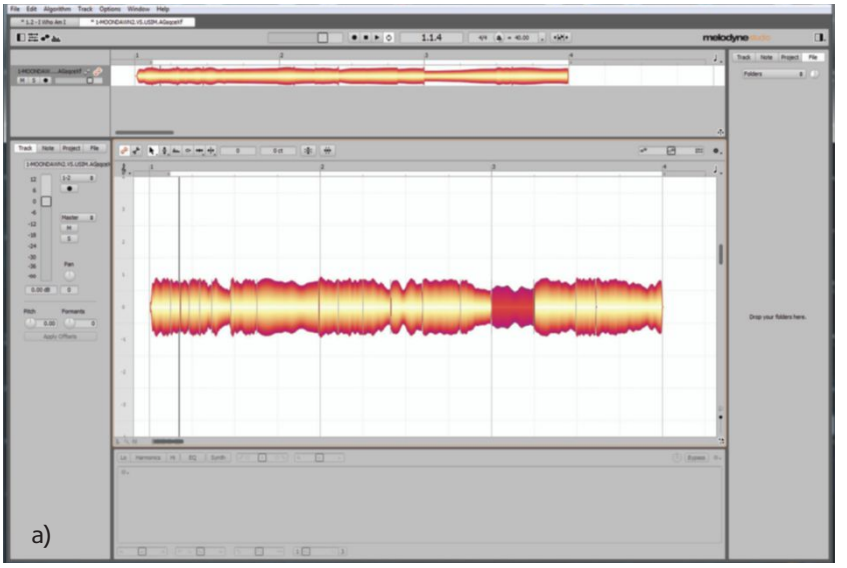
Before ARA (and still in DAWs not currently compatible with ARA), we had to "transfer" audio for the DAW into Melodyne. This is how we worked with an external audio editor from within a DAW. It's a convoluted process, takes time, and is not very friendly. With ARA, no "transfer" is necessary.
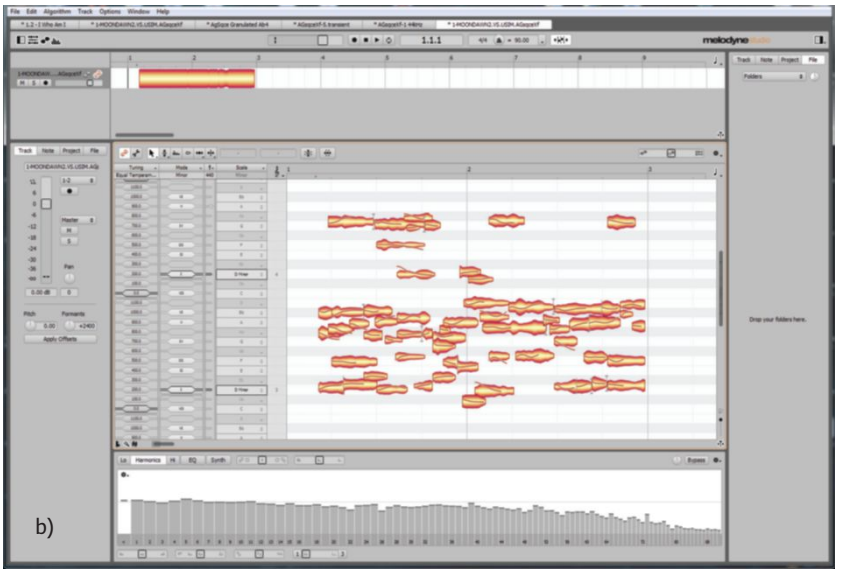
There are two alternative modes—Track Mode and Clip Mode. Track Mode lets us see the entire contents of the track open in Melodyne, independently of how many clips it is composed of in the DAW. In Clip Mode, we see only a single clip from the DAW track at a time. The track list remains in view, but in Clip Mode, the Edit and Reference buttons are grayed out. Track Mode and Clip Mode differ in the way notes are displayed at clip borders. In Track Mode, only notes lying within the clip borders determined by the DAW can be seen. In Clip Mode, notes on either side of the borders remain visible. Therefore, in Melodyne, we see—in the area with a gray background—what we might hear if we were to resize the clip in the DAW.

Then, there is the Sound Editor, the new pane editor in Melodyne. This editor greatly enhances the ability to manipulate audio inside Melodyne, thanks to what Celemony calls Polyphonic Spectral Shaping. Big words, but in fact, the feature is impressive. It is, in our opinion, one of the main new features in Melodyne 4, and the one that pleased us most—as people used to messing with sound in a creative way. Surely, it will not be the same way a regular studio works in regular recorded material, but the basic principle is the same. Melodyne decomposes the complex sound it their basic spectra (harmonic



We can also manipulate other audio characteristics. In the Synth tab, we have tools to manipulate the entire spectrum, the formants and the amplitude.

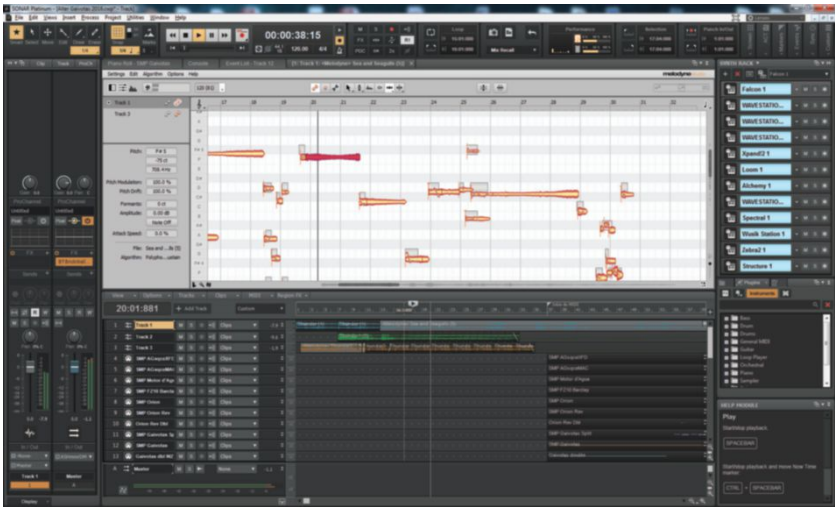




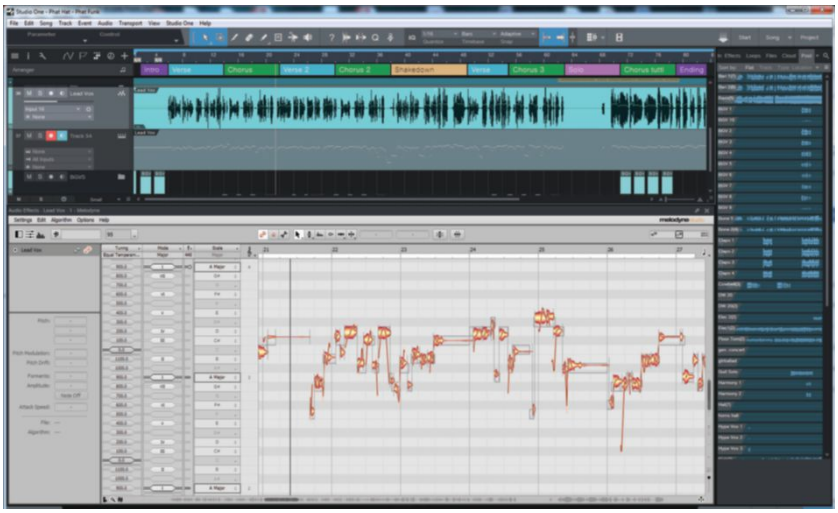
Melodyne enables tempi manipulation. By segmenting a note/melody, we can compress/extend the durations of each segment. We show two samples: Sample 1 (a) and Sample 2 (b).

components) displayed as bars in the lower part of the editor display. These bars can then be individually manipulated if we want to change the spectra of the audio (this is the creative manipulation we were talking about). But of course, if one wants to preserve the character, this should not be changed (at least not much). Yet, the way Melodyne manipulates the sound shifting and compression/expansion is greatly enhanced by this technology.

The Editor window has several panes. The top pane is the Track Pane, where all the selected tracks are displayed. This is, in fact, a Multitrack Editor since Version 4. We can select and have several tracks at our disposal. Once a track is selected, it is analyzed to separate the audio material in a graphic representation of notes. The analysis result is then displayed in the Note Editor. But we are not limited to just one track in this window. Thanks to a new feature in Version 4, we can now simultaneously display notes from several tracks. Using Celemony's own description of the feature:



Jointly developed by Celemony and Presonus, Audio Random Access (ARA) enables Melodyne to access any note without having to first analyze the audio. Here we can see Melodyne in action inside Cakewalk's SONAR also with ARA support.



Working together with Celemony in the development of the ARA technology, it's no surprise that Presonus' Studio One Pro is the DAW with the better Melodyne implementation.

"Consider the case where you have one track containing vocals and a second track with a guitar accompaniment. As soon as you switch the vocal track to edit mode, the notes of which it is composed appear in the Note Editor. Now, if you switch the guitar track to reference mode, its notes, too, are displayed in the Note Editor. The guitar 'blobs' in this case are gray and can neither be selected nor edited. They are simply displayed for the purpose of orientation in the background to the vocal notes. This makes following the melody child's play."

There are also differences between the stand-alone version and the plug-in version in the way Melodyne displays the tracks. In the stand-alone version, the tracks are displayed in the Track Editor, in ways similar to a DAW, in the plug-in version, multi-tracking is implemented as a collaboration between the DAW and the various instances of Melodyne. There is also a list of headers, on the left side, on which we can see which instance of the Melodyne plug-in is currently open, as well as any others inserted in the tracks of the current project. The concept enables users to work in one Melodyne plug-in window only, while being able to see and edit the contents in all instances. (No matter how many instances of Melodyne are loaded, we will always work in the same window.)

Last, at the bottom, we have the Sound Editor pane, which is the most important feature in the new Melodyne 4.

Audio from a Note Perspective

When Melodyne was created, DNA (which stands for Direct Note Access but could easily be interpreted as the "sound DNA" as well) clearly introduced a new and very special tool. Never before did we have the capability to access direct note editing in an audio file. The ability to fine tune a note, change it to a different pitch, fine tune its tempo by repositioning it and changing its duration, was the kind of wizardry that sound editors could only dream of.

Yet, that's exactly what Melodyne enabled. Obviously, things were limited in the beginning, and results sometimes generated unwanted artifacts but the features were there. With gradual refinement of the algorithms, things improved a lot. Version 3 offered notable improvements, while allowing access

to polyphonic material. This opened up completely new possibilities, and transported Melodyne to an entirely new universe. Of course, things still needed to be polished.

Version 4 introduces refinements in the algorithms. We could already change the position of the notes, the duration, and the pitch, but sometimes the notes displayed were not the right ones. In this new version, Celemony optimized the detection with two new algorithms—Polyphonic Sustain, better suited for strings, organs, vocals and all kinds of sustained notes; and Polyphonic Decay, for the sound of pianos, guitars, and the like.

Also, according to Celemony, the new Universal Algorithm is ideally suited for time-stretching and pitch-shifting of entire mixes (as long as we don't need access to individual notes, of course).

There are also new tempo algorithms that more accurately detect tempi, key signatures, and any changes in tempo. We can better correct tempi without removing the minor fluctuations that make the music "human." Also, we can synchronize recordings with different tempi or variations in tempi.

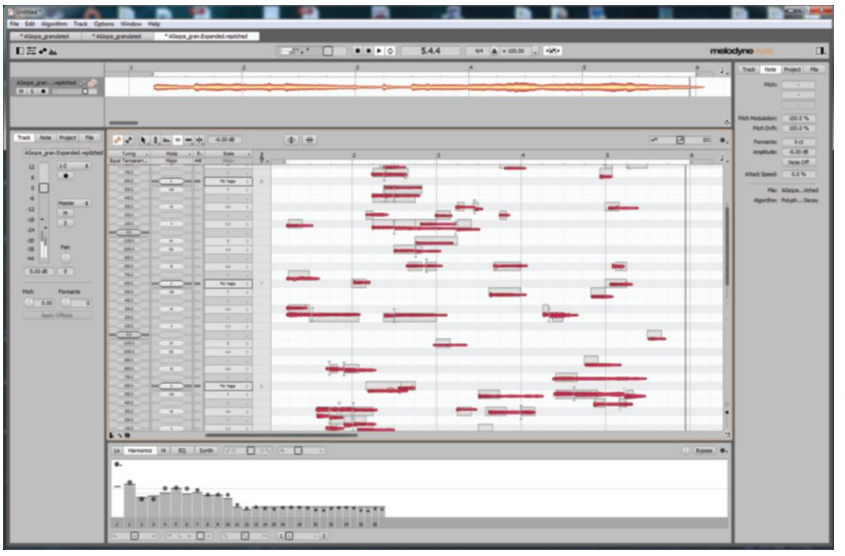
The mentioned Multitrack Note display now allows for a more precise editing of notes in context with an arrangement, enabling us to see what notes other instruments are playing. This may be precious for editing and fine tuning vocals or solos. Again, these tools are the ones that better suit the needs of the work in a studio, or to process live recordings.

At Work

We found working with the new version to be somehow friendlier (compared to previous ones). The new features in the Editor window in the Studio version, especially, are a breeze. We tested it with a complete track that included complex instrumentation and vocals, and Melodyne did an amazing job at separating the notes. We could immediately locate the vocals, and select them (among the sounds of the entire mastered track). We also tried loading separate tracks in an arrangement, working with it as a plug-in. More about that later.

Then, we tried Melodyne for our favorite task—sound editing and mangling. There were several things we wanted to try. Creating several samples from a single one, time-expanding some samples to the extreme to create some sort of "granular" picture of the original, and, of course, messing with the spectrum and the harmonics were some of Melodyne 4's new features that piqued our curiosity.

Starting with pitch-shifting and time expansion, we have several ways to accomplish this. The simplest one is globally. First, we must analyze the sound. As we already mentioned, Melodyne has several algorithms to accomplish this. To the existing Percussive and Melodic, Celemony has added the new Polyphonic Sustain and the Polyphonic Decay. Depending on the material and what we want to do, the selection of the right algorithm is very important. For pitch shifting, the selection of a melodic algorithm is what produces the best results. After analyzing the information, Melodyne displays the notes in the main section of the window. We can see the partial editing section, and on the left/right we have the global controls for pitch and formants, as well as volume and pan (see image above). These are in the first tab, called Track. There is another, called "note," which only becomes active if we select one of the notes in the main section.



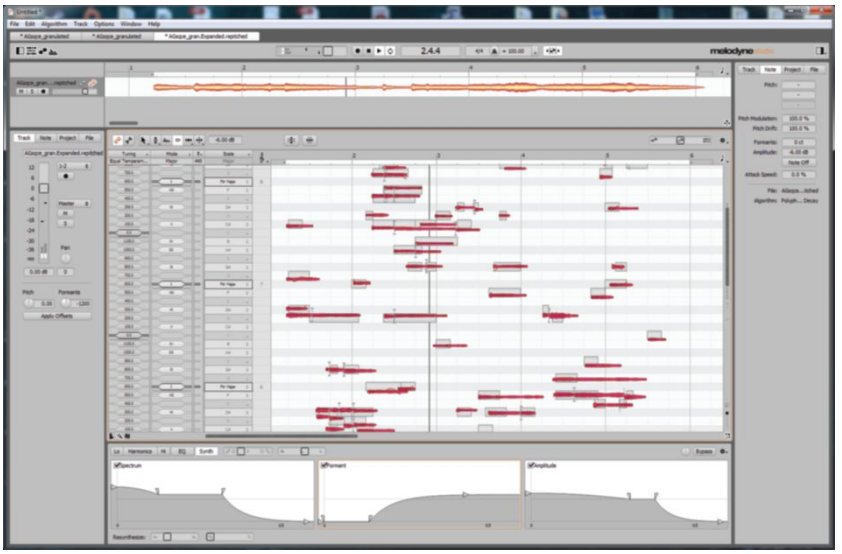
We can use Melodyne Sound Editor with a multi-track session for processing single files in a more creative way.

Now comes the really powerful editing. We can expand durations in individual notes inside the audio (but not compress). We can also shift the pitch up or down for each one of these individual notes. If we select a group of notes, we can edit globally on that group. All this as we usually do with MIDI tracks, except that in this case within the audio.

Of course, if we want to simply shift the pitch of the global track, we can do it by inserting the value in the Track controls in the left/right sections. For

About the Author

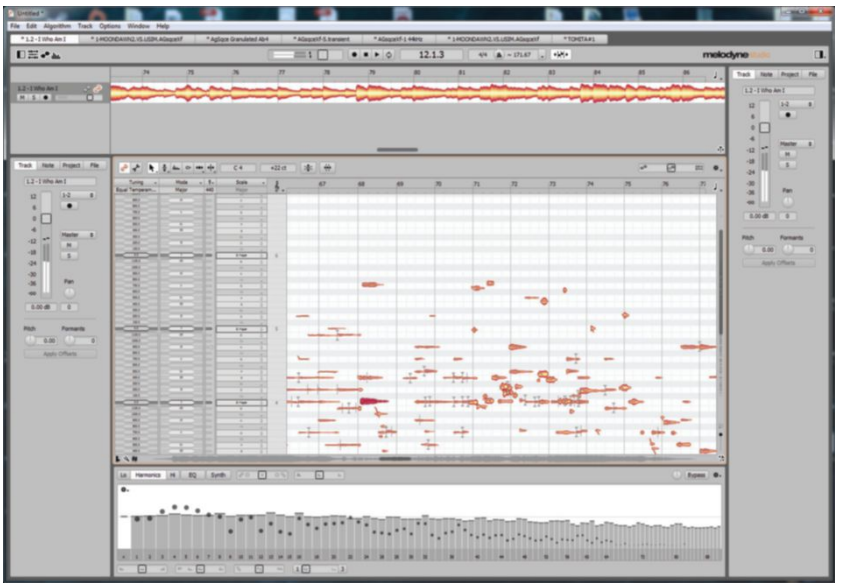
Fernando Rodrigues began studying music and technology in the 1970s. His goal was to marry his two passions: music and computers. As a student, he helped assemble the electronics music studio at the music college in Porto. Later, he directed the technology department at one of Portugal's major distributors while pursuing a career teaching musical analysis and composition techniques. He now concentrates on research and writing about music and technology, sharing his own perspectives about music and sound.



The Melodyne Synth tab toolbox enables users to access Spectrum, Formant, and Amplitude editing.

time compression/expansion of the entire track, it's as simple as changing the tempo in the section at right of the transport controls. Slower tempi will play the track slower, while faster tempi will play the track faster.

Really useful for sound designing (or the creation of "out of this world" sonic landscapes), is the ability to select all the notes in the note window and perform a time expansion on the notes themselves. Instead of globally expanding the track, this operation will merge the notes within, creating a complex cross fading sound. Of course, we must significantly lower the global volume before doing this, to avoid clipping.



The Multitrack Note display allows for a more precise editing of notes in context with an arrangement, enabling us to see what notes other instruments are playing.

After analyzing the notes, that's when audio engineers can perform pitch corrections and time corrections for each audio track. Melodyne performs greatly in these tasks, correcting out of tune notes for singers, or bad tempi in rhythmic tracks. As we saw, being able to edit each note individually is powerful stuff.

The Editor and Audio Mangling

We mentioned an example of sound mangling, but there are many other things we can do with Melodyne. We can edit the partials (harmonics) in the file (globally). This can drastically alter the character. We can also change the formants without touching the pitch (e.g., to transform a female voice into a male one, or vice-versa,) which can also be useful in sound-design, to transform the tone quality. We didn't mention it yet, but the global formant correction and pitch shifting in the Track section is performed in real time, which means that we can listen to the track while we change the parameter until we find the optimal result.

We cannot compress the duration of individual notes (we can just expand). But we can alter the attack speeds. Make the notes "punchier" or "softer." Also, when performing the already mentioned expansion of the note durations, we can reposition some of them, to create different transitions within the global sound. It is a great way to completely transform a sound into something else.

Another powerful set of tools is the mentioned Polyphonic Spectral Shaping Tools. These are in the Sound Editor, which is the lower section of the composite Melodyne window, only available when we have the sound analyzed with one of the pitched algorithms (Melodic, Polyphonic Sustain, or Polyphonic Decay).

The tools are all powerful, but the most important are the ones related with the harmonics, of course. In these, we can manipulate the partials in an entire view or divided between Hi and Lo. The partials are edited one by one, which can be tedious, but in the lower bar we also have some tools that manipulate things as a whole, affecting Brilliance, Contour, Odd/Even balance, or Comb (special arrangements of harmonics).

Finally, we have the Synth tab. This has several controls for changing the spectrum. Again, we have global controls both above and below the main tools, designed to edit the Emphasis, Dynamics, and then Resynthesize Magnitudes and Phases of the harmonics. In the main part of this tab, we have

controls for global editing of the Spectrum, Formant, and Amplitude. These assume the form of a kind of ADSR envelopes.

The "Plug-In" and ARA Integration

Working with DAWs that support ARA is not as straightforward. We tried it with Presonus Studio One Pro Version 3 and Cakewalk SONAR Platinum and we had to resort to reading the manual. The benefit is that the notes detected appear as events over the audio track. We can then drag them to an instrument track, and we then have a MIDI track that perfectly duplicates the audio performance. Very cool and handy for creating double tracks with virtual instruments, for example.

Audio editing can be performed in real time, but the result only becomes permanent after rendering. So, everything done before rendering can be reverted.

On DAWs that don't support ARA, the use of Melodyne is not as straightforward, of course. We also tried it in Cubase Pro—the working is similar in any other DAW—and the workflow is not that great. Yet, it is the only way to work with Melodyne in the DAWs that don't support ARA. Although the ARA integration would be the best option for pitch corrections and more surgical editing, in the case of DAWs that don't support it, we found off-line editing in Melodyne Studio much better.

Conclusions

The new Melodyne Studio does wonders! As a pitch-tuning and duration-correction tool, it is invaluable. The possibility of individually editing notes within a sound file, even polyphonic, is a dream. But Melodyne is also invaluable for sound mangling and sound design in general. The time compression/expansion and pitch shifting capabilities are top notch. Even with extreme settings we can get great results, provided we choose the right algorithm and are careful with the parameters. The learning process to get the best results can be long, but it is always rewarding and exciting.

This is even more true for the new Polyphonic Spectral Shaping tools. The toolset offers enormous power, which comes at a price, demanding careful studying of the different capabilities and how to better use them to get results. Impatient users will easily get the wrong idea and become disappointed.

For those brave enough to embrace the learning curve, Melodyne can be extremely rewarding. A big thumbs-up to Celemony for the software and its continuous support and continuous updates. Anyone serious about audio and music needs to try it.

Melodyne Studio Audio Editor and Plug-in
Celemony Software GmbH
Valleystrasse 25
81371 Munich, Germany
www.celemony.com
*Other editions available: Essential, Assistant, and Editor
All editions are 64-bit and 32-bit compatible and can operate as plug-ins or stand-alone, supporting VST, AU, AAX, RTAS, ARA and Rewire (client) as well as being compatible, with each other.

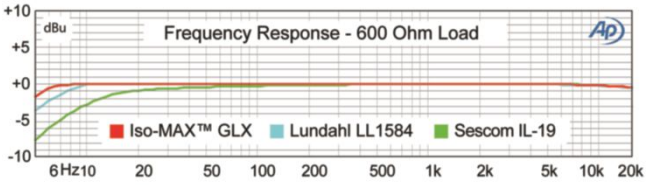
Iso-Max® GLX isolator... dwarfs the others!

When it comes to performance, build quality, warranty and value, the Jensen Iso-Max GLX™ is in a class of it's own!

Jensen Iso-Max... noise elimination done right!



Ground loop isolator comparison	Iso-Max GLX	Lundahl LL1584	Sescom IL-19
Freq Response	10Hz ~ 30kHz	10Hz ~ 30kHz	40Hz ~ 30kHz
Distortion @ 20Hz:	.001%	.01%	.1%
Distortion @ +4dB:	.005%	.06%	.3%
Phase shift @ 20Hz:	0°	0°	20°
Ground lift switch:	Yes	No	No
Construction:	Metal	Metal	Plastic
Warranty:	3 Years	1 Year	1 year
MAP Price (estimated street)	$69	$130	$45



Jensen Transformers Inc.
9304 Deering Avenue
Chatsworth, CA 91311
Phone: (818) 374-5857
©2016 Jensen Transformers Inc. All rights reserved.
Specifications and appearance subject to change without notice.
www.jensen-transformers.com

jensen™ transformers



Primacoustic Recoil Stabilizers

A Solid Speaker Base



Photo 1: Primacoustic's Recoil Stabilizer is designed to provide a rock-solid foundation for a monitor speaker, holding the speaker firmly in one position, and isolating the supporting surface from vibration.

Radial began importing acoustical treatment products from Europe 20 years ago, but became frustrated with them because most were designed for industrial noise control rather than music. In 2000, the company created Primacoustic to provide practical acoustical solutions to the problems encountered in recording studios, performance venues, churches, and home listening environments. Most recently it has tackled the problem of loudspeaker stability and isolation.

By
Gary Galo
(United States)

In the October 2015 issue of *audioXpress*, I reviewed the IsoAcoustics Aperta Speaker Stands, which I found to be highly effective at preventing vibration transmission from the loudspeaker to the surface below. The IsoAcoustics Aperta stands actually enable the speaker and the top portion of the stand to move, while the bottom portion of the stand remains stationary (www.isoacoustics.com). In IsoAcoustics' own words, "Decoupling the audio speakers or studio monitors from the supporting surface reduces the energy from exciting other objects, and allows the speaker and enclosure to float independently."

Primacoustic Recoil Stabilizer
Primacoustic
Division of Radial Engineering
1588 Kebet Way, Port Coquitlam
BC Canada V3C 5M5
604-942-1001
info@primacoustic.com
www.primacoustic.com
RX7-HF: $99.99 (other sizes vary)

Recoil Stabilizer

In designing the Recoil Stabilizers (see **Photo 1**), Primacoustic has taken the more conventional view that the loudspeaker should never move. Primacoustic has chosen to prevent recoil, noting that "By reducing the backward 'recoil' as the speaker coil pushes energy forward, the initial transient no longer suffers lag and the sharpness of the resulting impulse is more defined."

The Recoil Stabilizer incorporates four key components to achieve the desired combination of isolation and stability (see **Photo 2**). Isolation is provided by a high-density, open-cell urethane foam base measuring 2 1/4-inches (57 mm) thick. The foam prevents resonant frequencies from travelling from the loudspeaker to the supporting shelf. This open-cell foam is dense enough to support the weight of the speaker, and soft and thick enough to provide isolation from the surface below. Foam can deteriorate over time, due to oxidation and the effects of ultra-violet radiation. Primacoustic embeds the foam with carbon to lessen these effects, extending the life of the foam.

The foam is glued to a 0.25" thick (6.4 mm) laser-cut steel plate, which places a heavy mass on top of the foam and acts as a platform for the speaker, and as a stabilizing counterforce to the movement of the low-frequency driver. The steel base is shaped into a curved "L" in front to minimize standing waves; the extra steel also adds mass to the base.

The top of the steel plate is covered with a 0.125" thick (3.2 mm), non-skid neoprene pad, which ensures that the speaker remains securely in place, and does not move as a result of the cone motion of the low-frequency driver. Without any speaker

mounted on the stabilizer, the steel plate rings when you strike it. But, the stabilizer is designed so that the speaker's weight dampens the vibration of the steel, resulting in an inert thud when struck.

Radial manufactures the Recoil Stabilizer in 12 different models (see **Photo 3**), which range in size from 7.5" × 9.5" (241 mm × 190.5 mm) up to 20" × 22" (508 mm × 559 mm). The two smaller models—the RX5 and the RX7—are available either flat, angled down 5°, or angled up 10°. The mid-sized RX9 and RX12 are available flat or angled down 5°. The RX17 and the RX20, which are the two largest, are only available flat, since they're intended for loudspeakers too large for desktop or shelf-mounted applications.

At my request, Radial Engineering supplied the RX7-HF for review, since it's just the right size for the Monitor Audio Bronze BX2 loudspeakers that I use on my editing and audio restoration computer at The Crane School of Music. This model measures 10.375" × 13" (263 mm × 330 mm) and weighs a hefty 10.8 lbs. (4.9 kg).

I found the Recoil Stabilizer to be a highly-effective product. Clarity and definition improved across the spectrum, but especially in the bass region. The stereo image is rendered with greater precision, and there is a noticeable reduction in transient smearing. Primacoustic doesn't publish any isolation specifications for the Recoil Stabilizer, but I conducted a simple, admittedly subjective test of its isolation effectiveness. On an orchestral recording with loud bass drum material, I could feel a pronounced vibration on the sides of my Monitor Audio loudspeaker enclosures. On the shelf below the Recoil Stabilizer, the vibration was all but eliminated. Primacoustic has testimonials from several dozen audio and recording industry professionals, as well as musicians, on its website. If your speakers need stability and isolation—which they probably do—the Recoil Stabilizers deserve serious consideration.

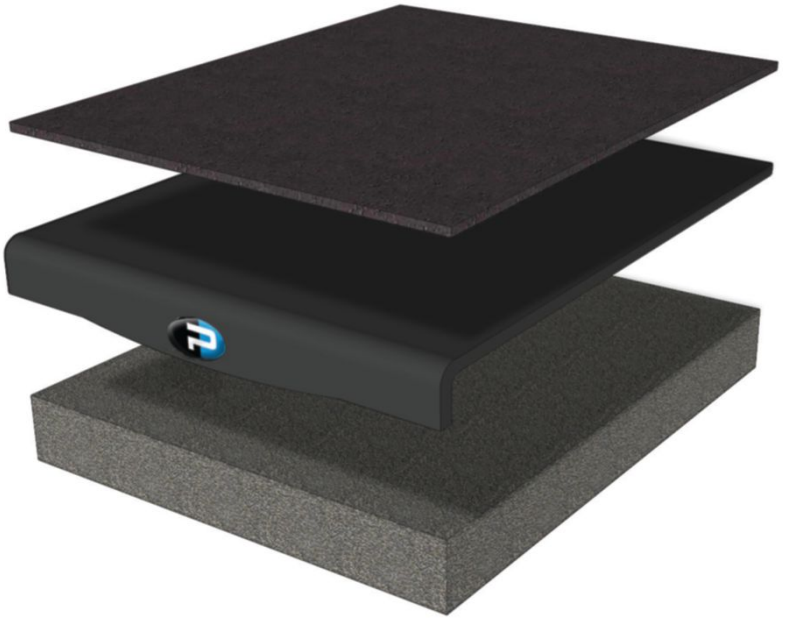


Photo 2: The cutaway view of the Recoil Stabilizer shows the design components. From bottom to top, the high-density, open-cell urethane foam isolation pad, steel stabilizing plate, front curved lip to prevent standing waves, and non-slip neoprene foam covering.

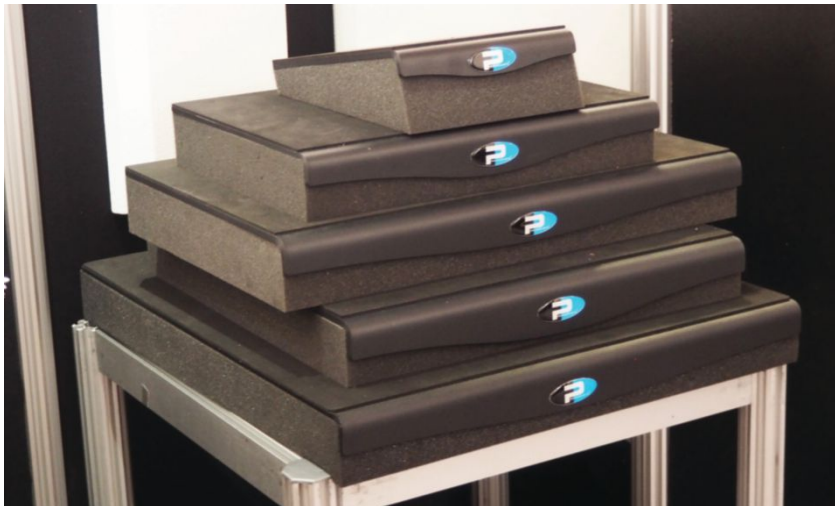


Photo 3: Primacoustic's Recoil Stabilizer is available in various sizes, weights, and firing angles to address the vast array of speaker designs currently on the market and personal mounting preferences.

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.

Manufacturer's Specifications

Primacoustic Recoil Stabilizer

Deck: 0.25" steel, black powder coated with 0.125" neoprene pad
Base: High-density polyurethane foam
Number per box: 1
Weight:
RX5 - 5.75 lbs. (2.6 kg)
RX7 - 10.8 lbs. (4.9 kg)
RX9 - 12.5 lbs. (5.7 kg)
RX12 - 22 lbs. (10 kg)
RX17 - 22 lbs. (10 kg)
RX20 - 36 lbs. (16.3 kg)

Acoustics and Microphone Placement



Even if you do audio recording using a cell phone the acoustical principles of good microphone placement still apply.

By
Richard Honeycutt
(United States)

In 1960, when I first became interested in audio, there were frequent articles in *Popular Electronics*, *Electronics World*, *Modern Electronics*, and *Audio Magazine* concerning the art and science of microphone placement for audio recording. Many amateurs were still making only mono recordings, and there was an ongoing debate among purists, who argued for using a single well-placed microphone, and "modernists," who supported multi-microphone techniques, even close-miking all the instruments in an orchestra, as was done for the *Lawrence Welk* television show.

Today, most amateur recording is done by non-technical family and friends of the performers, using cell phones. A few companies do provide on-site recording of audio and/or video, using professional or semi-pro equipment. But the principles of acoustics apply to all audio recording systems, regardless of type or cost.

The first consideration should be, "What microphone should I use?" Although the answer may seem obvious if you're using a cell phone to record, you might want to consider that different cell phones have different quality levels of microphones and on-board audio. A Google search for cell phone audio quality will bring up some information and many opinions. If you like the portability of a cell phone, but want better recording quality, you might want to spring for a really good microphone (e.g., one by Studio Six Digital) or a portable digital recorder (e.g., those made by Olympus, Tascam, Zoom, etc.).

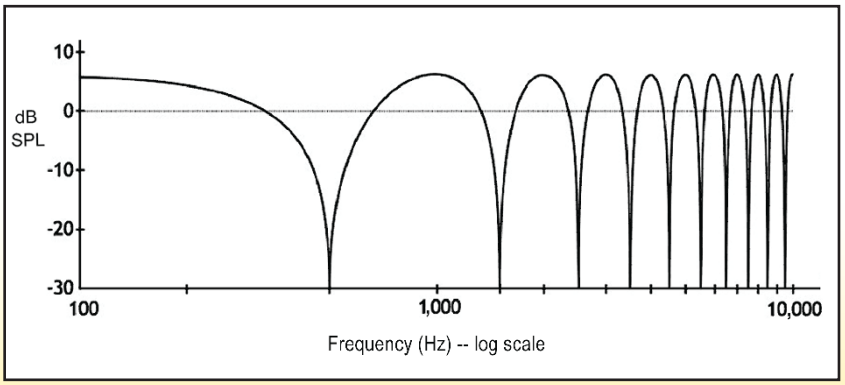


Figure 1: This depicts the sum of the signals from two microphones, each picking up a frequency sweep at the same level, but with a 1-ms time difference between sound arrivals at the two microphones.

Inverse Distance

Irrespective of the microphone choice, you can only get a good recording if you consider certain acoustical principles. The first is a phenomenon often called "spreading loss." In fact, this principle is not truly about loss because it does not involve conversion of acoustical energy to heat, but the name persists in the industry. The sound pressure from a source decreases in inverse proportion to the distance between source and receiver (i.e., mic, ear, or whatever). Strictly, this applies only

to sources that are vanishingly small compared to the wavelength of the sound—so-called "point sources." Sound pressure is usually specified as sound pressure level (SPL) in decibels, so we can say that the decrease in SPL between a receiver located—say—1 m from the source and the SPL that is 2 m from the source is 6 dB. We have doubled the distance, so the sound pressure is halved, and a factor of half in pressure corresponds to -6 dB.

This diminution of sound pressure with distance is called the "Inverse Distance Law" and is often confused with the Inverse Square Law, which states that sound intensity decreases as the inverse square of the distance from the source. Sound intensity, which is power per unit area, is proportional to sound pressure squared, so a halving of sound pressure represents a quartering of sound intensity. In decibels, intensity still drops at 6 dB per doubling of distance, since a 6 dB change in power or intensity represents a ratio of 4.

The effect of the Inverse Distance Law means that a microphone picking up sound from two or more sources should ideally be placed equidistant from the sources. Thus, using a single microphone to record a group often involves placing the microphone above the group. Doing so can have unfortunate consequences for singers, whose high frequencies do not project well from the tops of their heads; or for violins, whose high frequencies dominate the sound radiated upward.

Interference

Another vital acoustical principle is interference. This consideration comes into play when multiple microphones are used. Acoustical interference is the constructive (adding) and destructive (canceling) effect of instantaneous sound pressure from the same source arriving at two microphones spaced some distance apart, with their signals subsequently mixed (added) together. The results of interference are illustrated in **Figure 1**.

This effect is called "comb filtering" because of the shape of the resulting response graph. Comb filtering also occurs in sound playback or reinforcement if a given listener is in the sound field of two or more speakers. Notice the depth of the ripples in response is 36 dB. When the waves arrive at both microphones in phase, there is a 6 dB adding effect, but when the two waves are exactly out of phase, there is a 30 dB canceling effect.

If one microphone is enough farther from the source that the sound reaching it is 3 dB lower in level than the sound reaching the other microphone, the ripple depth is less. For example, if microphone 1 is 5′ from the source, and microphone 2 is 3.5′ from the source, the ripple depth will be 16 dB. In practice, if the same source must be picked up by multiple mics, the 3:1 rule applies: the furthest microphone should be at least three times as far from the source as the nearest microphone.

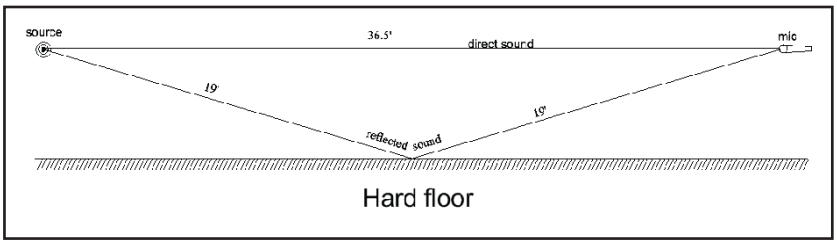


Figure 2: Comb filtering can result from sound reflecting off hard surfaces.

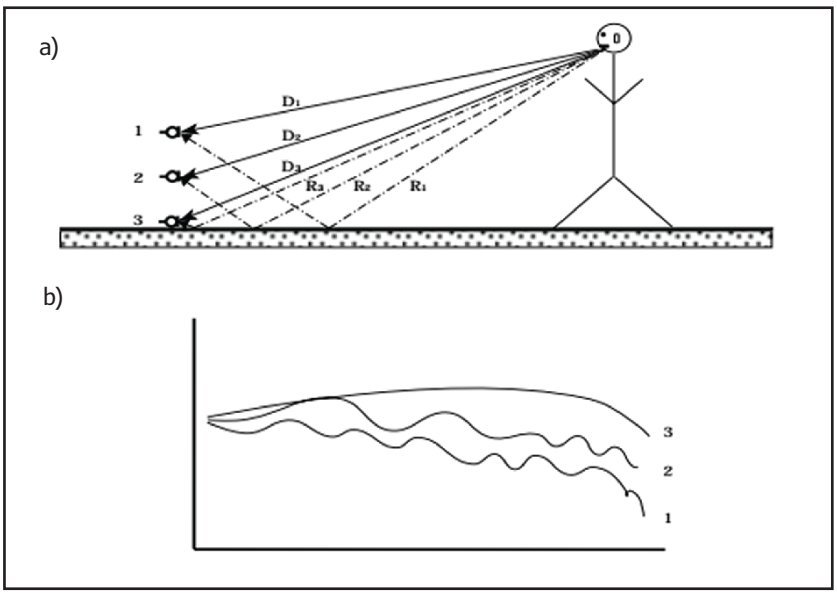


Figure 3a: Direct and Reflected paths vary for sound arriving at microphones placed at three different distances from the floor. b: The frequency response illustrates the comb filtering produced by three different microphone heights. Position 3 represents a boundary microphone.

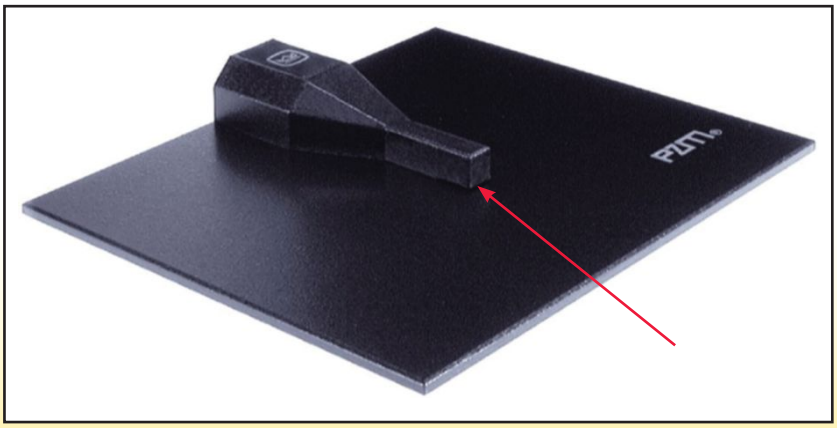


Photo 1: This pressure-zone microphone (PZM) made by Crown International has the sound-entry port located very close to the metal plate, as shown by the red arrow.

Comb Filtering

Comb filtering also occurs with a single microphone if the microphone picks up both direct sound and a strong reflection from the source, as illustrated in **Figure 2**. In this case, the reflected sound travels 38′ from the source to the microphone, while the direct sound only travels 36.5′: a time-of-travel difference of 1.36 ms.

Comb filtering is of concern in single-microphone mono recording only in the case of reflections. When recording using a cell phone's built-in microphone, the user is usually in the audience, and the people in front of him/her do a pretty good job of absorbing reflections. For more sophisticated setups using actual recording microphones, pressure-zone microphones (PZMs)—sometimes called "boundary mics"—can be used (see **Photo 1**). These microphones have the element mounted so close to the floor or to a boundary plate that the time-of-travel difference between the direct and reflected sound is negligible (see **Figure 3**[1]).

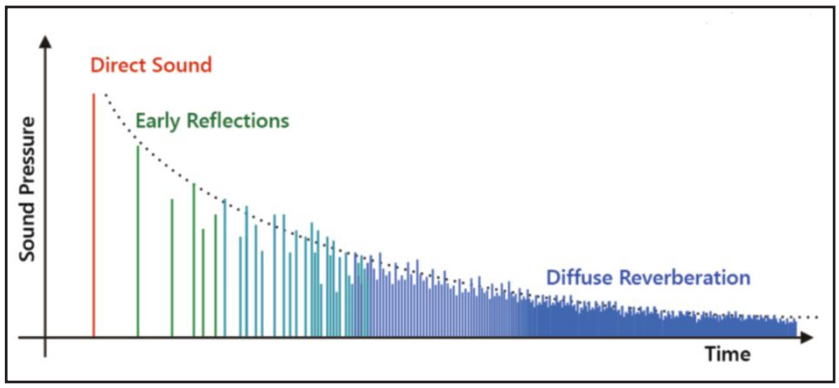


Figure 4: The sound captured by a microphone can be grouped by time of arrival as direct, early reflections, and diffuse reverberation.

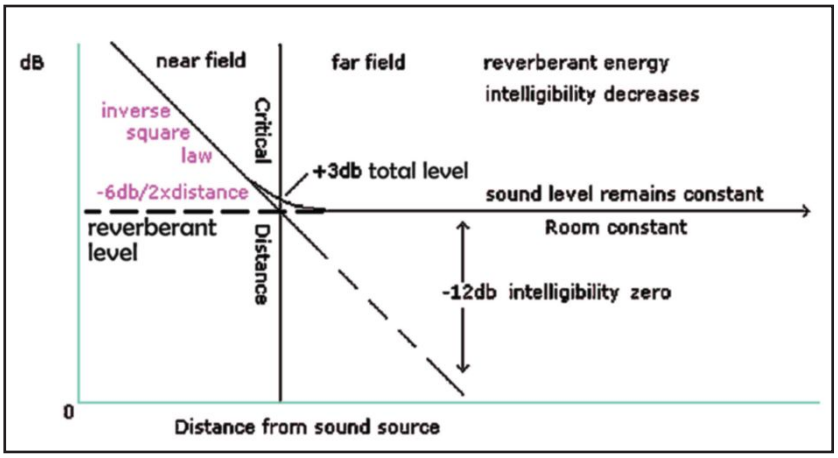


Figure 5: The critical distance is the distance from the sound source at which the direct and reverberant levels are equal.

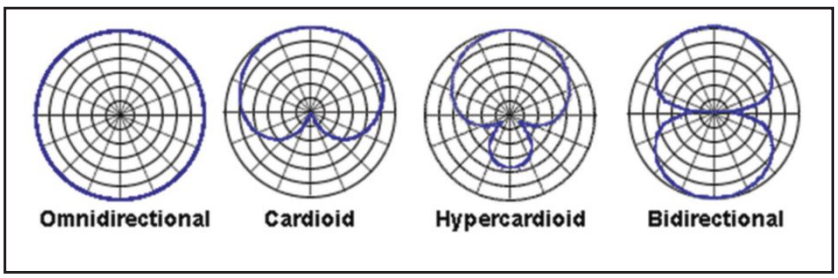


Figure 6: These are the four most common microphone directionality patterns.

Reverberation

Reverberation is another important consideration in optimum microphone placement. When a sound in a room is picked up by a microphone, the direct sound from the source arrives at the microphone first. If the microphone has been selected or placed so as to avoid reflections near enough in time to cause comb filtering, the next sound arriving at the microphone is early reflections from room boundaries, mainly walls and ceiling. The human ear/brain system uses these early reflections to determine the size of the room. Early reflections also contribute to the listener's sense of "presence" and "apparent source width." Apparent source width is controlled by the early sidewall reflections. Finally, the diffuse reverberation or "reverberant tail" begins, lasting until the reverberant sound has weakened to imperceptibility. **Figure 4**[2] illustrates this concept.

As the listener or microphone moves away from the source, the level of the direct sound decreases at a rate of 6 dB per doubling of distance, but the level of reverberant sound is the same throughout the room in most listening environments. There is a certain distance from the source called the "critical distance," at which the direct and reverberant sound levels are the same (see **Figure 5**[3]).

The change in direct:reverberant ratio has three significant effects. First, as the microphone is moved farther from the sound source, the added reverberance decreases the sense of intimacy. Second, since the acoustical absorption in most venues is greater at high frequencies, the reverberant field has less high-frequency sound energy than the direct field. Recordings made from the back of an auditorium will be lacking in presence. When listening to a performance from the back row, we may not notice the lack of presence, because the ear/brain factors out the effects of the acoustical environment. But those listening to the recording later will not have that benefit, and the lack of presence will at least be noticeable, if not outright disturbing.

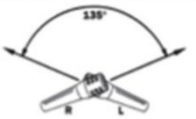
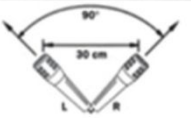
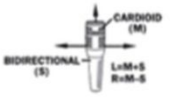
STEREO PICKUP SYSTEMS	MICROPHONE TYPES	MICROPHONE POSITIONS	
X-Y	2 - CARDIOID	AXES OF MAXIMUM RESPONSE AT 135° SPACING: COINCIDENT	
ORTF (FRENCH BROADCASTING ORGANIZATION)	2 - CARDIOID	AXES OF MAXIMUM RESPONSE AT 110° SPACING: NEAR-COINCIDENT (7 IN.)	
NOS (DUTCH BROADCASTING FOUNDATION)	2 - CARDIOID	AXES OF MAXIMUM RESPONSE AT 90° SPACING: NEAR-COINCIDENT (12 IN.)	
MS (MID-SIDE)	1 - CARDIOID 1 - BIDIRECTIONAL	CARDIOID FORWARD-POINTED; BIDIRECTIONAL SIDE-POINTED; SPACING: COINCIDENT	
SPACED	2 - CARDIOID OR 2 - OMNIDIRECTIONAL	ANGLE AS DESIRED SPACING: 3-10 FT.	

Table 1: Coincident microphone techniques can help create a better stereo recording.

Next Generation Headphone Testing

for better test results

Consumers are demanding a higher definition sound experience. For manufacturers, this has given rise to a number of challenges when it comes to testing their products.

At G.R.A.S. we are proud to introduce the next generation of headphone testing technology.

After all, the better your testing, the better your products



G.R.A.S.
SOUND & VIBRATION

gras@gras.us | **800-579-GRAS**

www.gras.us/next-generation-headphone-testing

FREE
Software
FRONT PANELS
& ENCLOSURES
YOU DESIGN IT
WE MACHINE IT
NEW
DIGITAL PRINTING
FRONT PANEL EXPRESS
1-800-373-9060
www.frontpanelexpress.com



Photo 2: Some portable digital recorders incorporate X-Y or ORTF coincident microphones.

The third effect relates to the use of directional microphones. Microphones are available with various types of directionality, as illustrated in **Figure 6**[4]. Omnidirectional microphones generally have the flattest frequency response at all working distances. Directional microphones (except for Variable-D microphones) boost bass when placed close to the source—typically closer than about 2′ to 4′, depending on the design of the microphone. But even at larger distances from the source, directional microphones are usually not as flat as omnis, because of the more complex acoustical design of directional microphones.

Reverberation comes from all directions, and if the microphone is placed closer to the source than the critical distance, directional microphones reject the reverberant sound better than omni microphones used at the same distance. The effective difference in working distance from the standpoint of avoiding reverberation is called the "distance factor." A cardioid microphone has a distance factor of 1.7; a hypercardioid, 1.9; and bidirectional, 2.0. I personally always record using cardioid microphones, because placing an omni close enough to achieve the desired presence often runs me afoul of the "keep the mic equidistant from all sources" rule, resulting in poor balance among different instruments.

Close-miking individual instruments or vocalists is often done for live sound reinforcement in order to control feedback squeal. And it is an effective way of maintaining presence by avoiding picking up too much reverberation. But effective close miking of more than a very small group requires a lot of mixer channels, and even with a small group, following the 3:1 rule becomes a real challenge. And even if well done, close-miked recordings are devoid of room ambiance.

Although it is counter-intuitive, the use of a directional microphone beyond the critical distance has absolutely no benefit. We think that it would be helpful, but that is because we judge a sound's position of origin mainly by the time of first arrival of the sound at our ears. This enables us to tell where the sound is coming from, even when we are deep in the reverberant field. A microphone understands only sound levels and direction of travel of the sound wave, and cannot separate the direct from the reverberant sound when used at distances beyond the critical distance.

Additional Factors

One other acoustical phenomenon that affects microphone placement and presence is the absorption of sound by air. This has a small effect except in very large rooms or outdoor spaces. Air absorption depends upon the air temperature, humidity, and wind speed. It is usually negligible in calm wind for frequencies under 1 kHz, but attenuates sound at about 4 dB per 100 m distance at 4 kHz, and the attenuation increases rapidly at higher frequencies. This is one more reason not to record from the back of a large auditorium or gymnasium.

Stereo and surround recording introduce additional factors. If we could simply place

two microphones at the positions of a listener's ears, record, and then play back the recording for headphone listening, our job would be much simpler. (In fact, this is similar to what is done in making binaural recordings, but other significant factors must also be considered; thus binaural recording is a topic for another time.)

In normal stereo listening, the right- and left-channel sound mixes together acoustically in the listening room, creating something resembling a mono signal in the center between the speakers. But since that's where we sit while listening, we can also benefit from those other directional cues we mentioned earlier if we're hearing a stereo recording. And often stereo recordings are played on mono equipment. So we must consider what happens to a stereo recording when mixed to mono.

It would seem that all we need to properly record stereo is to use a single right-channel and a single left-channel microphone placed to pick up the appropriate sound. But this "spaced pair" arrangement will not mix well to mono, because the two microphones picking up sound from the same sources will produce comb filtering just as they would if mixed together into a mono recording. Also, the sound will not be very realistic to listeners in the "sweet spot" between the speakers: the stereo effect can be extreme.

Possible Solutions

In order to address this and other issues, a variety of coincident and near-coincident microphone techniques has been developed, as illustrated in **Table 1**[5]. The ORTF or X-Y coincident pair is incorporated in some portable digital recorders, as illustrated in **Photo 2**. The weakness of coincident pairs is that their stereo separation is limited.

My personal preference when recording a choir, concert band, jazz "big band," or orchestra is to use an X-Y pair mounted on a boom a few feet above the director's head (if the director will allow me to do so), and a spaced pair to provide greater stereo separation.

I know we live in a video age, and as Don Davis reminds us, "if bad sound were fatal, audio would be the leading cause of death."[6] But perhaps I have offered some healthy approaches to optimizing your amateur or professional on-site audio recording, regardless of the sophistication of your equipment.

References

[1] R. Streicher, "Stereo Boundary Microphone Techniques For Live Staged Performances," presented at the 21st Audio Engineering Society (AES) Conference, St. Petersburg, Russia, 2002. Adapted from John Eargle's *The Microphone Book,* Taylor & Francis, 2004.

[2] P. Bellinghausen, "Home Studio Acoustics: Speaker Placement," ProAudioBlog.CO.UK, http://proaudioblog.co.uk/2015/11/home-studio-monitoring-speaker-placement.

[3] "Room Acoustic: Absorption Diffusion," www.mh-audio.nl/tips%5CAcoustic.htm

[4] A.T.ChandlerVoice, "Check 1, 2: Microphone Technique, A Basic How To!" August 26, 2011, www.chandlervoice.com/single-post/2011/08/26/Check-1-2-Microphone-Technique-a-basic-how-to.

[5] E. Tarr, "4 Common Stereo Microphone Techniques," Pro Audio Files, https://theproaudiofiles.com/4-stereo-miking-techniques.

[6] Don and Carolyn Davis, *If Bad Sound Were Fatal, Audio Would Be the Leading Cause of Death,* AuthorHouse, 2004.

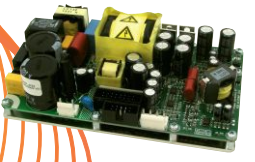
hypex NCxxxMP series

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

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



NC502MP

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



ncore audio power conversion

hypex electronics
www.hypex.nl

Acoustic Solutions Round-Up 2017

In our annual tradition for the Acoustic Focus issue of *audioXpress*, we highlight some innovative "first-approach" acoustic solutions launched in the past 12 months, which serve also as an introduction to the companies behind the products.

By
João Martins
(Editor-in-Chief)

Prosocoustic WaveRoom Kits

A First Approach to Acoustic Treatment

Prosocoustic is a relatively new player in the field, a US manufacturer of Acoustic Treatment Panels for commercial, professional, and residential use, founded in 2013. Starting with a clean slate, Prosocoustic designed acoustic products using the latest materials, design science, and modern-day manufacturing. All Prosocoustic products use recycled material or are recyclable after the manufacturing process. The company's acoustic material of choice is a mixture of slag steel, a by-product of the steel manufacturing process and natural basalt rock and it utilizes framing material products that include wood chip waste and molded via press forms, and favor staples and splining cords over industrial glues.

Prosocoustic recently introduced its new WaveRoom kits, an easy way to bring whole-room acoustic treatments to any environment. A complete ready-to-install solution, WaveRoom kits are first-approach solutions for studios, offices, restaurants, churches, or anywhere better audio clarity is needed. Each WaveRoom kit is designed for effective coverage of up to 100 ft^2, and includes two 2′ × 4′ × 1″ and four 2′ × 2′ × 1″ Wave hybrid panels with all the mounting hardware necessary to hang them on your walls. Additional Booster Kits with 2′ × 4′ or 2′ × 2′ panels can be added for larger spaces. Corner Bass Trap Kits help tame lower frequencies.

Prosocoustic's Wave series products absorb and diffuse sound at the same time. By utilizing a three-dimensional molded frame, sound waves are scattered, increasing the acoustical properties. This innovative solution to acoustic control features the hybrid Wave panel with 1″ absorptive material on the front and diffusion on the back. The acoustically transparent diamond weave fabric harmonizes with any decor and comes in black or stone.

Prosocoustic also offers Gobo Kits for use in recording studios, stages, and live sound applications. Evolution Series aluminum frame panels can be paired with couplers and feet to create portable sound barriers. Additional panels and accessories can be bracketed together or arrayed in various configurations—a versatile way to avoid sound bleed when recording instruments or amplifiers.

Prosocoustic products are distributed worldwide by Full Scale, AV.

prosocoustic.com



This display showcases Prosocoustic Evolution, Waves Hybrid, and custom printed panels.



Prosocoustic's Wave Hybrid panel is made from highly compressed wood fibers and provides both absorption and diffusion.



Prosocoustic's WaveRoom basic kit is designed for 100 to 120 ft^2 spaces, including six Wave hybrid panels with Z-Clip mounting hardware.

Auralex Acoustics

40th Anniversary with Many New Solutions

Located in Indianapolis, IN, Auralex Acoustics is one of the world's most established brands of acoustical treatment solutions, celebrating its 40th anniversary year throughout 2017. Eric Smith, company Founder and President, began the Auralex journey in 1977. "Our staff will tell you that we consistently remind each other to think like the man on the street, which helps us keep our product innovations well-grounded and appropriate for a wide demographic," he states.

Auralex's personalized customer service and free room analysis provides tailor-made solutions. The company is also recognized by its vast and constantly updated catalog of products, sold worldwide through a network of dealers and distributors. Recently, the company announced a series of new acoustic solutions, starting with its new GeoFusor Sound Diffusors, a versatile sound diffusor panel based on the geodesic dome pattern, with a contemporary aesthetic that complements residential or commercial environments. The GeoFusor's dimensional shape offers smooth, even diffusion and enables back filling with absorptive material for enhanced low-end control. The solution addresses standing waves and flutter echo without removing acoustic energy, while enhancing the accuracy of a critical listening environment.

The GeoFusor is wall- and ceiling-mountable and is offered in two sizes (1′ x 1′ and 2′ x 2′). The 2′ × 2′ GeoFusor is sized to fit in ceiling grids and offered in fire-rated and non fire-rated variations. The complementary design of both models permits arraying for acoustical and design purposes. The GeoFusor 22 FR version offers ASTM E48 Class-A fire rating.

Studio Solutions

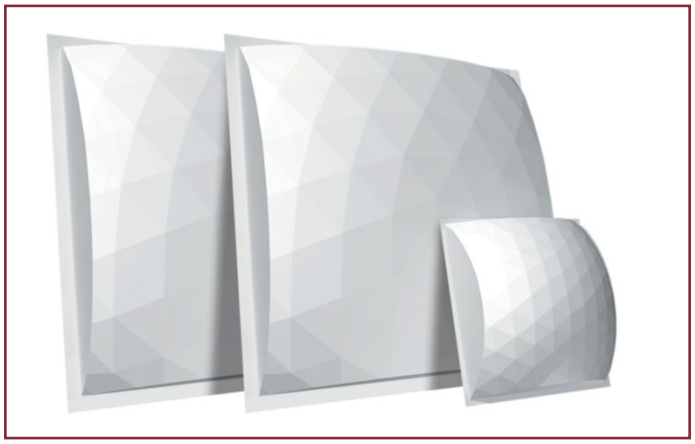
Another new useful product from Auralex Acoustics, is the RoomCaster Freestanding Acoustical Treatment System, a customized broadcast/podcast single-room acoustical treatment solution. Available at a fraction of the price of comparable products, the RoomCaster uses Auralex's acclaimed Studiofoam panels and is a portable, lightweight, and highly effective solution for a variety of spaces and applications. It can be implemented in any location where boundary-mounted acoustical treatments aren't feasible or desired, and it reduces unwanted reverberation and standing waves. The system is ideal for podcasting/broadcasting or small-room recording sessions. It also keeps room noise to a minimum when using a condenser microphone. RoomCaster panels are available in 3″ × 24″ × 48″ and 3″ x 24″ x 24″ sizes and ship in kits of four units (two of each size) with stands. The three available colors, charcoal, burgundy, or purple should fit most environments nicely.

ProKit SonoSuede ProPanel Kits

Finally, as a way to signal its 40th anniversary, Auralex created the SonoSuede editions of its ProPanel ProKit-1 and ProKit-2 kits, featuring the company's popular all-in-one acoustical room treatment systems designed to address the acoustic and aesthetic needs of the small and medium-sized project studio (ProKit-1) as well as the most demanding professional mix/mastering environments (ProKit-2).

These kits are a highly effective first-approach solution for controlling critical mid- and high-band frequencies in studios, consisting of beveled-edge 2′ × 4′ × 2″ ProPanel SonoSuede fabric-wrapped Class A absorptive fiberglass panels. The SonoSuede ProKit-1 and ProKit-2 kits also include 2′ × 4′ × 2″ mitered-edge ProPanel corner treatments that provide low-frequency absorption, which is critical in small- and medium-sized rooms. Based on the 2″ thickness of the ProPanels included in both kits, there is a greater surface area of coverage per panel over typical studio kit offerings. Both SonoSuede ProKits are available in two fabric choices, red or black, and include all mounting hardware.

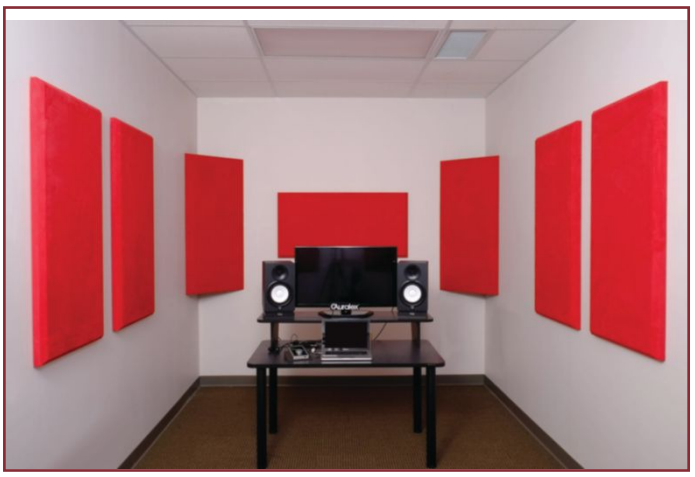
www.auralex.com



Auralex GeoFusor Sound Diffusors offer highly effective diffusion with a contemporary aesthetic.



Auralex RoomCaster setups are shown in burgundy and charcoal.



This is a first-approach example application with a ProPanel ProKit-1 SonoSuede for project studios.

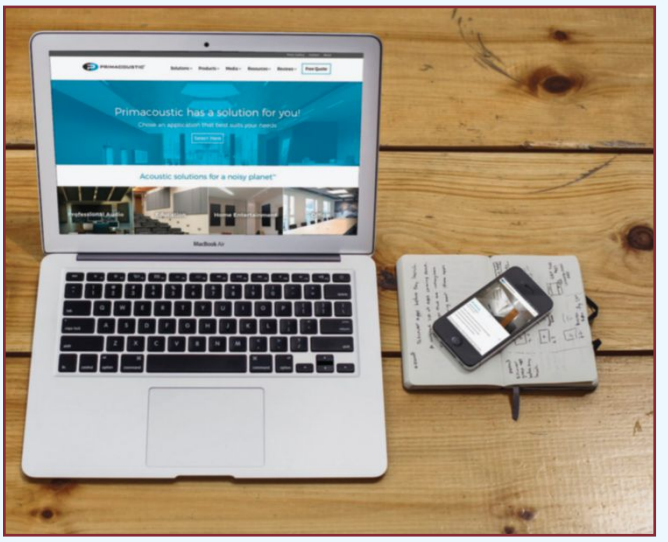
Primacoustic

Expanded First-Approach Acoustic Problem Solving

Throughout 2017, Primacoustic (a division of Radial Engineering, Ltd.) has been constantly restructuring and reinforcing its first-approach market interface. During the year, Primacoustic launched a new website featuring a streamlined user experience that customizes to the individual user's needs, featuring a new blog section announcing the latest product news and featured applications as well an extensive library of educational resources. The site retained its popular online room calculator/quote feature from the previous version.

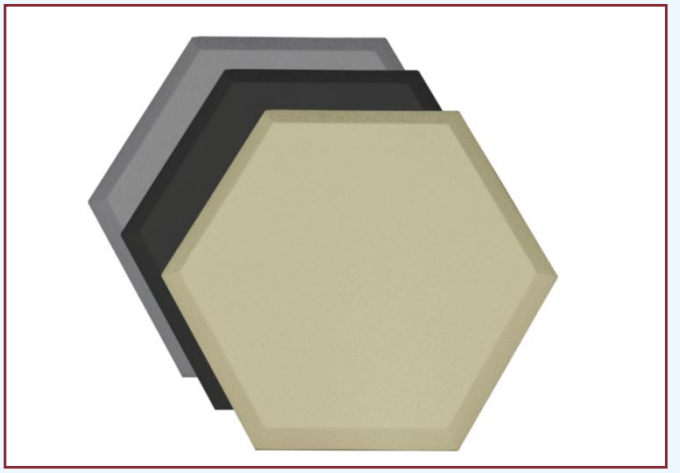
As Primacoustic recognizes: "We know when a potential customer lands on our site, they need to find their application quickly and be walked through potential solution scenarios with as little fuss and distraction as possible. Users also have immediate access to case studies and stories about installations that relate closely to their own situation. As always, a free quote feature allows users to submit their room specifics and measurements to learn more about their solution and the associated cost of materials."

"The idea of treating acoustics can be a daunting situation for people who are unfamiliar with the subject but know they need to address sound issues. This new site not only takes the mystery out of acoustic treatment, it walks the user easily and logically through the information they need," the company adds.



New Honeycomb Broadway Line Panels

Primacoustic Broadway is a range of high-performance fabric-covered acoustic panels designed for use in applications where superior acoustic performance is desired and safety is of the utmost concern. Made from high density 6 lbs. per ft^3 (96 kg/m^3) glass wool, Broadway panels have greater density than typical foam panels for balanced absorption throughout the audio listening range. Primacoustic Broadway panels now include a hexagonal option called the "Honeycomb."



Primacoustic Broadway panels are now available in hexagonal "Honeycomb" shapes.

The Primacoustic Honeycombs are a range of acoustic absorbers that are designed to give users more aesthetic options. Rather than being limited to the typical square and rectangular panels found in most studios, the hexagonal shape and beveled edges enable the panels to be installed in unique patterns and clusters. With these panels, primary reflections and flutter echo can be controlled with an acoustic treatment layout that can be personalized to any space.

The 16" (406 mm) Honeycombs are 1.5" (3.8 cm) thick, and are constructed from 6 lb. high-density glass wool for optimal broadband absorption. As with all Primacoustic panels, the Honeycombs have been tested to achieve stringent Class-A fire ratings, making them safe for use in installations worldwide. The Honeycomb absorbers are available in black, grey, or beige Broadway acoustic fabric, or in Primacoustic's proprietary Paintables white finish. They ship in boxes of 12 panels.

www.primacoustic.com



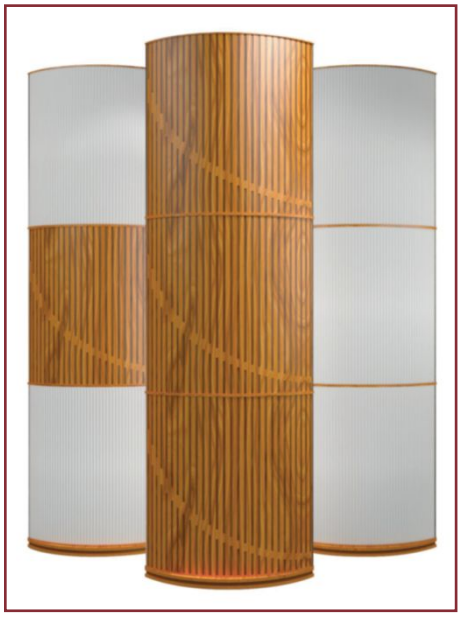
Broadway panels can be hung as easily as pictures using a choice of impalers.

Vicoustic

VicTotem Configurable Acoustic Solution for Hi-Fi and Pro Audio Applications

Acoustic solutions specialist Vicoustic continues to expand its portfolio. Following the recent release of several new acoustics products for building and construction applications, Vicoustic announced the release of its VicTotem configurable acoustic totem, combining absorption, diffusion, and bass traps in a single modular structure. VicTotem combines VicPET Wool, Vicycle Microsuede Fabric, and Wood materials available in a series of modules with multiple finishes and colors.

Vicoustic's catalog of acoustic solutions already includes an extended range of absorption and diffusion panels, insulation products, bass traps, portable acoustic products and acoustic solutions for Music and Broadcast, Hi-Fi and Home Cinema, and Building and Construction applications.



A versatile first-approach acoustic solution, VicTotem is a free-standing panel combining absorption, diffusion, and bass management elements that can be easily arranged and repositioned.

Ten years ago, the Portuguese company introduced its Wavewood flagship acoustic panel, which propelled Vicoustic's international expansion with its mix of design and function. At the core of the panel's success was the combination of acoustic foam and wood creating a series of non-linear sequential cavities, enabling it to act as both an absorber and a diffuser. The solution proved particularly effective in treating medium and high frequencies as well as solving issues such as flutter echoes. Wavewood can also be highly effective in controlling low frequencies when fitted in the corners of a room and used as a bass trap.

The panel's characteristics were instrumental in promoting Vicoustic in the acoustics world, enabling users to treat acoustic problems without destroying a room's ambiance, or over-deadening the sound, while its elegant design enhances the visual appearance of any room, with five different wood finishes available to match a range of studio or listening room interiors. Currently available in different sizes for smaller or larger spaces and with complementary solutions for corners and ceilings, Vicoustic's Wavewood panels are used in control rooms, broadcast studios, recording, rehearsal rooms, and other performance spaces. For music listening rooms, Wavewood panels also became a popular choice given its flexibility and proven effectiveness.

Now, Vicoustic has expanded the concept with a new standard for acoustic treatment, introducing a free-standing concept, which can further improve installation flexibility. The new VicTotem is a free-standing, variable acoustic product combining absorption, diffusion, and bass management and any mix of those principles in the same solution. The design enables the clever combination of stackable two-faced elliptical modules (three modules per box), which can be placed freely in any room, in any combination and can be repositioned and changed for different acoustic environments—a flexible solution for demo rooms and trade shows, with a high "Spouse Acceptance Factor" for living room applications.

For hi-fi and home cinema users, VicTotem is a flexible system that can change with system upgrades and is freely configurable. For commercial and project studios, it offers the most versatile and flexible solution—freely movable between recording rooms and mixing situations.

VicTotem modules are designed with the utmost attention to detail in terms of wood and fire-resistant fabric, combined with Vicoustic's new core material PET wool foam. The product is available in a range of fabric colors and wood finishes to fit any room and space, including a choice of Wenge, Nordic, Light Brown, Cherry, and White woods. Although VicTotem was primarily designed to be free-standing, the company also released solutions for fixed installs in walls and ceilings. VicTotem is sold in sets of three stackable modules for $799 USD and is available now.

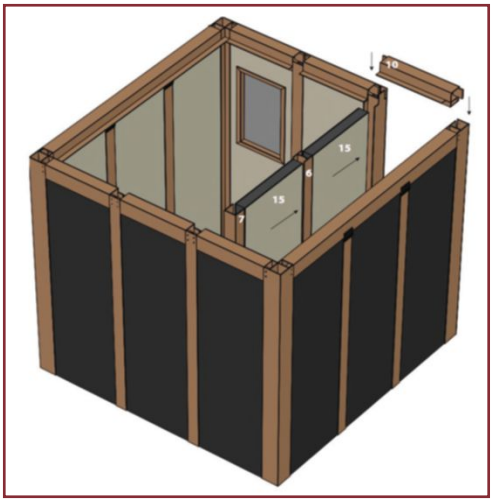
www.vicoustic.com/product/victotem

Audio Plus Services (www.audioplusservices.com) is the new distributor of Vicoustic in North America

Vicoustic VicBooth Premium Acoustic Cabin

Acoustic treatment and insulation specialist, Vicoustic, has launched a new doorless, portable, and affordable acoustic booth with a wood structure and window, ideal for voice-over or instrument practice without bothering neighbors. The new VicBooth Premium enables anyone to minimize sound propagation with a unique, portable, and inexpensive system that can be set up in minutes. The structure is built with premium materials for a more pleasant look and better isolation, using natural wood, fire-retardant fabrics, and acoustic foam.

The solution was designed for recording studios but it can be also adapted to classrooms, to be used as a phone booth in noisy environments such as factories, and for private use at home, complementing home studios, music practice, voice recording, and more. Its adjustable size and functionality fulfill a vast range of demands. VicBooth can also be easily expanded, enabling the creation of a bigger space at any time, with no extension limits. The basic VicBooth structure ships in five boxes, weighing 141 kg, and approximately 3 m^3 in volume.



www.vicoustic.com

Acoustical Solutions Update

Acoustics Vendor and Services Directory

Acoustic Fields
www.acousticfields.com

Acoustics First
www.acousticsfirst.com

Acoustic Systems
www.acousticsystems.com

Acoustic Sciences
www.acousticsciences.com

Acoustical Solutions
www.acousticalsolutions.com

Acoustical Surfaces
www.acousticalsurfaces.com
www.asiproaudio.com
www.proaudioacoustics.com

Artnovion Acoustics
www.artnovion.com

Aural-Aid
www.auralaid.com

Auralex
www.auralex.com

ClearSonic
www.clearsonic.com

Delta H Design
deltahdesign.com

GIK Acoustics
www.gikacoustics.com

Golden Acoustics
www.goldenacoustics.com

G&S Acoustics (Golterman & Sabo)
www.gsacoustics.com

International Cellulose Corp. (ICC)
www.spray-on.com

Jocavi Acoustic Panels
jocavi.net

Listen Audio Systems
www.listen-speakers.com

Media Specialty Resources
www.msr-inc.com

Overtone Acoustics
www.overtoneacoustics.com

Pinta Acoustics
www.pinta-acoustics.com

Primacoustic
www.primacoustic.com

Resolution Acoustics
www.resolutionacoustics.com

Ready Acoustics
www.readyacoustics.com

RealTraps
realtraps.com

RPG
www.rpginc.com

Silent Source
www.silentsource.com

Sonex
www.Sonex-online.com

Vibrant Technology
www.vibetech.com

Vicoustic
www.vicoustic.com



Sound Diffusion: The Movement Toward Omnipresence

By
Jim DeGrandis
(United States)

This article discusses recent developments in diffusion, diffusor design and use, the science around diffusion simulation, and contributions for an upcoming standard.



Photo 1: The NWAA labs test rig has been designed for both direct energy and reflected energy measurements.

Not long ago acoustic diffusers were considered a "dark horse candidate" in the industry. Although little was known about their performance, they had nevertheless been used to great effect in many locations. This placed diffusers in a category that few trusted because of the limited data available on how to use them and what to expect. That has changed. The use of sound diffusers has become more widespread. From houses of worship to school classrooms, and home theaters to board rooms—their contributions are extraordinary.

While the practice of using diffusers has become common and the results are undeniable, the science has remained a bit of a mystery. There is still a bit of that dark magic/voodoo quality to the diffuser, even with its exposure. Luckily, this exposure has spurred every sector of the industry into action, and the diffuser has infiltrated in a way that is unprecedented. While, in the past, diffusion was

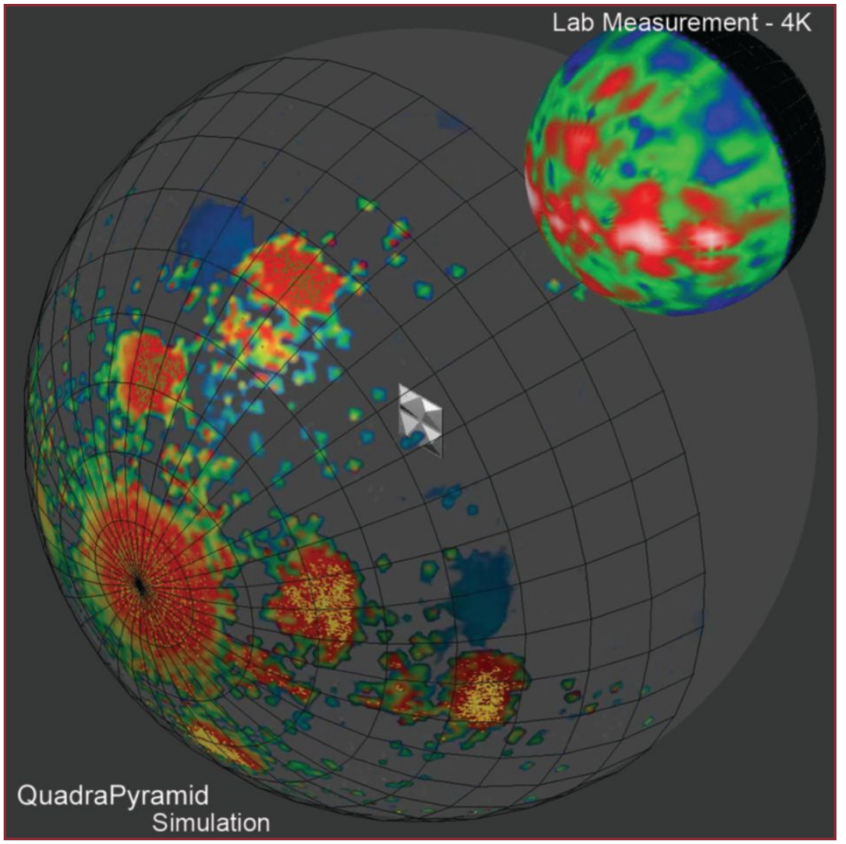


Figure 1: Simulations of diffuser performance are compared to actual lab measurement at 4 kHz (inset upper right).

a small segment with few champions, it has now become a hot topic; researched by scientists, universities, manufacturers, laboratories, acousticians, musicians, audiophiles, architects, interior designers, software developers, and Bob—that guy who wants the best sounding home theater on his block. This ubiquity has begun a mass movement toward resolving those unanswered questions about diffusion, and to better define how the industry can harness this dark horse.

Use of Diffusers

When designing and using a loudspeaker system, acousticians have long since referred to specifications that show polar responses of those speakers. Tests (e.g., the AES56) have shown how the energy radiated from a loudspeaker can be measured and defined, enabling the accurate prediction of their acoustic performance. This advancement catapulted the development and effective use of these systems. Taking a cue from this, the industry has reevaluated how we all look at diffusers and their performance—think of a diffuser like a speaker.

Diffusers, like loudspeakers, radiate sound. The difference is that diffusers are excited externally. Facilities (e.g., NWAA Labs, Elma, WA) have repurposed their loudspeaker test rigs to also measure reflected energy (see **Photo 1**). By measuring the radiation polar patterns of a diffuser, we can predict their performance. This helps define their use and enables the accelerated development of newer diffuser designs that perform in ways never before imagined. However, measurement is only one chapter of this story.

If manufacturers had to measure every design variation in their research and development cycle, it would slow down the process, as every design would have to be physically created, sent to the lab, measured, and analyzed. After another experimental modification is attempted, the entire process would need to be repeated. Enter computational physics simulations (see **Figure 1**). By knowing how to measure a diffuser's performance in the real world, we can virtually model that measurement and analyze new geometries in minutes—instead of days or weeks. This allows for a streamlining and virtualization of the development process, refining a device's geometry virtually, before committing to a design to be physically prototyped, measured, and evaluated (see **Figure 2**).

This isn't to say that good old experimentation has been thrown to the wayside. The original, early

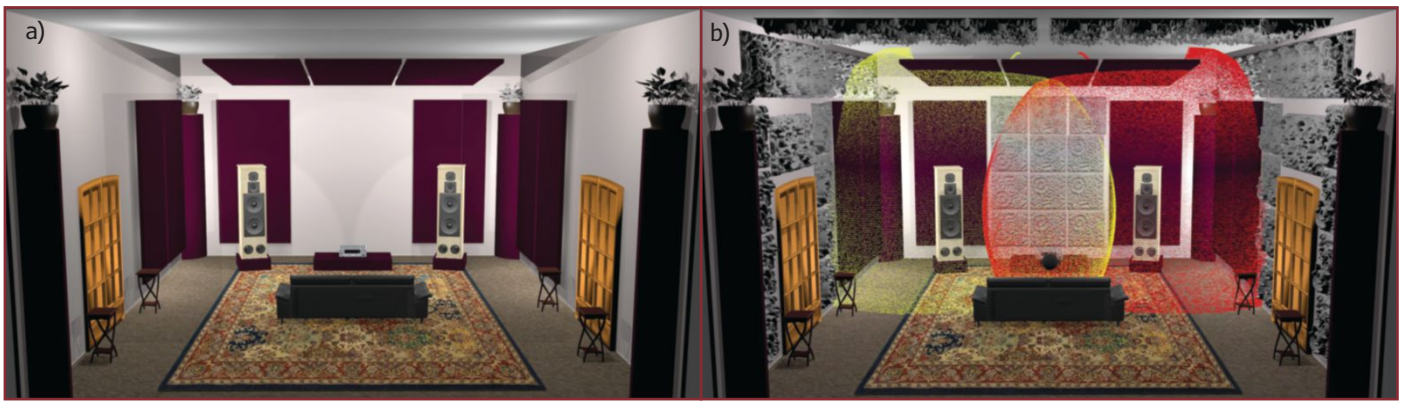


Figure 2: A room model for acoustic performance simulation (a). Same room model with diffusers added, simulation in progress (b).

adopters of diffusers still like to get their hands dirty and measure first with their ears. They know that the final measure of any acoustic implementation is subjective, and therefore, subject themselves to hours of "guinea pigging" to determine if an idea works for them. This author has personally gone into a room with some noise makers, sound meters, and a binaural head, and then started filling the room with diffusers—testing and documenting its performance at intervals along the way (how it sounds, how it affects intensity, and reverb times, etc.) Some experts have gambled, experimenting in live environments to solve problems with diffusers that others wouldn't attempt.

Effectiveness of Diffusers

In a recent conversation with acoustical consultant Chips Davis, we were discussing the effectiveness of bouncing reflections off a diffuser before they reached the listener. We both had made use of this method for different reasons. I had recommended bouncing ceiling-firing Dolby Atmos speakers off diffusers to widen the reflection pattern in rooms with low ceilings while Davis had fired stage monitors up at diffusers to improve monitor coverage on a theatrical stage. These are both novel uses for diffusers which solved problems in a new way—unless you consider the JBL Paragon, which reflected the direct speaker signal off a curved enclosure, producing a wide acoustic image (uncanny, considering it was initially produced in 1957).

With the advent of new technologies, testing methods, advancing developments, simulations and old-fashioned elbow grease, diffusers have finally cemented their place in the palette of tools used for fine tuning an acoustic system. As we learn more about how these dark horses run, we find new ways to widen the sweet spot, improve imaging, remove destructive interference, and harness that diffusion magic to work for us.

About the Author

Jim DeGrandis is a member of the Acoustical Society of America (ASA), ASTM International, and is the Chief Science Officer at Acoustics First Corp. He has been working with the ASTM in developing a useful sound diffusion standard and separately on simulations that predict the lab test results.





ACOUSTIC FOCUS

Early Reflections



This article explains what "early reflections" are and why they must be avoided if your goal is accurate sound reproduction.

By
Ethan Winer
(United States)

For recording and mixing engineers, accuracy is especially important to make "portable" mixes that sound as intended through other systems. If music or dialog is obscured by reflections and other room anomalies, mixes that sound good now might not sound good later, or in the car, or on your friend's stereo. I'll also debunk some related acoustic myths along the way.

But first, understand that all acoustic problems are caused by reflections off the walls, the floor, and the ceiling. At bass frequencies, below around 300 Hz, bass traps are needed to absorb reflections that cause peaks, nulls, and modal ringing. At higher frequencies broadband absorption is used to tame excess reverb, ambiance, and echoes. Early reflections are a specific type of echo—those reaching your ears within about 20 ms of the direct sound from the speakers—and they affect mainly mid and high frequencies.

Early reflections reduce clarity and confuse stereo imaging because sound arrives at your ears at different times and from different directions. You hear the direct sound from the loudspeakers first, then the reflected sound arrives slightly later after bouncing off one or more nearby surfaces. When an echo arrives within about 20 ms, the ear doesn't identify the echo as a separate sound source so the two sounds are merged into one. But when the sounds come from two different locations—the loudspeaker and a side wall—it's difficult to tell where in the stereo field the source is located. You can hear when an instrument or voice is panned fully left or right, but in-between positions are more poorly defined. Worse, in an untreated room reflections often arrive from more than one nearby surface, which further confounds imaging.

Comb Filtering

When direct sound from a loudspeaker (or any other source) combines in the air with early reflections, a specific type of skewed frequency response known as comb filtering occurs. Depending on the difference in arrival times, some frequencies are boosted (combined in phase) while others are cancelled (out of phase).

Figure 1 shows the comb filtered response caused by a delay of 1 ms., which is equal to about 1′ of difference between path lengths. This is typical with near-field listening when sound bounces off your desk. A delay of 1 ms gives 180° of phase shift at 500 Hz and 360° at 1 kHz, creating a null at 500 Hz followed by a peak at 1 kHz. Subsequent peaks and nulls occur at 500 Hz intervals corresponding to phase shifts of 540°, 720°, 900°, and so forth. Because the peak and null frequencies are related to the difference in arrival time, which depends on distance, the response often isn't even the same at both of your ears. Worse, the

peak and null frequencies change with very small head movements, which further harms localization.

The solution for all of these problems is to create a Reflection Free Zone (RFZ) at the listening position by putting absorber panels at specific places called reflection points on the side walls and the ceiling. It's also useful to have thick carpet over any bare floor area between you and the loudspeakers. Listening to music free of reflections is similar to listening through headphones—musical instruments and voices sound much clearer, and their left-right placement is better defined. Once all early reflections are removed, depth and soundstage magically come to life. The stereo field is also wider and even seems to extend slightly beyond the physical left-right speaker boundaries. For mixing engineers, all of a sudden you can easily hear very small changes in left-right panning, EQ, and applied reverb. Even at my age (68), I can easily hear EQ and volume changes of 0.5 dB at midrange frequencies in both of my well-treated listening rooms.

Note that the term reflection "point," while common, is a bit of a misnomer because it's really an area. Sound radiates outward from a loudspeaker source in an arc, so once it reaches the side wall or ceiling more surface is affected and the entire area should be treated. Therefore, a reflection point more accurately identifies the center of an acoustic panel you'll put there. The goal is for early reflections to be at least 15 dB softer than the direct sound from the speaker at all frequencies, and 20 dB or more is better still. When a reflection is 20 dB softer than the direct sound, the comb filtering peaks and nulls are both less than 1 dB. Software such as the fabulous (and free) Room EQ Wizard can identify individual reflections and show their relative strength.

Where to Place Absorber Panels

Figure 2 shows the main paths that sound takes from a loudspeaker to arrive at your ears:

- Direct sound from the speakers
- Early reflections from the left and right side walls
- Additional reflections from the rear wall behind you, which might or might not be early.

Sound travels at approximately 1′ per millisecond, so reflections from a wall closer than about 10′ behind your head are considered early because the round-trip distance is less than 20′ = 20 ms. The narrow rectangles at both side-wall reflection points shown in Figure 2 are absorbing panels centered vertically at ear height.

The side-wall reflection points are slightly forward of halfway between you and the front of

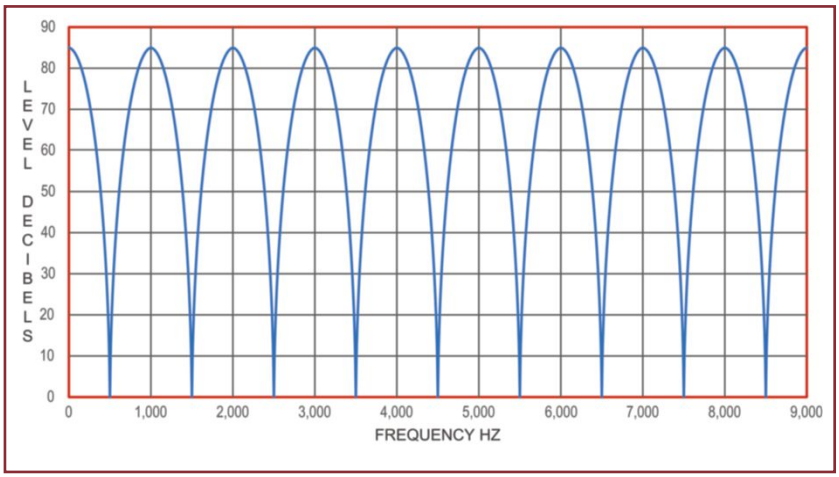


Figure 1: Comb filtering is characterized by a repeating pattern of equally spaced peaks and deep nulls. The response shape resembles a hair comb, hence the name. Note that this graph uses a linear frequency scale rather than the more usual logarithmic spacing. Comb filter peaks and nulls occur at equal frequency multiples, rather than by octaves as with musical overtones or harmonic distortion products.

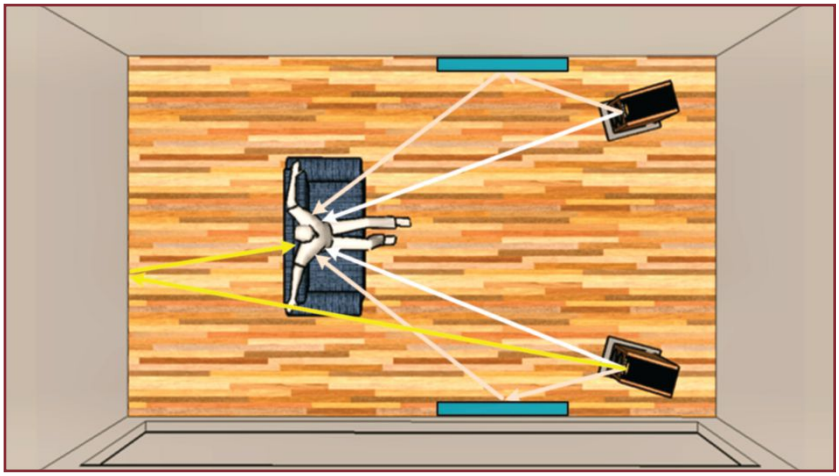


Figure 2: Early reflections arriving from the side walls damage clarity and imaging. Reflections from the rear wall are also considered early if that wall is closer than about 10′ behind you.

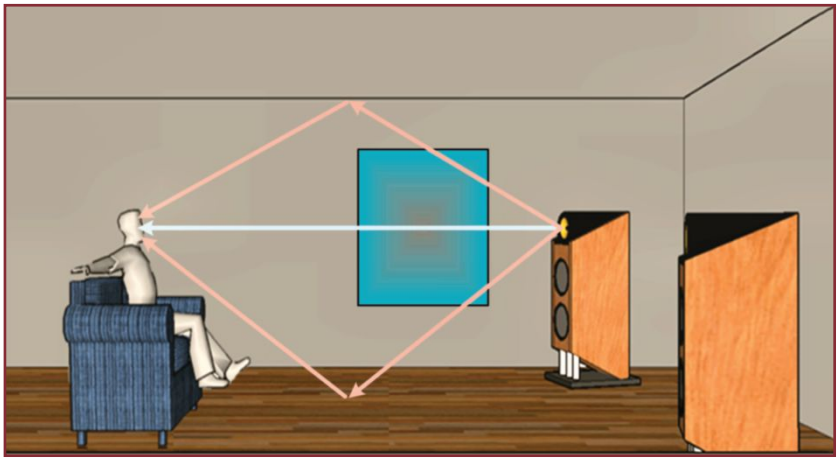


Figure 3: Additional early reflections arrive from the floor and ceiling, so for best results those surfaces should also be treated. The center of these reflection areas is halfway between you and the loudspeakers.



Photo 1: My trusty assistant Mrs. Emma Gray demonstrates this thick seat cover that absorbs damaging early reflections from a high chair back.

the speakers—how much forward depends on the room's width and the distance between your speakers and the adjacent side walls. But the ceiling and floor reflection points are exactly halfway between you and the speakers as shown in **Figure 3**. This presumes that the tweeters are at ear height, which they should be. The left and right side acoustic panels should be symmetrically placed. This also assumes your speakers are centered left-right in the room, which is required for good stereo balance. With the help of a hand mirror you can identity the exact location for each panel. While you sit in your usual place, have a friend hold the mirror flat against the wall and slide it around. When you can see the tweeter in the mirror, the mirror is at the center of the area to be treated.

In hi-fi and home-theater forums, when someone asks if they'll benefit from absorbing early reflections, I always suggest they hang folded-over bath towels on the side walls with masking tape to see if they hear an improvement. Towels aren't as good as real acoustic panels that absorb more and over a wider range of frequencies. But it's easy and free to try, and then you can get proper panels if you like what you hear. Related, a reflective couch or seat back directly behind your ears also causes comb filtering. I had a local sewing shop make the thick seat cover shown in **Photo 1** from two layers of 0.5" thick cotton batting, covered with plush fabric I bought at a local fabric store.

The range of frequencies absorbed affects the quality of a RFZ. Early reflections are mainly about clarity at mid and high frequencies, so you can often get away with relatively thin panels, maybe 1" or 2" thick. However, having thicker panels extends the absorption to even lower frequencies, which is always a good idea because it puts more bass trapping in the room. Thick absorbers are especially useful on the rear wall because those peak and null frequencies extend all the way down to the bass range. Corners are considered the most important place for bass traps—which is true!—but the entire rear wall is just as important even if the reflections aren't early.

Another way to avoid early reflections is by angling the side walls outward and sloping the ceiling upward, toward the rear of the room. With an angle of 35° or more, reflections go around or over the listening area without having to apply absorption. In a large professional control room, this lets you retain more room ambiance because you don't need additional absorption just to get rid of the reflections. But most people don't have the luxury of building walls with such extreme angles, so their only option is to apply absorption at the reflection points. (And angles don't avoid reflections from the rear wall either.) Of course, absorption works perfectly well, and rooms the size you'll find in most homes sound better with less ambiance anyway. Indeed, I believe the notion "I don't want a dead sounding room" is mostly misguided. Which leads us to our next topic.

Early Reflection Myths

Many times, I've read that early reflections in a home-size studio or listening room enhance spaciousness and should not be treated. By "home-sized" I mean a room less than about 25' wide, where reflections from nearby surfaces arrive soon after the direct sound from the speakers. The argument is that adding absorption robs the room of desirable ambiance needed for realism. On the surface, this might seem plausible but it's wrong. The ambiance and reverb already present in most recordings is typically that of a large venue—a concert hall, a movie scoring sound stage, or a similar size space created with artificial reverb. But, when played back through loudspeakers in a small untreated room, strong early reflections drown out the larger sounding reverb in the recording. This makes music sound smaller and narrower, not larger and wider.

About the Author

Ethan Winer has been an audio engineer and professional musician for more than 45 years. He's currently a principal at RealTraps where he designs acoustic treatment products for recording studios and listening rooms. Ethan's "Cello Rondo" music video has received nearly two million views on YouTube and other websites, and his book *The Audio Expert* published by Focal Press is available at amazon.com and his own website ethanwiner.com.

ISOACOUSTICS
Sound quality by design

GAIA

ISOLATION FOR SPEAKERS & SUBWOOFERS

The most astounding improvement For your hi-fi system



GAIA I
SPEAKER WEIGHT
MAX: 220LBS (100KG)

GAIA II
SPEAKER WEIGHT
MAX: 120LBS (54KG)

GAIA III
SPEAKER WEIGHT
MAX: 70LBS (32KG)

isoacoustics.com

Product Design

GAIA isolation are precision machined to provide a high degree of speaker isolation. The IsoAcoustics patented design resists lateral movement and oscillation to maintain alignment with the listening position and provide greater clarity and focus. Internal reflections are attenuated resulting in a sound stage that blossoms open.

"…the biggest improvement in sound I have ever heard on the system. Better than changing cables or electronics, the noise floor was lower, the focus was better, and the bass went deeper."
Harry Weisfeld, Founder, VPI Industries

"…imaging is even more amazingly precise. Truly incredible!"
Nick Squire, Chief Sound Engineer, Boston Symphony Orchestra

"… I expect to see more of these packages become a "required option" for larger floor mount monitors."
John Storyk, Founding Partner of Walters-Storyk Design Group

Another common myth is that rooms mainly used for playing stereo music should be treated differently, and less, than home-theater rooms. But a big part of most movie sound tracks is music! Whether listening to music alone or as part of a movie, music and dialog should both come through clearly, with a flat frequency response and low distortion, and without strong early reflections clouding the sound. The only difference between surround sound and stereo is the center channel and surround speakers. Stereo creates a virtual center channel, so for someone sitting in the middle there's no real difference between having a center speaker or not. The only change I make when listening to stereo music vs. movies is to switch the playback mode in my receiver.

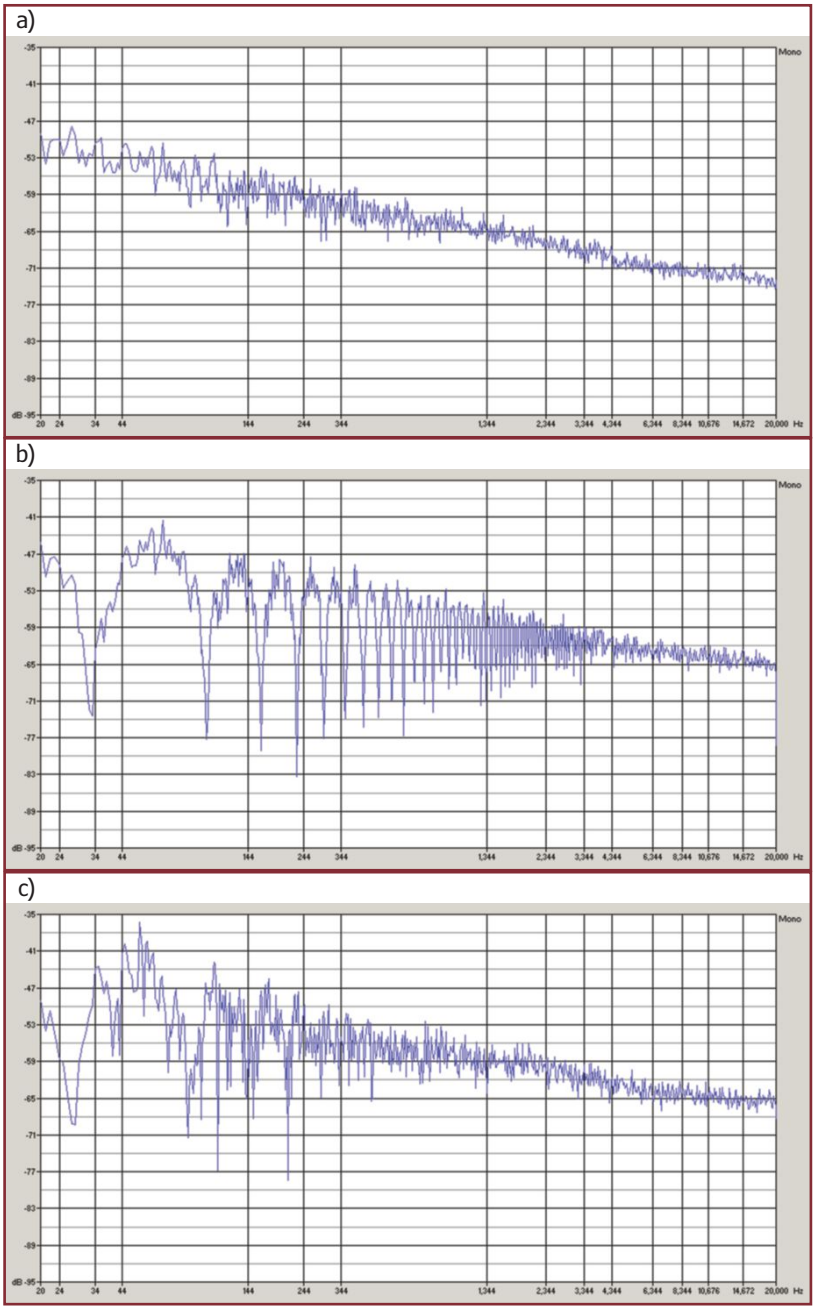


Figure 4: This is the spectrum of a pink noise source (a), the same source combined with a simulated reflection (b), and the same source with the simulated reflection low-pass filtered before being combined (c).

Yet another myth is that damaging reflections arrive from the front wall. It's common to see people put acoustic treatment on the front wall behind the speakers, though that's pretty much the last place that benefits! The front wall doesn't generate much reflected energy unless you have dipole speakers that send sound equally out the front and rear. Normal "box" type speakers send most of their sound forward, though most speakers become omnidirectional at low frequencies starting around 300 Hz. So bass trapping on the front wall is always useful, just as it is anywhere else. But the thin foam we often see has little effect. I have no absorption on the front wall of my living room home theater other than bass traps straddling the wall-floor corner.

Some people claim that diffusion is better than absorption at reflection points, but I disagree. When I tried that in my living room home theater, my wife and I both thought the sound of diffusers was hardly better than bare walls. Perhaps diffusers would be useful in a wide room where side-wall reflections arrive later than 20 ms. But even in large professional control rooms, diffusers are used mainly on the rear wall. So to my ears, reflection points are better treated with absorption than with diffusion, and that leads to our last myth:

There's a common belief that early reflections are less damaging if they're "neutral," coming from loudspeakers that have a flat off-axis response. This is easy to disprove using logic and science—when reflections from a wall have the same frequency content as the direct sound, the comb filtering is most severe. Peaks can be up to 6 dB, and nulls can be 30 or even 40 dB deep at higher frequencies.

If the loudspeakers have a limited off-axis response such that side-wall reflections contain less high frequency energy, the comb filtering in the important upper "clarity" range will be less severe. Therefore, the more closely the original and delayed sounds match in spectrum and volume, the more severe the comb filtering will be. This is basic math, and a rolled-off response at higher frequencies is identical to having high frequency absorption on the walls. So while a flat off-axis response is an important goal for most loudspeakers, it actually harms audio quality when early reflections are allowed. In other words, if you have good speakers, it's even more important to absorb reflections from the side walls, not less important as some people claim.

To show why this happens I created a set of test

files and plotted the responses. The three graphs in **Figure 4** show the pink noise source (a), the same noise mixed with a 15 ms delay to simulate a loudspeaker having a flat off-axis response (b), and the noise mixed with a delay whose high frequencies were rolled off at 6 dB per octave (c). You can clearly see that the comb filtering is less severe when the simulated reflections have less high frequency content. When the source and reflections don't match exactly, the peaks and nulls are both less pronounced.

If you're curious, you can calculate peak and null amounts based on reflection strength using the Decibels.xls spreadsheet found on my website (http://ethanwiner.com/decibels.xls) or in the Supplementary material section found on the audioXpress website (www.audioXpress.com). I created this spreadsheet as a companion for my *Audio Expert* book, but I'm glad to share it publicly so readers can experiment for themselves.

Admittedly, a well-treated room can be an acquired taste for some people. But in my experience, once you get used to the sound and learn to appreciate the improvement in clarity, there's no going back. The goal for all listening rooms is a neutral sound—when you listen in a room that adds no character of its own, you're closest to hearing the true tonality of the music. Indeed, in a well-treated room you'll hear more spaciousness and ambiance in your recordings, not less. The sound will be richer, fuller, and much larger overall. As proof, headphones are fully anechoic having no reflections at all, yet music generally sounds larger on headphones than through loudspeakers, and you can hear musical details more clearly.

Listening in a neutral sounding room also lets you better appreciate the intent of the artist. The music you love to hear was mixed, mastered, and approved by people listening in rooms free of early reflections. So you should too, if your goal is to enjoy the same experience.

Project Files

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

Resources

Room EQ Wizard, REW, www.roomeqwizard.com.

E. Winer, *The Audio Expert: Everything You Need to Know About Audio*, Focal Press, 2012.





ACOUSTIC FOCUS

New Research on Low-Frequency Absorption Using Membranes

This article addresses the problem of destructive room modes (also referred to as eigentones, resonances, or standing waves) in critical-listening and sound recording environments, and the difficulties associated with testing low-frequency absorbers, which are most often used to mitigate room modes.



Photo 1: These are membrane views of the Curve Diffusor (top) and two CornerSorbers (corner configuration, at left).

By
John Calder,
Acoustic Geometry
(United States)

Recent research into the effectiveness of membrane-based low-frequency absorbers has revealed partially-unexpected results of the lab tests with a new low-frequency absorber design, inspiring corresponding applications toward mitigating room modes (see **Photo 1**).

The Room Problem

Room modes consist of nodes (sound energy cancellations) and anti-nodes (sound energy additions) that result from sound excitation in a room by frequencies at wavelengths that match one or more of the room's primary (axial) dimensions (length, width, and height). Modes result in resonances at quarter-wavelengths and nulls at half-wavelengths along each room axis throughout the room, as well as at multiples of the fundamental mode frequencies. Though room modes exist throughout the room at all audible frequencies (and all modes are found in all trihedral corners), they are strongest in the 20 Hz to 200 Hz range in small rooms, usually defined as under 5,000 ft^3 in volume.

The destructive extent of modes is also dependent on listener/microphone and sound-source positions in the room. Modes are one of the greatest concerns when considering accurate sound recording and reproduction. As a distortion of the original sound, modes change the perceived tonal characteristics of the sound (e.g., much more apparent bass or much less apparent bass, depending on position), and cause longer reverberation times at resonant frequencies, which is perceived as a "smearing" of the sound. This results in a loss of sonic detail and impairs the perception of sound-source localization (imaging).

All room anti-node (energy-addition) pressure is at maximum at hard-wall and corner surfaces, which can be exploited with proper placement of bass absorbers. No acoustic absorber can lessen a node (energy cancellation) directly, because energy cannot be subtracted from zero energy. However, when an anti-node (addition of energy) is diminished, its frequency-corresponding node (cancellation of energy) is equally diminished (i.e., as you decrease the strength of an addition, the strength of the cancellation equally decreases). We need only address the anti-node, or resonance areas, to diminish the destructiveness of both.

Low-frequency absorbers—often called "bass traps" or "bass absorbers"—are used to mitigate modes in professional and consumer audio rooms, and are also used for industrial and commercial workplace low-frequency noise reduction (see **Figure 1**). Low-frequency absorbers are typically rated in effectiveness for frequencies under 200 Hz, which is the approximate frequency below which most small rooms become pressure-responsive (the room crossover), rather than velocity-responsive, because one or more of the dimensions of the room are shorter than the longest wavelengths to be propagated in the room.

The Product-Testing Problem

Sound absorbers are used to reduce sound energy (echo and reverberation) in rooms. To ensure the effectiveness of absorber products, they are tested in laboratories using reverberant (diffuse) chambers per ASTM International Standard C423-09a. An excerpt from Section 1.1 reads: "This test method covers the measurement of sound absorption in a reverberation room by measuring decay rate." This standard requires reporting on the absorption amount, in sabins and Absorption Coefficient (a derivative of sabins and absorber surface area), from 125 Hz to 4 kHz, and stipulates that test results below and above those frequency requirements may also be reported when measured per C423-09a.

As a side note, a "sabin" is defined as a unit of sound absorption. Sabins could be calculated with either imperial or metric units. In theory, 1 ft^2 of open window is one imperial sabin; 1 m^2 of open window is 1 m sabin. The unit is named in honor of Wallace Clement Sabine.

Nearly all independent testing laboratories have reverberant chambers of under 300 m^3 in volume, which is large enough to measure accurate results above 160 Hz but not below. This is a problem when measuring the effectiveness of products designed to absorb below 160 Hz. While test methodologies exist for in-situ testing (measuring in non-calibrated non-test-lab room situations), they are not repeatable elsewhere, and are therefore not useful for accurate effectiveness comparison purposes.

Fortunately, for product designers—and consumers—there is one test lab that is large enough, at 738 m^3, to be accurate for measuring absorption down to 40 Hz—NWAA Labs, in Elma, WA. This facility, containing the two largest independent reverberant test chambers in the world (situated in a never-online former nuclear power plant), is run by former NASA scientist Ron Sauro, who has authored many professional studies and papers on acoustical testing and research.

Sound-absorber products fall into four broad categories: the most common is frictional absorbers, which are primarily fiber- or foam-based (including glass, cotton, mineral, etc.), diaphragmatic absorbers, which are primarily membrane-based (including mass-loaded vinyl and other thin, flexible materials), resonator absorbers (e.g., Helmholtz chambers), and a newly-discovered category, diffractive absorbers. All four absorber categories reduce reflected sound energy by turning a portion of it into heat energy.

Frictional absorbers (fiber and foam) act on sound velocity in air, which is approximately 1130 feet-per-second (fps), using the friction caused by the large available surface area ratios (relative to product working surface area, up to approx. 50-to-1). Diaphragmatic (membrane) absorbers act on sound pressure, which is maximum at wall, ceiling, and floor surfaces, especially in corners. Sound has a velocity at hard reflective surfaces of zero feet-per-second, and this is the primary reason fiber- and foam-based absorbers work less effectively than properly-designed membrane absorbers to reduce low-frequency energy under 200 Hz.

Therefore, a large amount of fiber or foam—both in volume and depth—is needed to absorb low-frequency

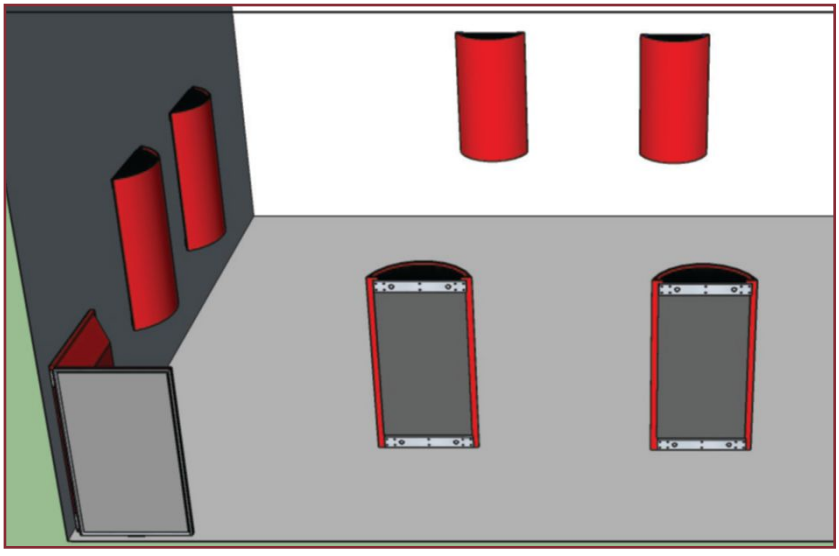


Figure 1: Computer simulation of low-frequency absorbers in room

energy, on the order of many cubic meters. However, for membrane absorbers, the differing impedances on either side of an enclosed membrane (free air vs. enclosed-volume air) enable it to react in response to low-frequency energy by displacing. This displacement reduces some of the sound energy at lower frequencies and results in more efficient energy absorption—by

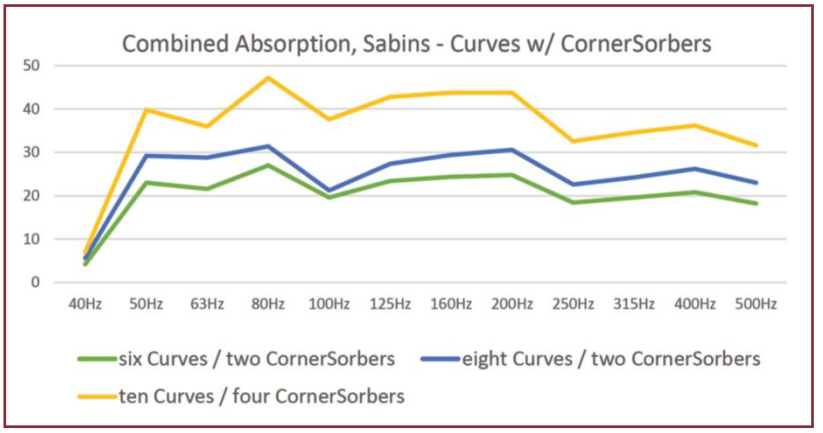


Figure 2: These absorption charts are in sabins, which are the actual measured units of sound absorption (as opposed to less-reliable Absorption Coefficients) with suggested combinations of Curves and CornerSorbers, and separate Curve and CornerSorber test results for the units shown in the combination chart. The Combination chart utilizes single-unit sabin amounts multiplied by number of units in each suggested example. For Figures 2–4, the title designations—1" for Curves and 3" for CornerSorbers—refer to the distance in inches of the membranes from the wall surfaces for that test.

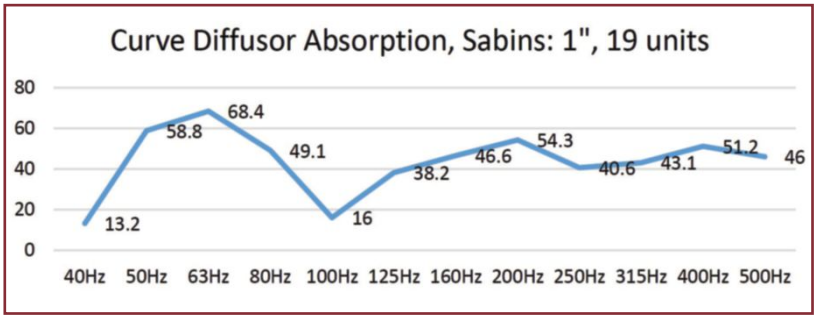


Figure 3: This Curve Diffusor chart shows the absorption in sabins for the units as tested.

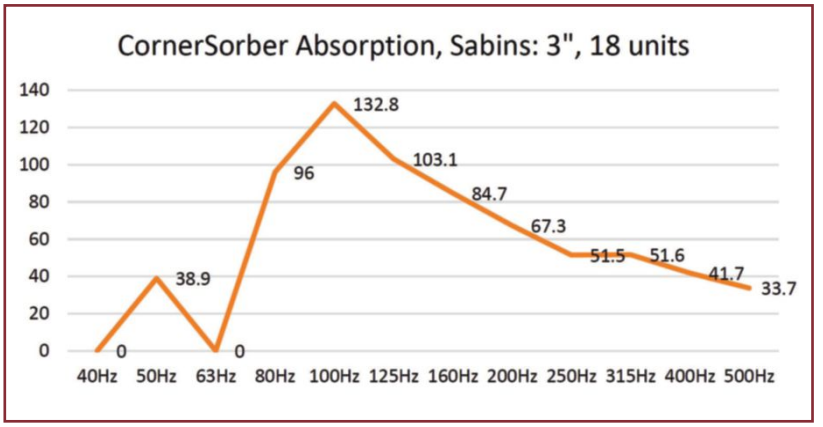


Figure 4: This CornerSorber chart shows absorption in sabins for the units as tested.

frequency and volume of materials used (and room space taken up)—than is possible with fiber- or foam-based absorbers.

In addition to the size of the test lab and the category of absorber, another consideration in laboratory testing of low-frequency absorbers is their intended-use mounting or deployment configuration. While many absorber products can be tested per ASTM E-795 standard mounting practices (e.g., A-Mount: products mounted flat on a wall or floor surface), products with less-frequently utilized test mounts are also included in the E-795 standard (e.g., J-Mount). This enables absorbers—especially low-frequency units, for which test results are more susceptible to room placement variations—to be tested where they are designed to be used in real-room situations while still inside the calibrated test laboratory. This is a more accurate measurement method, leading to more relevant conclusions from the test results, especially for end-users.

Test Method and Results

Based on the results of earlier testing done on the mass-loaded vinyl (MLV) membrane absorbers built into the back of each of Acoustic Geometry's Curve Diffusor, the CornerSorber was designed as a dedicated low-frequency absorber for different frequency bands to compliment the Curve Diffusor's low-frequency absorption capabilities (see **Figures 2–4** and **Photo 2**).

The design goal was to use different membrane densities to offer different effective frequency ranges for the same low-frequency absorber enclosure, making them easily changeable in manufacturing to target desired room mode ranges. The design assumption was that, based on a theory of membrane density-dependent resonances, the various membrane/enclosure combinations would yield different center frequencies and ranges of frequencies absorbed.

To test the effects of different densities of membranes, 18 beveled enclosures were built with slide-in membrane-frame retainers (see **Photo 3a**), which accepted MLV carrier frames holding different sets of densities, with 18 frames per density. The three densities tested were 1 lb./PSF, 0.5 lb./PSF, and 0.25 lb./PSF, contained in frames that were simply "swapped-out" prior to each test run.

The number of test enclosures (18) was dictated by the minimum surface area (120 ft^2 or more) needed to obtain accurate results in NWAA's large main chamber. The design goal of the three-bevel-sided enclosures was to place a pair of membranes at a 90° angle as closely as possible to—and in parallel with—the corner surfaces, either vertically or horizontally (see **Photo 3b**). The volume of each enclosure and its internal damping (cotton-fiber absorbent material) did not change, and

therefore changing the membrane density, based on resonance theory assumptions, should have yielded different results for each of the tests we ran. It did not. Another assumption, also based on actual data from the Curve Diffusor tests, was that the most effective distance from the corner wall surfaces for the new product's membranes would be the same as for the Curves: 1" from the wall surface, parallel to the wall. And again, this assumption was not correct for this design.



Photo 2: Here the Curve Diffusers (a) are being put into place (b) for the low-frequency absorption test at NWAA Labs on August 18 2015.

When the test results were examined and compared (a total of 10 full tests were run), many of the outcomes were different from predicted results, which were based on our understanding of various resonance theories and subsequent logical assumptions, in these areas:

- The absorption ranges and center frequencies, as well as the effectiveness ("Q") of the low-frequency absorbers, did not change significantly with changes in membrane densities (excepting the 1# density membranes, which showed less efficiency at the center frequency)

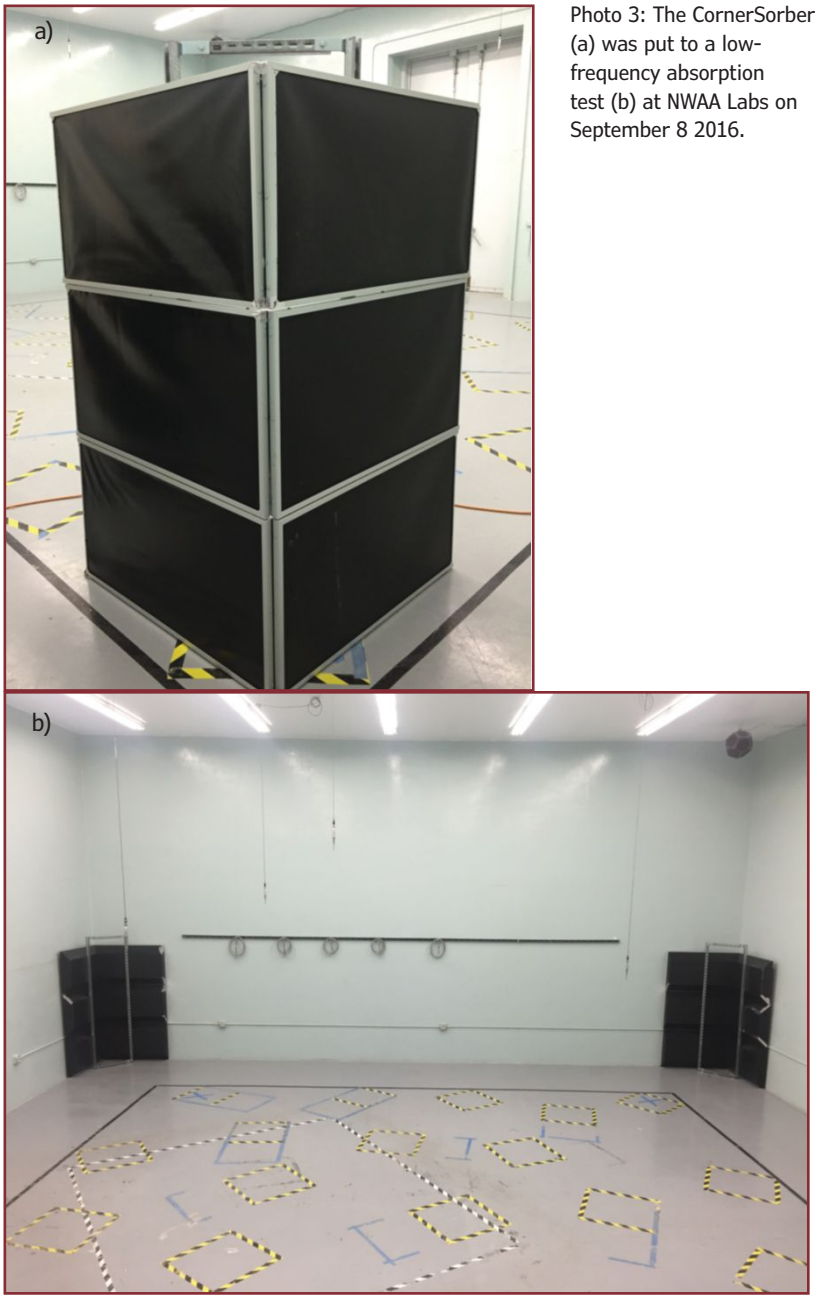


Photo 3: The CornerSorber (a) was put to a low-frequency absorption test (b) at NWAA Labs on September 8 2016.

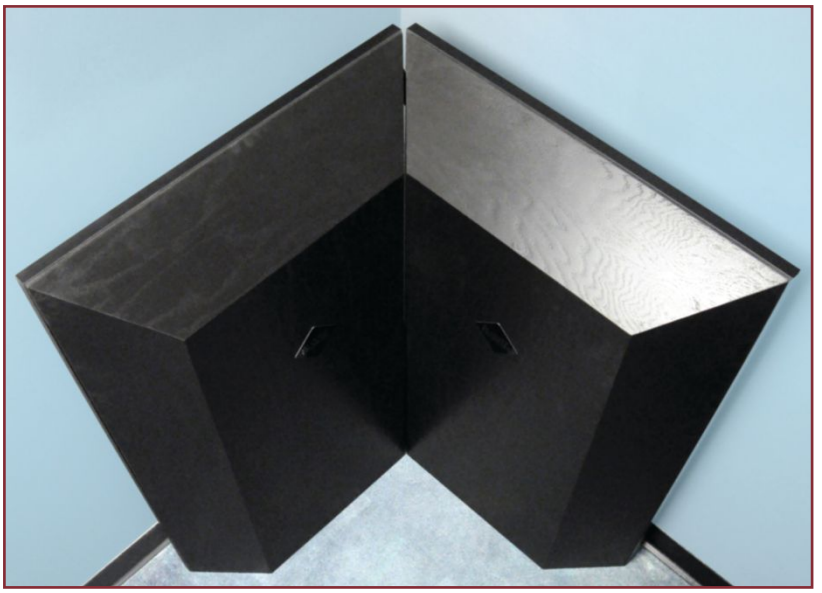


Photo 4: Here are two prototype CornerSorber panels in use. The test results prove that this close-corner location makes the CornerSorber a very efficient and effective low-frequency absorber and can realistically reduce destructive low-frequency room modes.

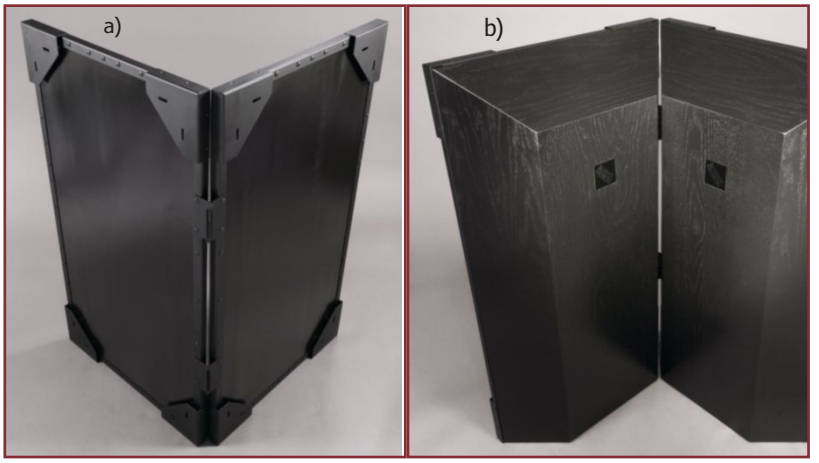


Photo 5: Acoustic Geometry's CornerSorber is a corner-placement bass absorber (see front (a) and back (b), utilizing a Mass-Loaded Vinyl (MLV) absorption membrane in a cutting-edge design that mitigates low-frequency room modes for the most accurate sound reproduction in a room. Beveled on two sides and the top, a pair of CornerSorbers can be set up three inches from, and parallel with, each corner wall, either vertically or horizontally, for the most efficient and effective reduction of low-frequency room-mode pressure waves.

About the Author

John Calder has worked as a musician, recording engineer and producer, concert sound mixer, sound designer, audio post-production mixer, studio owner, theater performer, marketer, product developer, and instructor. He has more than four decades of professional sound experience, engineering records in Minneapolis, Los Angeles, Milan, and Montreal, earning Platinum and Gold Records, as well as Gold Reel, VPA-Monitor, Muse, and AZA Awards. He is currently Director of Retail for Acoustic Geometry in Minnesota. Credits include Neil Diamond, The Band, Joe Cocker, Umberto Balsamo, Callaway Golf, DuPont, BBDO, Fallon, CM Advertising, Cornell University, MN Zoo, MN History Center, and MN Science Museum.

- The most effective distance from the wall for the membranes of the new design (CornerSorbers set 3" from the wall surfaces in parallel) was not the same as for the earlier-tested design (Curve Diffusors mounted 1" from the wall surface in parallel)
- The absorption profiles of the new designs in "semi-free space" (more than 1 m from any wall) was approximately an octave higher in range and center frequency than when in the "pressure zone" or close-coupled in parallel with the wall surface
- When the enclosures were reversed in membrane orientation—with their backs against the wall surfaces in the corners and the membranes facing into the chamber—the results were also higher in range and center frequency than expected.
- Additionally, we were testing our design assumption that the most effective and efficient location for the membrane absorbers would be closely-coupled to and in parallel with wall surfaces in room corners, placed on the floor. This design assumption proved to be accurate.

To summarize our test results: Some of the design assumptions we made, based on our understanding of how low-frequency diaphragm absorbers work in rooms, were incorrect and caused us to re-think how we designed and specified the final product. The type of absorber and its location and orientation in a room are all highly critical to low-frequency absorber effectiveness.

Photo 4 shows two prototype CornerSorber panels in use. The test results prove that this close-corner location makes the CornerSorber a very efficient and effective low-frequency absorber and can realistically reduce destructive low-frequency room modes.

This set of tests proves the value of actual laboratory testing. If we had released these low-frequency absorber products to the public without accurate testing—guided only by intuition, experience, broad theory-based assumptions, impedance-tube testing, interpolation, or other approximations of product effectiveness in real-world, in-use configurations—our claims of product absorption ranges and efficiencies would have been wrong. This cannot be overstated: Without standardized laboratory absorption testing in a lab capable of accurately testing absorption down to 40 Hz, we would not be able to state conclusively that our low-frequency absorber products perform as we claim. We believe this holds true across product categories.

Applications

The Acoustic Bass Management (ABM) System consists of the newly-released CornerSorber, a unique close-corner-located MLV membrane bass absorber

(see **Photo 5**), and the Curve Diffusor, containing a built-in MLV membrane bass absorber (see **Photo 6**). The two products, used in combination, have been shown in separate tests at NWAA Labs—under real-use mounting conditions—to effectively absorb low frequencies from 45 Hz to over 200 Hz. Sound is only pressure at boundaries, not velocity. ABM System absorbers are pressure-based, and are the most efficient and effective way to mitigate boundary-created room modes.

The dimensional equivalents for room-mode frequency (wavelength) absorption ranges for each product are (approx.): 5.5′ to 14′ (200 to 80 Hz) for CornerSorbers, and 14′ to 25′ (80 to 45 Hz) for Curve Diffusors.

CornerSorbers have an approximately 3-to-1 efficiency ratio to Curve Diffusors—they are dedicated low-frequency absorbers, whereas Curves are also highly-effective phase-coherent cylindrical (constant-radius) diffusors (see Photo 6). We recommend a minimum of six Curves and two CornerSorbers per room, depending on room size and geometry, and room diffusion requirements. Further information about the Curve Diffusor and CornerSorber low-frequency absorption Test Reports (per ASTM C423-09a) is available on Acoustic Geometry's website (www.AcousticGeometry.com).

Test Reports

Acoustic Geometry tested both the Curve Diffusor and the CornerSorber for low-frequency absorption per ASTM C423-09a protocol at NWAA Labs. The Curve Diffusors were tested on August 18, 2015 and the CornerSorbers tested on September 8, 2016. NWAA Labs has the world's largest absorption test chamber (approximately 42′ × 35′ × 17′, 26,000 ft^3 or 738 m^3 volume), which can accurately test with better than 95% precision and bias (variation) down to 40 Hz.

The tests were conducted with the test units mounted as they would be in real-use configurations. The curves were mounted on the test wall (A-Mount) with standard Z-Clips, with 21 units covering 120 ft^2, which is the NWAA Labs standard minimum test-sample area (both medium and small curves with an equivalent membrane area of 19 Medium Curves).

CornerSorbers were set on the floor (J-Mount), oriented horizontally in close-corner configuration, with 18 units covering 126 ft^2 (overage accounted for in calculations), grouped six units per corner, in three of the test chamber's corners.



Photo 6: Curve Diffusors are a natural-sounding improvement to room acoustics. Acoustic Geometry's ABM system allows for a high level of low-frequency performance when utilizing both CornerSorbers and Curve Diffusors, which also feature accurately-tested low-frequency MLV membrane absorption. The combination has a complementary range of frequencies that produces consistent bass-energy absorption from 50 Hz to 500 Hz. The CornerSorber's form factor of 24″ × 42″ with a 6″ depth provides a small footprint and either vertical or horizontal orientation. With the Curve's smooth phase-coherent diffusion, the combination of Curves and CornerSorbers will evenly diffuse frequencies above 500 Hz and evenly control frequencies below 500Hz, which is the ideal acoustic treatment solution for nearly any room.

Resources

J. Calder, "New Research on Low-Frequency Absorption Using Membranes," White Paper, Acoustic Geometry, www.AcousticGeometry.com

Acoustic Geometry Channel Trailer, www.YouTube.com/AcousticGeometry.

NWAA Labs, www.nwaalabs.com.



JOCAVI AS Acoustic Shells

Enhancing the Stage

Acoustic science intersects art and sound engineering. In its continuous search for the best possible acoustical solutions, Portuguese company JOCAVI Acoustic Panels tapped into its knowledge and understanding of practical effects of acoustics in performances to create its successful AS Acoustic Shell catalog.

By
João Martins
(Editor-in-Chief)

Since 1992, JOCAVI Acoustic Panels—based in Sintra, Portugal—has gained recognition for supporting some of the most prestigious acoustic consultants and designers worldwide, while also gradually expanding distribution of a vast catalog of acoustic panels and solutions for virtually any kind of application—from studios, commercial and industrial applications to performance spaces. The JOCAVI catalog is particularly interesting because of its unique combination of products and designs, particularly diffusers, absorbers, and low-frequency tuned panels—all manufactured using carefully selected high-quality materials, including natural and recycled resources. All solutions are available in diverse colors and finishes and are extensively documented. JOCAVI also offers first-approach online consultancy services.



"The use of our acoustic shells increases the natural level of the sound that is genuinely produced by the instruments and naturally increases the level of the acoustic pressure of the room, thus leading to a balanced diffusion throughout the whole range of the sound spectrum," states João Carlos Vieira, JOCAVI's founder.

The AS Acoustic Shell brand and range of products are a result of its founder, João Carlos Vieira's vast experience as a sound engineer, combining his passion for acoustics and his deep understanding of sound reinforcement. What started as side project to create a highly specialized acoustic shell, quickly evolved into a complete range of unique solutions, currently under great demand globally. The positive response to JOCAVI's Acoustic Shell range is even fueling an interesting business of "long-term rental" contracts for events—a well-received side business for sound rental and production companies—while generating a continuous flow of inquiries for permanent installation in theaters, auditoriums, rehearsal and performance spaces.

JOCAVI carefully researches, plans, and carries out the installation and commissioning for each project, given the extremely demanding requirements and the need to directly interact with the multiple entities normally involved in such projects, from government representatives to cultural organizations and, naturally, musicians, musical directors, technical directors, and conductors.

audioXpress was invited to witness the commissioning and debut of one such installation, when the highly specialized acoustic shells were used by the prestigious Malta Philharmonic Orchestra and premiered in a recent concert of Beethoven's "Eroica" in the historical Republic Hall, part of the Mediterranean Conference Centre in Valletta, Malta. In this installation, a series of unique Woodfoil Acoustic Shell portable panels dramatically improved sound diffusion during the performance, with results praised by the orchestra, the public, and the government

executives responsible for the institution. As Vieira explained, "While the requirements for such solutions are extremely demanding, when the parties involved understand the benefits and quality of results, the word quickly spreads. Orchestra musicians, conductors, and artistic directors in some of the more demanding fields have been the best ambassadors for our Acoustic Shell range. Once they start using it, they don't want to perform without it."

The Acoustic Shell Range

JOCAVI's AS Acoustic Shell range was designed for optimal diffusion in theaters and concert halls and includes different ABS and wood panels with unique geometries, which make them particularly effective in any sort of live entertainment environment. The installation of these acoustic diffusion components, available in permanent installation and portable versions is meant to project the non-amplified original sound from the stage toward the audience, while also enhancing the performers' awareness of stage sound. This makes the acoustic shells ideal for rehearsal spaces and a piece that changes the room's acoustics by enhancing its features.

This is an application area where acoustic science interleaves with art and directly connects with sound engineering. Being an experienced audio engineer in live sound and understanding the shortcomings of electroacoustic sound reinforcement in different music genres and performances, Vieira started to research ways to enhance natural acoustics and the music sound source generated by the musicians themselves.

Having toured extensively with different artists, Vieira was fascinated with the differences the concert spaces would trigger, both in the musicians' mood and the audience's reactions to the performance. While already involved in the acoustic treatment business with his own company, Vieira couldn't help thinking how he could apply his research on acoustic panels to deal with those variations, trying to dispense with using a complete public address reinforcement system.

"Acoustic comfort for a musician or musical director is fundamental. A room with bad acoustics is a frustrating situation for musicians, maestros, and sound technicians. The quality of the interpretation of musical pieces may be enhanced when conditions are excellent. The acoustic shell harmonizes the sound and enables musicians to clearly listen to each other, thus making the performance pleasant and perfectly harmonious. It also helps to highlight or mildly moderate the several groups of instruments in the orchestra, in accordance with the Maestro's preferences," Vieira states.

The JOCAVI AS Acoustic Shell range includes acoustic diffusing components that naturally enhance the sound coming directly from its sound sources and instruments, without the inherent electro-acoustic characterization or coloring. The panels also prove extremely effective in creating a proximity effect between the stage and the room, eliminating the apron separation. JOCAVI's different EffectFuser, Plura, DynamicFlow, and Woodfoil Acoustic Shell models have been designed at the specific scale of these needs with a shape and depth to generate a high diffusion coefficient on higher frequencies as well as medium/low frequencies, down to 100 Hz.



Orchestra rehearsal session using the Woodfoil Acoustic Shell. The historical Republic Hall, part of the Mediterranean Conference Centre in Valletta, Malta, is a beautiful historical building, but it wasn't designed to be a concert hall.

Manufactured in ABS or wood with a rigid framework, these pieces can be coupled and multiplied to suit each project's demands and the specific areas, emphasizing the sound diffusion at different covering angles and shorter or longer incidence distances, with more or less accentuated scattering effects, which focuses or widens the energy. Modules can be stacked or hung from the ceiling to generate optimal sound diffusion in the required angles and can even be mounted with a motorized rigging system over the stage. Their low weight also makes mounting easier.

Based on field experiments with sound wave diffusion and the positive aspects that result from



The Malta Philharmonic Orchestra performs Beethoven's *3rd Symphony*, "Eroica," on March 11, 2017. This was the first concert using the new Acoustic Shell.

JOCAVI's EffectFuser, Plura, DynamicFlow, and Woodfoil Acoustic Shell panels



the presence of diffusers in rooms, the Woodfoil Acoustic Shells can be coupled and multiplied for individual projects and to obtain a diffusing area that is proportionate to the space in question. Each vertical module uses four 120 cm × 120 cm (3.9′ × 3.9′) pieces, creating a diffusing homogeneous surface of 480 cm × 120 cm (15.7′ × 3.9′), enabling different diffusing effects, depending on its position and pattern combination.

The Range

Following its research on acoustic panels, JOCAVI adopted four of its models of acoustic diffusers to create acoustic shells. The EffectFuser, the Plura, the DynamicFlow, and the Woodfoil Acoustic Shells can all be placed at different angles, defined according to the degrees of incidence toward the audience. The range was designed to be versatile and discreet, so that their presence doesn't hinder the use of the space for the most varied types of performances, while the use of light and attractive materials also facilitates their practical use and provides nice aesthetics when their presence is highlighted. All shells can be adapted to the performance needs by increasing the number of modules and can be easily assembled and disassembled, shipping in purposely designed flight cases for touring applications. The cases can also form part of the basic structure of the shell itself, creating a solid footing. Two people can easily install a complete acoustic shell on the stage floor.

Manufactured with recycled ABS, EffectFuser diffusing acoustic shells feature a unique design with large protrusions, only possible to obtain through a combination of computer-aided design and molding. They are large-sized M1 fire-resistant diffusers that provide a uniform diffusion within the sound spectrum above 180 Hz. Available in two different sizes—180 cm × 120 cm (7.8′ × 3.9′) and 120 cm × 120 cm (3.9′ × 3.9′)—EffectUser panels can be configured in many ways, to emphasize the sound diffusion with a smaller covering angle, effective at a longer incidence distance and generating higher sound pressure levels, or emphasizing the sound diffusion at a wider covering angle, effective at a shorter incidence distance.

Plura acoustic shell modules are meant to diffuse mid and mid-high frequencies, ideal when used in large rooms. The panels can be rotated 90° or 180° between them, and positioned according to the



These images detail the use of the DynamicFlow Acoustic Shell, which has been a popular solution in JOCAVI's range.



Acoustic Shell portable module examples are configured with flight cases.

room's requirements, while also providing interesting aesthetic effects. Since the shape of the Plura ABL model is also an absorbent, when applied together in a shell, the combination provides sound diffusion and absorption. The inner part of this model uses a composite substance of impregnated mineral fibers and textiles, which gives it a specific mass and contributes to its consistence. The use of ABS as the external material provides the required properties for a diffuser, while providing UV protection, impact resistance, and fire resistance M1.

DynamicFlow ABS acoustic shells are easy-to-install portable acoustic diffusion shells meant to be used in musical concerts. The shells change the room's acoustics by enhancing its features. DynamicFlow includes 180 cm × 120 cm (7.8′ × 3.9′) or 120 cm × 120 cm (3.9′ × 3.9′) modules, designed with unique non-linear convex shapes, based on a set of three ellipses optimized for angular diffusion coverage. The interior also combines impregnated mineral fibers and textiles to create a specific mass and consistence. The back part consists of an even surface and the modules can be also applied on "T" profile false ceilings or on walls. Depending on the space available on stage, elements may be used to create the shape of a perfect shell. They may also be used outdoors.

With a clearly distinctive and more classic look, Woodfoil acoustic shells use slightly concaved diffusion panels, made of varnished birch plywood on a soft wood structure. This diffuser is ideal to be used in concert halls and provides two options: the Woodfoil diffusion panel, which is made of plain birch plywood, and the Woodfoil Ab, which has holes to provide a higher absorption coefficient. Its format enables users to create different effects through 90° rotations, making it extremely versatile. In medium or large size configurations, it provides an effective diffusion of sound energy to 100 Hz, while also providing an aesthetically pleasing background for performances.

According to Vieira, the development of the Acoustic Shell range started with the DynamicFlow and the EffectFuser models, with its large protrusions that were enabled by the production method and the modulation scale. The main differences with the newer Plura and the Woodfoil models is the result of trying to achieve a large scattering factor, providing a lively result with larger impact in the lower frequencies, while not being as effective in the diffusion of the higher frequencies as the DynamicFlow model, which typically enhances the attack and the detail on string instruments. For purely acoustic applications without any active reinforcement, the wood model is more balanced, also creating a stronger effect in projecting energy in a larger scale.

The Woodfoil models are designed to create different effects, by rotating each panel in a large acoustic shell array. Combining four panels covering 2.4 m × 2.4 m (7.8′ × 7.8′), the Woodfoil can create a concentric diffusion area with a concave surface, enabling a stronger projection at larger distances but with more focus; or a convex surface that provides a wider angle of diffusion, with less energy in terms of projecting the sound.

On stage, the two shapes can create the desired effect of providing the correct sound environment for the performers, while enabling a stronger or more subtle projection toward the audience and providing a tight control of the projection angle, as needed for the auditorium. This is ideal for orchestras, where recommended reverberation times vary between 1.7 to 2.5 seconds, while certain types of performances



A stage perspective of the Woodfoil Acoustic Shell solution, configured in A/B mode for this performance, was designed to be used wherever the Malta Philharmonic Orchestra plays.

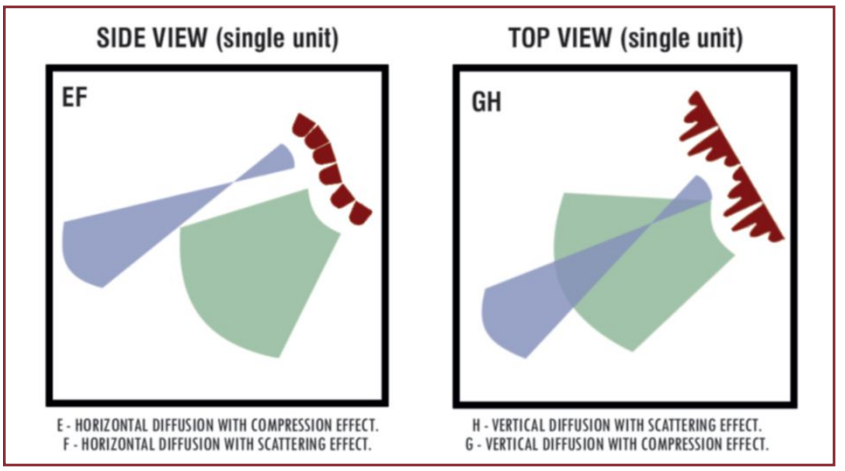


A back view of the Woodfoil Acoustic Shell solution, which uses the custom-built flight cases as a solid footing.

would be clearly impossible to sustain with such reverberation times.

Each model in the AS Acoustic Shell range was carefully designed to respond to different types of performances, music, and environments, while its modularity enables users to carefully tune the acoustical effect according to the real conditions. The differences between Acoustic Shell models also include key features for the musicians on stage, by scattering energy in a very effective way. Not surprisingly, touring performers that have adopted the DynamicFlow shells, express their satisfaction precisely for the fact that they are able to create consistent conditions wherever they perform, while adjusting the projection effect also according to the needs in each location.

The choice of materials is one of the main assets as is the formula used by JOCAVI to optimize each project, which includes the weight and rigidity of the surface materials used on each shell (e.g., the wood on the Woodfoil shells). The DynamicFlow panels are always assembled in a fixed position, have a smaller scale, and still provide a horizontal scattering effect of 63° to 67°, while the vertical dispersion goes from 15° to 17°. When creating the larger shell structure using the existing modules, which are all the same, the arch angle can be accentuated or diminished with corresponding effectiveness.



This diagram shows the positive combined effect of the compression and scattering vertices present in the acoustic shell surface.

"JOCAVI always spent a lot of its research focusing on the study of diffusion and scattering effects. While absorption panels are more common, diffusers are in a way part of how a company addresses the market and a clear differentiator. JOCAVI's catalog is particularly rich in its variety of diffuser panels," explains Vieira.

"JOCAVI also distinguishes its approach by providing a visual simulation for all products and projects, using 3D computer animations. Every project is documented with a video of the simulation computed by the software, using key parameters such as weight, density, porosity, and environmental parameters such as temperature, humidity, etc., while considering a sound source in a specific frequency bandwidth in a specific volume, according to specific templates that we use consistently. Every model is extensively tested in terms of shape. With all these variables at our disposal, the computer simulation becomes a powerful tool for researching different designs but also to proof-test every single project as closer as possible to its real dynamics," he adds.

For the AS Acoustic Shell range, the challenges also included the need to find the most effective solution considering artistic and even aesthetic requirements. An area where Vieira's extensive experience working with performing artists proved valuable.

"The art of sound diffusion is always depending on the actual measurement in real world conditions, since all "lab measurements" are based on anechoic chamber measurements," Vieira states. "With the acoustic shells, JOCAVI targets specific values of reverberation and SPL that have now been consolidated in real conditions, by measuring the real impact and comparing those parameters before and after. After that, it is the acceptance by the critical hears of musicians, conductors, etc., that validates the effectiveness of the implemented solution. Acoustics are something that everyone recognizes has a great impact, even if in general people don't understand how that impact is felt. With our Acoustic Shell range, performers recognize the actual comfort resulting from the ability to hear themselves better, the actual definition of individual notes and pitch, which contributes to a better performance."

The Performance

For the Malta Philharmonic Orchestra (www.maltaorchestra.com), JOCAVI delivered a "first-of-its-kind" Woodfoil Acoustic Shell solution, configured in a portable version. The installation was provided by JOCAVI's distributor in Malta, Digital Magic Pro Audio Co., which specified the shell solution for both rehearsal situations and different orchestra performances.

On March 11, 2017, the orchestra had its first concert using the Acoustic Shell at the historical Republic Hall, which is part of the Mediterranean Conference Centre in Valletta, Malta and an historical building from the 16th century that was converted to a conference center and auditorium in the 19th century. Unfortunately, the space was not designed to be a concert hall and provides several acoustic challenges.

As Alec Massa, Digital Magic's technical director, explained, "The space on stage was not designed for music and we looked at having some kind of solution to better project the sound of the orchestra back to the audience and for the musicians on stage to hear themselves in a much clearer way. The portable Woodfoil Acoustic Shell model was chosen because the installation was not permanent and the orchestra needed something that could easily be assembled and disassembled, wheeled in and wheeled out to venues where the orchestra will perform. In this space, the transformation on how the orchestra is heard was clear. For the orchestra, it's like a revolution. The solution provides a much better experience for the orchestra and for the public."

The inaugural event was also an important moment for the orchestra and Malta's government, which funds the institution. Malta held the European Union's presidency and the orchestra had a full schedule of events. For the occasion, they performed Beethoven's 3rd Symphony—"Eroica"—under the baton of the orchestra's Principal Conductor and Artistic Director, Brian Schembri. The first part of the concert featured Nicolas Isouard's "Joconde Overture," followed by Johannes Brahms' "Double Concerto in A minor," with renowned soloists Jiman Wee (violin) and Kirill Rodin (cello) joining the orchestra to perform.

Schembri, was extremely pleased with this first experience with JOCAVI's WoodFoil Acoustic Shell solution. As he explains, "In this hall, we had a problem of sound projection. Of the sound on stage, where the musicians had problems to listen to each other, we had a problem of not really being able to judge the dynamic range of the orchestra. Sometimes, since the acoustics where working against us, the orchestra had to play forcefully to get a quality sound and so our dynamic range was very limited, and obviously, it was very tiring. In the rehearsals, we tried different positions, and I think the installation helped us a lot. Not only to have more comfort but even to be more attentive to details, allowing us to work on the balance of the sound and know where the problems are. Now, I wouldn't want to do a concert without it."

The orchestra musicians also praised the experience and the acoustic dynamics, allowing them to hear more detail and the different instrument sections. "Now we just need to play normally without applying so much energy," one of the musicians stated.

Being both a pro audio distributor and service provider for the region, Digital Magic already has multiple successful installations with JOCAVI's acoustic panels in Malta and in fact the company had already supplied multiple solutions for the Malta Philharmonic Orchestra spectacular rehearsal building (in a former church), which is also undergoing renovations. Among several other solutions, the orchestra uses JOCAVI's Wallblind portable acoustic blind panels in some of their rehearsal rooms, while the philharmonic is already looking at expanding the Acoustic Shell configuration adding extra panels to accommodate different performances.



The Malta Philharmonic Orchestra performs with the new acoustic shell solution. In the auditorium, the positive effect on increased acoustic pressure and the detail of single notes and instruments was clearly noticed. The inaugural event was hailed as "a much better experience for the orchestra and for the public."

"The installation and the measured response proved once again what our simulation had anticipated. That its effectiveness in the medium lower and lower frequencies was much higher, because of the added mass of the Woodfoil panels. Its configurability allows for a very effective result. In the forte sequences of the music, and moments with the full orchestra playing, the positive effect in lower frequencies—as low as 80 Hz—could be clearly heard. Also, the sound projection across the auditorium was amazingly consistent all the way to the last upper rows, with excellent presence and definition," concluded Vieira.

www.jocaviacousticpanels.com | www.acousticshell.net



"It's all a new experience. Now, I wouldn't want to do the concert without it," commented Brian Schembri, artistic director and principal conductor of the Malta Philharmonic Orchestra, after his first concert using the Woodfoil Acoutic Shell. (Image courtesy of Jocavi/Option+)

Williamson's Amplifier Games

Learn more about David Theodore Nelson Williamson's power amplifier design, which incorporated negative feedback and explore Talbot M. Wright's suggested improvements to decrease Intermodulation Distortion (IMD).



By
Richard Honeycutt
(United States)

The earliest audio power amplifiers used a single-ended vacuum triode, and were quite suitable for most home radio receivers. However, serious listeners wanted more powerful amplifiers, given the inefficient speakers of the time. Fortunately, in 1915, Edwin H. Colpitts invented and patented a power amplifier using two Audion tubes in a push-pull circuit (US Patent 1137384), although he did not specifically claim the push-pull connection in his patent.

RCA introduced the first commercial push-pull power amplifier in 1924 for use in its Radiola III broadcast radio receiver. And the late 1920s saw the introduction of additional push-pull amplifiers to serve the cinema sound market that emerged after the introduction of talkies following the release of *The Jazz Singer*. (This movie—the first feature-length film originally released as a talkie—came out in 1927.) By using a balanced topology, push-pull amplifiers cancelled even-order distortion, which is the predominant order of harmonic distortion in triode amplifiers.

Williamson's Design

Although Harold Stephen Black of Bell Telephone Laboratories invented negative feedback in 1934, it was not used in power amplifiers until David Theodore Nelson (DTN) Williamson introduced his power amplifier design in *Wireless World* in April and May, 1947. His design incorporated negative feedback that included the output transformer, and thus produced noticeably less distortion than previous designs. Other reasons for the low distortion included the use of a cathodyne or "split-load" phase splitter and a balanced (essentially transformerless push-pull) driver stage. Maintaining stability against regenerative feedback due to phase shifts in the output transformer made design of the compensation network in Williamson's amplifier critical.

The original design used four L63 single triodes, with two KT66 beam power tubes and a U52 rectifier having a directly heated cathode. Williamson stated that there was no exact replacement for an Osram KT66, but a 6L6 or 6L6GT could be substituted, if a design change were made: the total anode and screen dissipation must be reduced from 25 W to 21.5 W. This involved reducing the total current from 125 mA to 110 mA through adjustment of potentiometer R21, which comprised part of the cathode resistor for the cathode-biased pair of output tubes.

In 1949, Williamson published an updated design, including several small changes to the circuit, along with a list of alternative triodes (two 6SN7 or Osram B65 dual triodes). Changing the rectifier to the indirectly heated Cossor 53KU or 5V4 prevented a voltage spike at turn-on that could damage the output tubes. (In general, B+ should not be applied to power tubes' plates until the tubes are warmed up. For this reason, tube-type high-power radio transmitters had separate switches for filament power and B+.)

The Radiotron Designer's Handbook (*RDH*) discusses both the original and the "new" Williamson circuit, and includes a complete schematic of the "new" version including the power supply. The *RDH* schematic also shows the peak signal voltages present at various

points in the circuit when it is outputting 15 W, as well as the operating currents of the various tubes.

Numerous individuals and manufacturers have created variations of the Williamson amplifier since the original's introduction. Probably the most significant was the introduction of the ultralinear method of feedback application. Often newer dual triodes (e.g., the 12AU7s) have been used instead of the original 6SN7s. Higher-power output tubes have also been used, the original 15 W maximum output being considered too low by many users. With the change in output tubes came a need to design a new output transformer, since the phase shift contributed by the transformer is critical in the Williamson design.

Wright's Design Improvements

In 1961, *Electronics World* published an article by Talbot M. Wright suggesting improvements to the Williamson amplifier to decrease Intermodulation Distortion (IMD). Wright suggested changing the values of three resistors to optimize the Q point of the first voltage amplifier, the cathode follower, and the phase splitter. He stated that these changes were necessary because of differences between the operating characteristics of European vs. American 6SN7s. My requests for details from tube experts on three continents have revealed such differences, a point we will discuss further later in this article. Regardless of the cause of the original Williamson amplifier's high IMD (Wright states about 2% at 8 Ω), Wright's article stirred my curiosity. First, it is important to note that Wright was discussing Williamson amplifiers having a rated power output of about 50 W: about 3.33 times that of Williamson's original. Changing from KT66 output tubes to 6550s (as an example) would involve changing the output transformer primary impedance from 8 kΩ to 5 kΩ plate-to-plate, and would change the necessary drive voltage and current to and from the driver stage.

Testing Wright's Modifications

Since Wright's modifications only affected the first voltage amplifier, the cathode follower, and the phase splitter stages, I modeled the first three stages. Without the rest of the amplifier in the model, I could not include the effects of feedback, but I could look at the DC performance, which would not be affected by AC feedback, and I could see whether Wright's modifications did reduce the percentage of IMD (%IMD). Naturally, the open-loop IMD results would not be as good as the results with feedback, but if the modifications improved IMD, that would affirm Wright's work. Another unknown is the effect of the output stage and the output transformer on IMD.

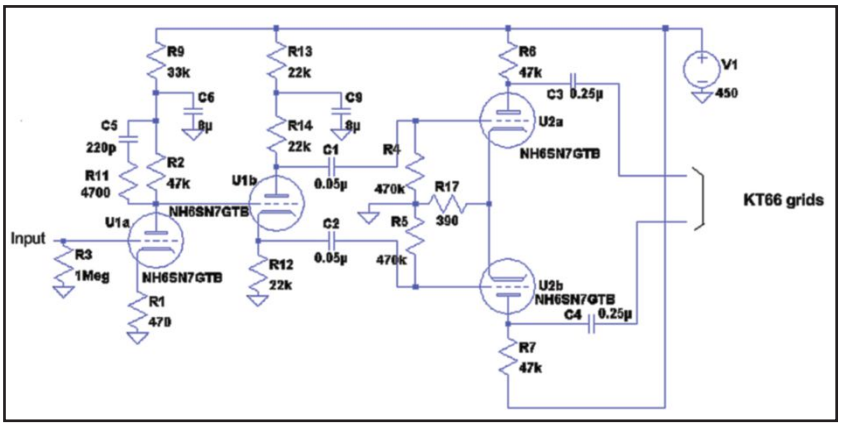


Figure 1: This shows the first three stages of the original "new" Williamson amplifier circuit.

Figure 1 shows the schematic of the LTSpice models of the original 15 W Williamson circuit as published on page 750 of the *RDH*. **Figure 2** shows the schematic of Wright's modified circuit. As you can see, Wright changed the following resistors:

- First-stage cathode resistor from 470 Ω to 910 Ω,
- First-stage decoupling resistor from 33 kΩ to 47 kΩ
- Phase-inverter decoupling resistor from 22 kΩ to 3900 Ω

He also changed the feedback resistor to 10 kΩ, but since feedback was not included in the LTSpice model, this resistor does not appear in Figure 2. In the *RDH* reprint of the Williamson circuit, the feedback resistor is given as "1200√(voice coil impedance)." But since Wright says he doubled the value of that resistor, his "original circuit" must have used about a 5 kΩ feedback resistor. 1200√16 = 4800 Ω, for a 16 Ω load. The 150 Ω resistor and 68 µF capacitor in the driver stage B+ supply did not appear in the original Williamson circuit, which used two filter chokes rather than the single one shown by Wright, so Wright's pre-modification circuit must have differed from the Williamson original on this point, or Wright did not list all his modifications.

Figure 3 and **Figure 4** show the load lines for the

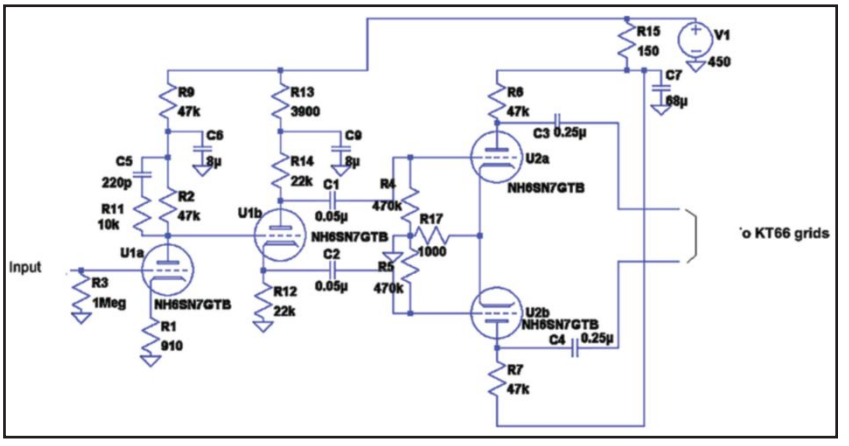


Figure 2: This circuit shows Wright's modifications.

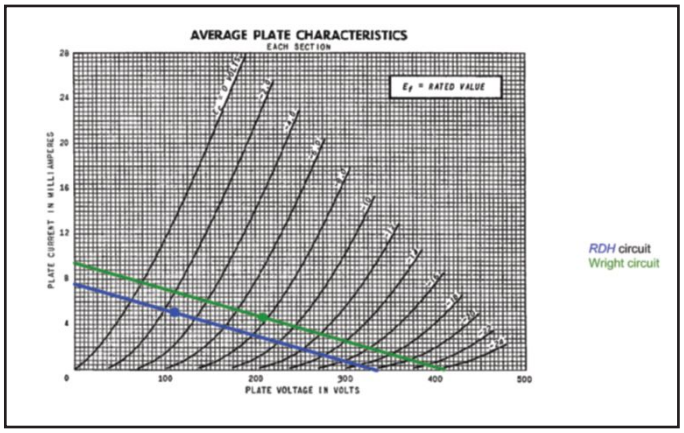


Figure 3: These load lines show the effect of Wright's modification on the phase splitter.

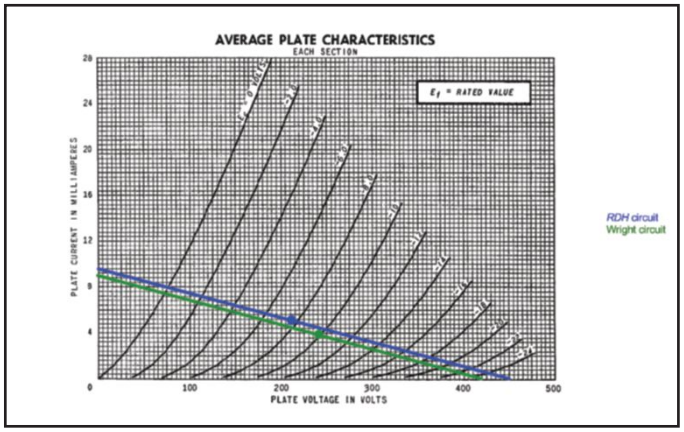


Figure 4: These are the original and modified load lines for the driver stage.

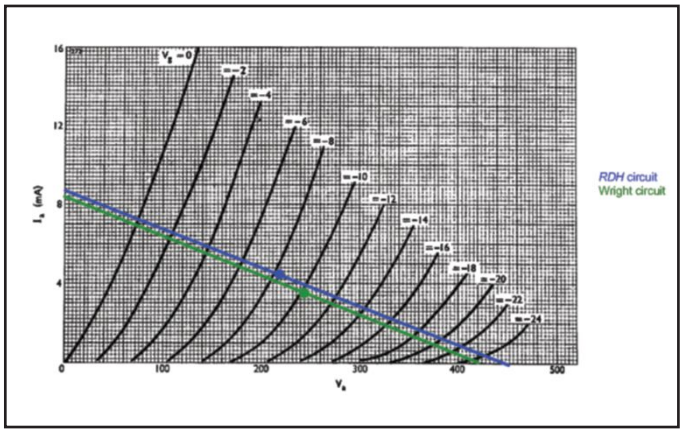


Figure 5: The Q-points are slightly different when the B65 dual triode is used instead of the 6SN7.

phase splitter and the driver stage, respectively. These help us to determine how much change in Q-point Wright's modifications made. As you can see, the more significant change was in the phase splitter. In my experience with hollow-state push-pull amplifiers, as you increase the input signal level, distortion appears most prominently in the phase splitter, so changing this stage could be expected to have the greater effect on IMD.

Note that the grid bias point for the Wright version in the driver stage is plotted according to Wright's stated plate voltage, but does not match the grid bias voltage he gives. Since his results were presumably measured, we might assume that his 6SN7s did not match the published curves.

While it is possible to predict the percentage of even-harmonic distortion by load-line analysis, IMD cannot be well predicted that way. Some references specify a constant ratio of %IMD/%2HD and for %IMD/%3HD, but these ratios are only very rough approximations.

Remembering that the true original "new" Williamson circuit used B65 triodes rather than 6SN7s (although Williamson later "blessed" the use of 6SN7s), we might find it instructive to look at the load lines plotted on the B65 characteristic curves, as shown in **Figure 5**. Having placed the Q points according to the quiescent plate voltages shown in the original and modified schematics, we find that the grid bias needed to obtain these plate voltages changes from just under -8 V for the *RDH* circuit and just under -10 V for the Wright circuit, to just over -8 V and 10 V, respectively. Perhaps rather than speculating about the existence of different "European" 6SN7s, Wright should have noticed the difference between the original B65 curves and those of the 6SN7 that has been used more recently. It is also notable that Williamson amplifiers have been built using ECC32 or ECC33 tubes for the triodes, and neither of these exactly matches either the 6SN7 or the B65. (Thanks to Alan Otto at Electro-Harmonix for alerting me to this information!)

It is always instructive to compare the DC voltages predicted by a model with those obtained by measurement of the actual circuit. **Table 1** does just that. As you can see, the model does not predict exactly (no model ever does), but it is within 10% or better for all the points that are compared.

Figure 6 and **Figure 7** show the signal amplitudes in peak AC volts at several levels in the original and modified circuit. Remember that these voltages are larger than they would be if the Williamson negative feedback were present. The value of 0.19 V_P is Williamson's stated input voltage for a 15 W output. If it were possible to obtain a 50 W output without changing the output transformer or output tubes, the necessary input voltage would be 0.35 V_P. The LTSpice model was run with a pretty extreme range of input voltages, just to see what it would do as it began significant clipping.

DC voltages and AC peak levels are all well and good, but what

	Stage 1 plate	Phase splitter cathode	Phase splitter plate	Top driver plate
RDH per schematic	100	105	215	215
RDH model	91.75	97.56	254.5	221.6
Given in Wright article	95	102	310	250
Model of Wright circuit	108.61	116	313.17	224.54

Table 1: DC voltages

we're really interested in are the waveforms. **Figures 8–10** show the output wave from the driver stage (actually, the output from the top driver plate minus the output from the bottom driver plate), to include the even-order-distortion-cancelling effects of the push-pull connection. The three figures show outputs for input voltages of 0.19 V, 1 V, and 1.5 V, respectively. Also included are pure sine waves (blue traces) for comparison, to make the distortion easier to see. I did not take the time to create an LTSpice model for the B65, so we don't have those waveforms. Note that for the larger inputs, Wright's modifications do indeed clean up the appearance of the output waves from the driver stage.

%IMD Conclusions

So, what can we find out about the %IMD of the two circuits? **Figure 11** shows the %IMD for both circuits, across the range of input voltages tested. Wright's article shows about 0.25% IMD at 15 W output (corresponding to a 0.19 V_P input) for his modified circuit, compared to about 2% for the original Williamson circuit. Of course, since Wright's measured results included the output stage with the transformer, as well as negative feedback, his results would not be expected to match the modeled results. But just examining the IMD performance of the first three stages of each circuit in an open-loop configuration does illustrate the benefits of Wright's modifications.

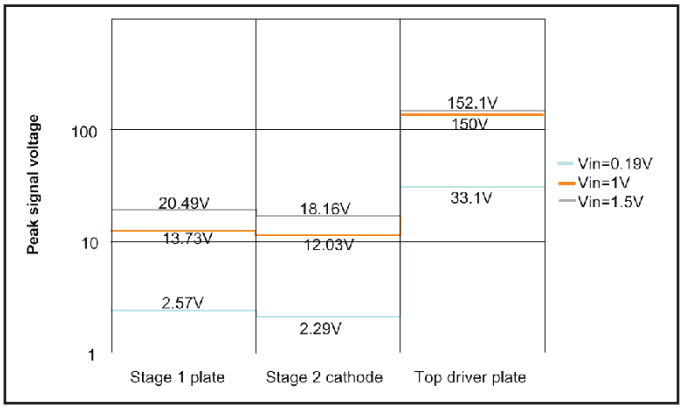


Figure 6: This chart shows peak signal levels at several points in the original Williamson circuit.

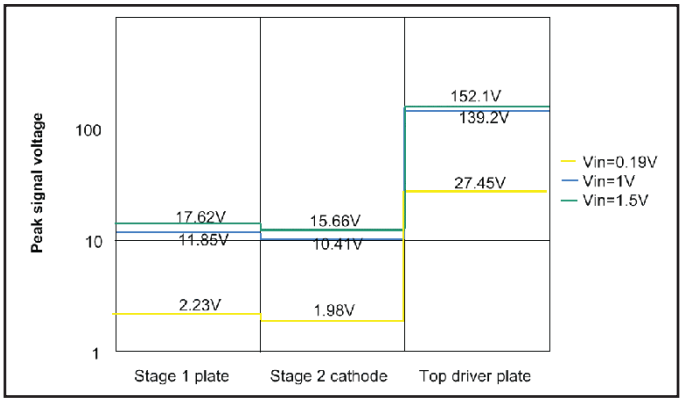


Figure 7: This chart shows peak signal levels at several points in Wright's modified circuit.

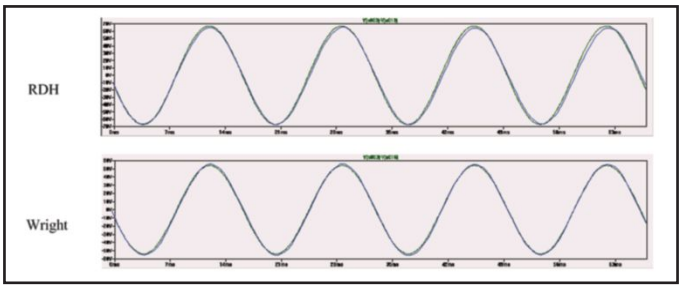


Figure 8: Both circuits produce clean waveforms with 0.19 V_P inputs.

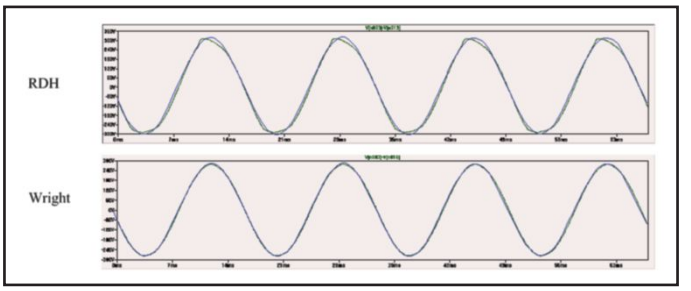


Figure 9: With a 1 V_P input , the original circuit is beginning to clip asymmetrically.

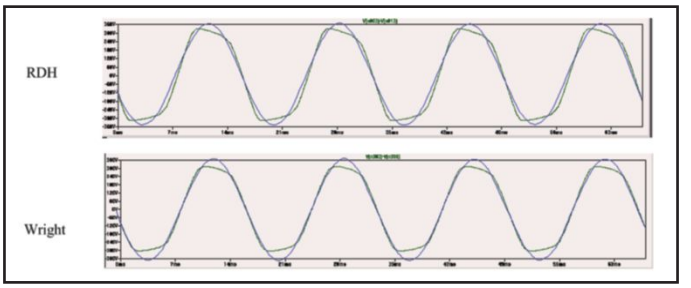


Figure 10: Increasing the input to 1.5 V_P creates significant clipping in both circuits.

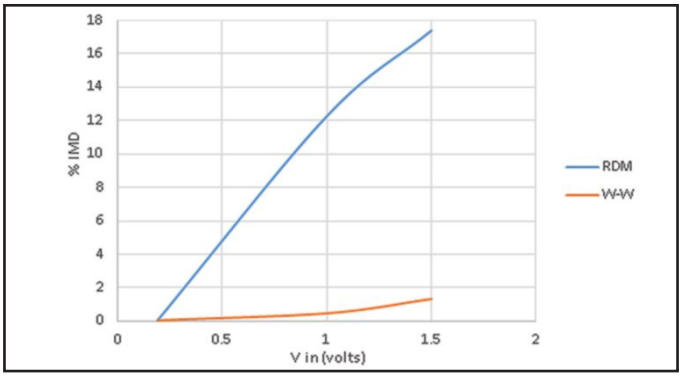


Figure 11: The percentage of intermodulation distortion (IMD) for the original circuit becomes quite high as input voltage is increased.

Resources

A. Blumlein, Patent No. 496,883, 1937.

D. Fraser, "Dynaco ST-70 Series 3, The Inside Story—a Long Road," audioXpress, May 2017.

F. Langford-Smith, Radiotron Designer's Handbook, Amalgamated Wireless Valve Co., Pty. Ltd., Sydney, 1952.

D.T.N. Williamson, "High-Quality Amplifier—New Version," *Wireless World*, August 1949), October 1949, and November 1949.

———, "Design for a High-quality Amplifier," Wireless World, April/May 1947.

T. M. Wright, "Improving the Williamson Amplifier," *Electronics World*, June 1961.

Industry Calendar

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

August 11–13, 2017
2017 Hong Kong High End Audio Visual Show
Hong Kong Convention and Exhibition Centre in Wan Chai, China
www.audiotechnique.com/avshow/11700



This three-day event, sponsored by Far Eastern Music & Audio Enterprises, Ltd., provides an opportunity to showcase all types of audio products in a vibrant market, find new suppliers, and do business, with plenty of displays in personal audio, headphones, speakers, cables, accessories, and all audio components.

August 21–24, 2017
SET EXPO 2017
Trade Show: August 22–24
Conference: August 21–24
International Technology Fair on Equipment and Services for Television Engineering, Broadcasting, and Telecommunications
Expo Center Norte Red Pavilion and Convention Center, São Paulo, Brazil
www.setexpo.com.br

The Brazilian Society of Television Engineering (SET) is one of the most prestigious associations in Brazil and Latin America with direct connections to the National Association of Broadcasters (NAB) and the International Broadcasting Convention (IBC) industry associations. Even though this is a broadcast-oriented event, SET Expo is undoubtedly the largest and most dynamic business opportunity for the audio and video industries in the Latin American region. SET is also increasing its cooperation with the local section of the Audio Engineering Society (AES). With a strong focus on radio—which is still a vibrant segment in the region—the SET Expo benefits from a strong direct presence of manufacturers and country delegations, with more than 21,000 m^2 for exhibits and over 400 brands exhibiting.

August 30–September 2, 2017
3rd AES International Conference On Sound Reinforcement—Open Air Venues
Struer, Denmark
www.aes.org/conferences/2017/sr

This conference on sound reinforcement is a follow-up to the 2015 conference in Montreal and will focus on sound reinforcement in open-air venues. It will bring together recognized hands-on experienced practitioners, scientific experts, self-taught gurus, young technicians, and students in a creative environment. There will be workshops, demonstrations, and live experiments combined with paper presentations, engineering briefs, and tutorials. Live performances will demonstrate the most advanced and up-to-date sound systems operated by renowned sound engineers on an outdoor stage. Technical papers and tutorials will be presented by recognized sound system designers and acoustics specialists. A one-track scheduling format is planned. However, it is expected that parallel tracks for various hands-on sessions and demonstrations as well as theoretical sessions can be organized to suit individual preferences.

September 1–6, 2017
IFA Berlin (The International Funkausstellung)
Berlin ExpoCenter City, Messedamm 22, Berlin, Germany
www.ifa-berlin.com

IFA Berlin (The International Funkausstellung) is probably the most important annual event for the consumer electronics industry. If the CES in Las Vegas, N.V., is the place were future trends and innovation are highlighted and get the most attention, the IFA Berlin show is where many of those products and brands make it or break it. Audio at IFA is mainly focused on personal/mobile devices and the more accessible and lifestyle-oriented products. A wide range of products, solutions, and presentations make a visit to IFA 2017 worthwhile. IFA 2016 featured more than 1,800 exhibitors and attracted 240,000 visitors, with the majority of trade visitors coming from outside Germany: 68,000. The 2016 floor space was already the largest in the show's history. Expect the 2017 edition to be even larger!

Article Updates

Our June 2017 *audioXpress* issue featured an article by Michael Kiwanuka titled "Current Feedback and Voltage Feedback Fallacies." Unfortunately, the article contained an incorrect image for Figure 2. The correct image and caption are provided here.

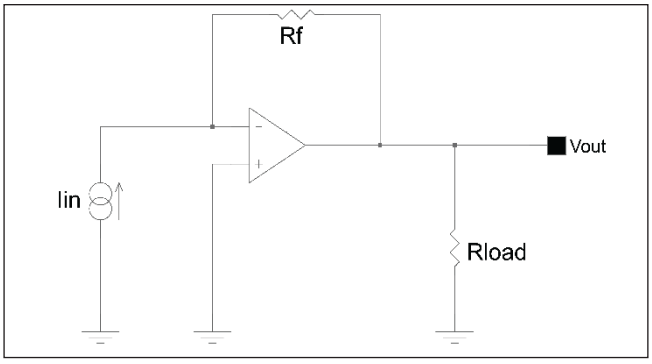


Figure 2: Transadmittance feedback gives rise to a transimpedance amplifier or a current-controlled voltage source (CCVS).

Our July 2017 *audioXpress* issue featured an article by James Lin titled, "Improving the Stax SRM-T1 Electrostatic Headphone Amplifier." After the article was published, Lin was contacted by one of his acquaintances who is knowledgeable about the Stax amps. He pointed out a simple safety modification for the T1, which we would like to share with readers. The modification is adding a 5 mΩ resistor to the bias supply lines after the last capacitor. This should be done for both the normal bias and pro bias, and can be done by desoldering the blue and gray wires that run from the circuit board just in front of the transformer to the output sockets, and interposing the resistors. Mouser has a suitable part: 594-5043DM5M100J, which is a 5M1 0.25 W metal film. Another suitable part from Mouser is: 594-VR37000004704JR5, which is a 0.5 W 4M7 resistor, rated for 3.5 kV.

Engineering Consulting for the Audio Community

Celebrating 34 YEARS 1983-2017

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

MAKING THE WORLD SOUND BETTER

FOR OVER 30 YEARS



Bringing the world of audio to your front door, Parts Express stocks the finest yet most affordable components and tools required to complete your next design. Whether you are a hobbyist or a large manufacturer, we truly have everything but the music for all your audio needs! **Visit parts-express.com today!**

Save $10 on orders $100+!
Use promo code AXP8AQ at checkout.
**Offer expires 9/30/17*

PARTS EXPRESS
YOUR ELECTRONICS CONNECTION
725 Pleasant Valley Dr. Springboro, OH 45066

Cabinets | Drivers | Amplifiers | Components | Measurement