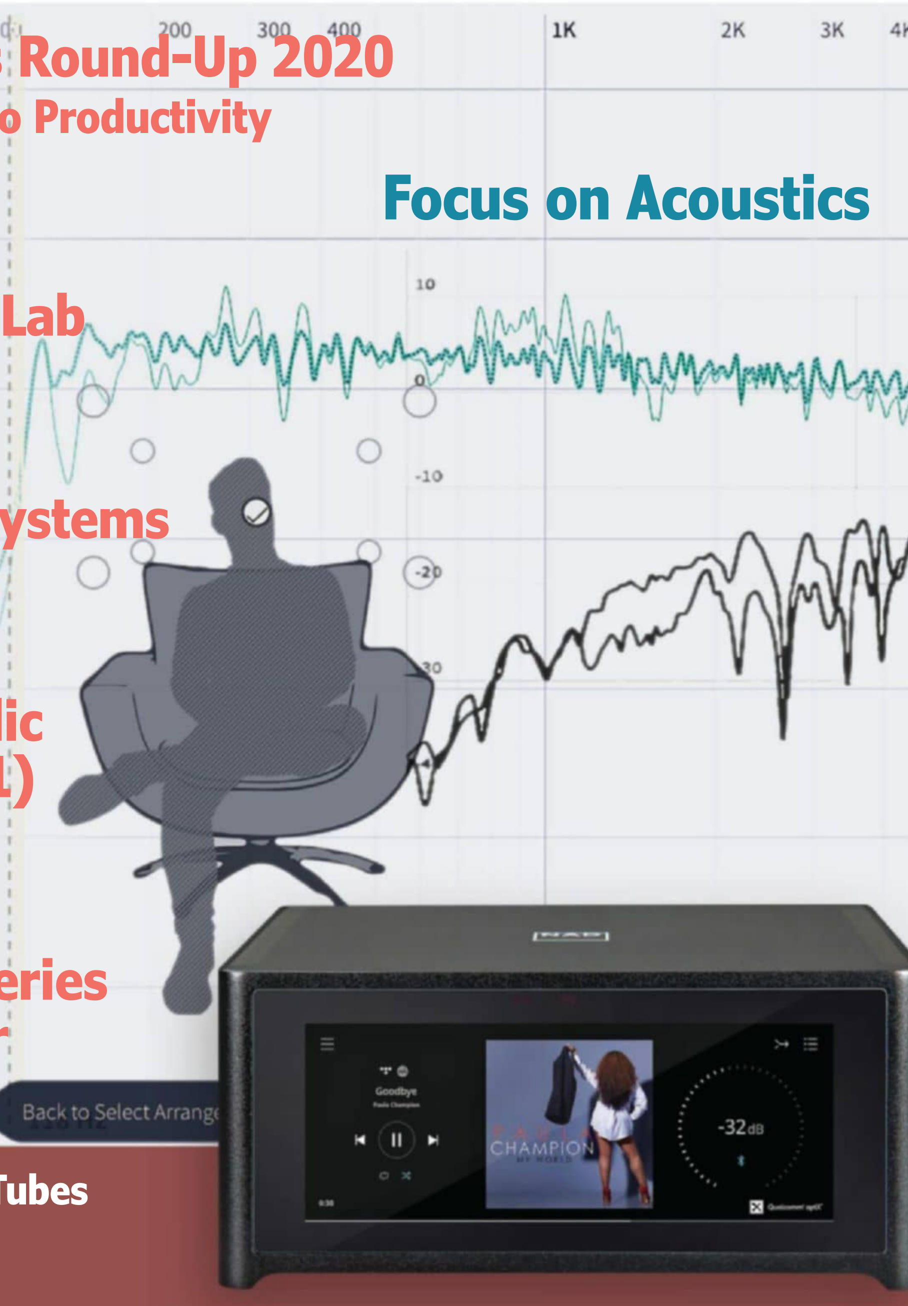
INNOVATIONS IN AUDIO • AUDIO ELECTRONICS • THE BEST IN DIY AUDIO

www.audioxpress.com

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY



Acoustical Solutions Round-Up 2020
From Acoustic Comfort to Productivity
By J. Martins

Focus on Acoustics

Acoustic Praxis
The Elephant in the Lab
By Jim DeGrandis

Sound Control
Testing Acoustical Systems
By Richard Honeycutt

You Can DIY!
The APR-17 Aperiodic Loudspeaker (Part 1)
By George Ntanavaras

Fresh From the Bench
NAD M10 Masters Series Streaming Amplifier
By Stuart Yaniger

Audio Electronics—Tubes
The Internal Life of Vacuum Tubes
By Jan Didden

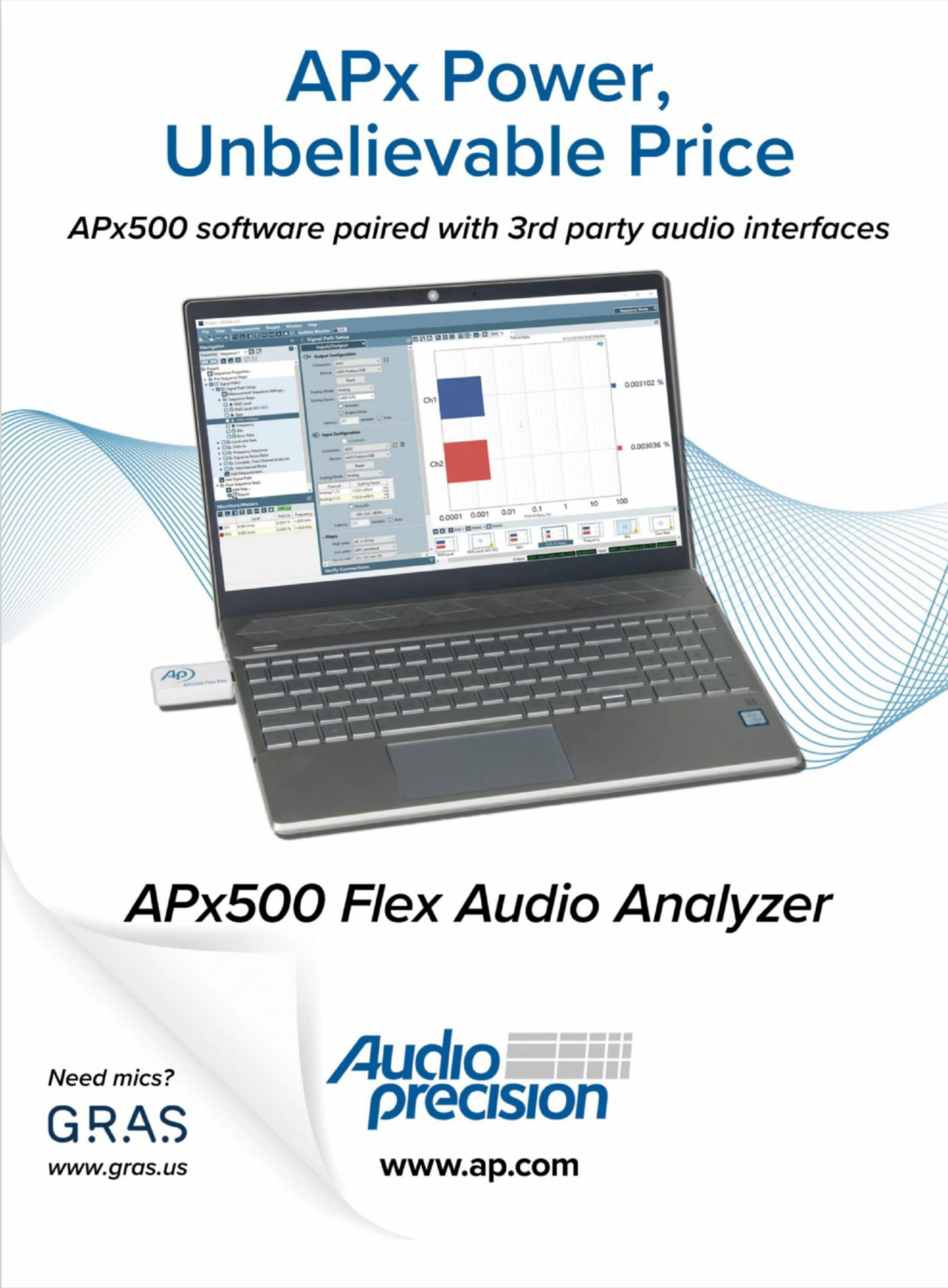
Hollow-State Electronics
The Early Development of Hollow-State Mixers
Richard Honeycutt

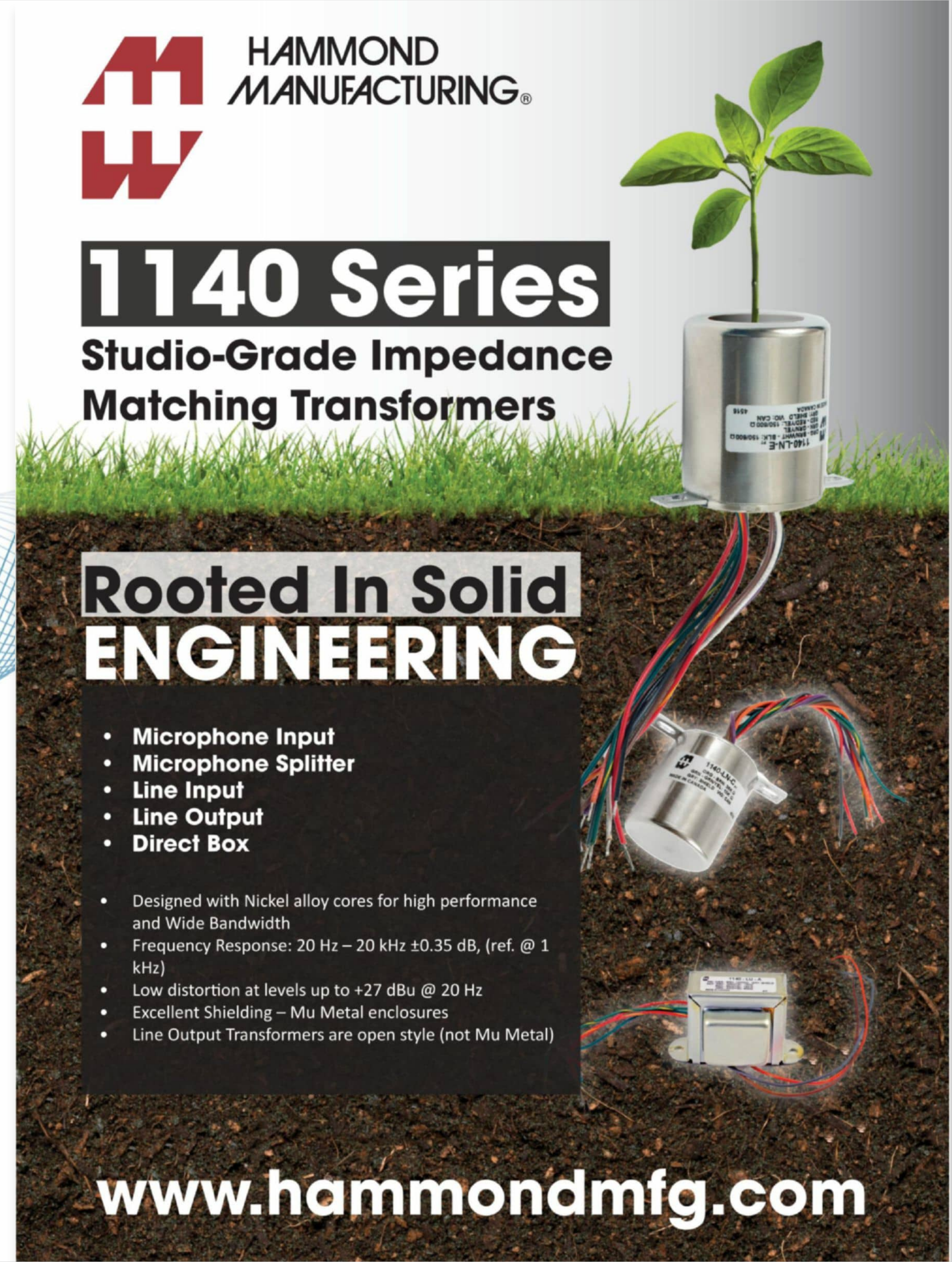
Recording Review
Sound Liaison's DXD Music Sample
By Gary Galo

$9.00US $10.00CAN

08

AUGUST2020


APx Power,
Unbelievable Price
APx500 software paired with 3rd party audio interfaces
APx500 Flex Audio Analyzer
Need mics?
GRAS
www.gras.us
Audio precision
www.ap.com


HAMMOND MANUFACTURING®
1140 Series
Studio-Grade Impedance Matching Transformers
Rooted In Solid
ENGINEERING
• Microphone Input
• Microphone Splitter
• Line Input
• Line Output
• Direct Box
• Designed with Nickel alloy cores for high performance and Wide Bandwidth
• Frequency Response: 20 Hz – 20 kHz ±0.35 dB, (ref. @ 1 kHz)
• Low distortion at levels up to +27 dBu @ 20 Hz
• Excellent Shielding – Mu Metal enclosures
• Line Output Transformers are open style (not Mu Metal)
www.hammondmfg.com

August 2020 ISSN 1548-0628

www.audioxpress.com

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

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

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

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

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

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

Home Acoustics, New Workspaces

This edition of *audioXpress* focuses on acoustics and once again we compiled a series of suggestions of products and solutions to help improve any acoustical environment—which under the current circumstances of a global pandemic, starts with our own households. To reinforce that perspective, in our traditional Acoustic Solutions Round-Up feature, we've added a few suggestions about basic solutions for everyone to use at home or in a personal workspace.

The Acoustic Focus for this edition is reinforced with a timely article by Jim DeGrandis discussing the important challenges still faced by acoustics professionals when specifying acoustic materials. Specifically, his article discusses the need to update basic lab measurements and test data to improve the accuracy of the advanced design, modeling, and simulation tools that exist today.

In another example on how acoustic tools are evolving, Stuart Yaniger reviews the M10 Masters Series Streaming Amplifier from NAD Electronics, a compact and powerful home audio amplifier and network streamer that surprises positively for its integration of the latest Dirac Live room correction technology with Bass Control. There's still a lot of useless debate about the merits of digital signal processing for room correction versus room treatment. The reality is both disciplines have tremendously evolved, but digital room correction is now on a completely new level that needs to be acknowledged as extremely valuable and, in certain circumstances, essential.

During the first half of 2020, we were glad to see that most acoustical solutions companies have demonstrated strong resilience through the shelter-at-homes and confinement periods. Manufacturers of first-approach acoustical products managed to keep production levels and distribution steady, and quickly adjusted, as the demand for acoustical products changed. According to several of the companies we contacted, as people were forced to focus on different issues and investments in large community spaces were temporarily suspended, the demand for specific first-approach acoustical solutions, including room kits, microphone isolation shields and reflection filters, floor mats, and insulation solutions in general, increased five-fold.

Most acoustic panels companies have launched special kits for conference call applications at home, and many professionals have been active writing articles about the best approach to home acoustic challenges. In general, companies have been promoting easy-to-ship and install solutions.

Not always was this increased demand completely fulfilled, so the demand is still there. Logistical problems, the inability of online e-commerce platforms to support the complexity and diversity of acoustical products, led many manufacturers to realize the need to either invest in direct-to-consumer online stores, or combine an improved online presence with expanded support to distribution partners—which, in general, is the better long-term solution for acoustical product companies that don't want to be limited to serving their hometown or states.

Many of those manufacturers also realized the need to rethink their way of offering services, and refocus their R&D efforts on first-approach and easy-to-install solutions, therefore being able to better serve the market in a time when custom installs and built-to-measure solutions are restricted.

No doubt, these are challenging times for everyone, but as always, there are many opportunities available. After experimenting with having its employees working from home, many companies realized this can be a great sustainable model for their specific businesses. It's not for all, but many companies will allow employees to work from home indefinitely or part-time. This will increase the need to create proper working conditions, including a correct acoustic environment.

Long term, a lot of the needs that existed for large commercial spaces will transition to more manageable facilities, where people can safely do their work.

J. Martins
Editor-in-Chief

The Team

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

Technical Editor: Jan Didden

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

OUR NETWORK

audioxpress circuit cellar LIS Loudspeaker Industry Sourcebook VOICE COIL

SUPPORTING COMPANIES

ACO Pacific, Inc.	59	Hypex Electronics B.V.	49
Acoustic Geometry	45	JLI Electronics, Inc.	53
Acoustic Sciences Corp.	44	KAB Electro-Acoustics	51
Acoustics First Corp.	44	Menlo Scientific, Ltd.	23
All Electronics Corp.	51	Odeon Room Acoustics Software	41
Audio Precision, Inc.	2	Parts Express International, Inc.	68
COMSOL, Inc.	9	PCB Piezotronics, Inc.	19
GRAS Sound & Vibration	13	Redco Audio	61
Hammond Manufacturing, Ltd.	3	Sonitus USA	15
HEAD acoustics, Inc.	43	Triad Magnetics	67

NOT A SUPPORTING COMPANY YET?

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

COLUMNISTS

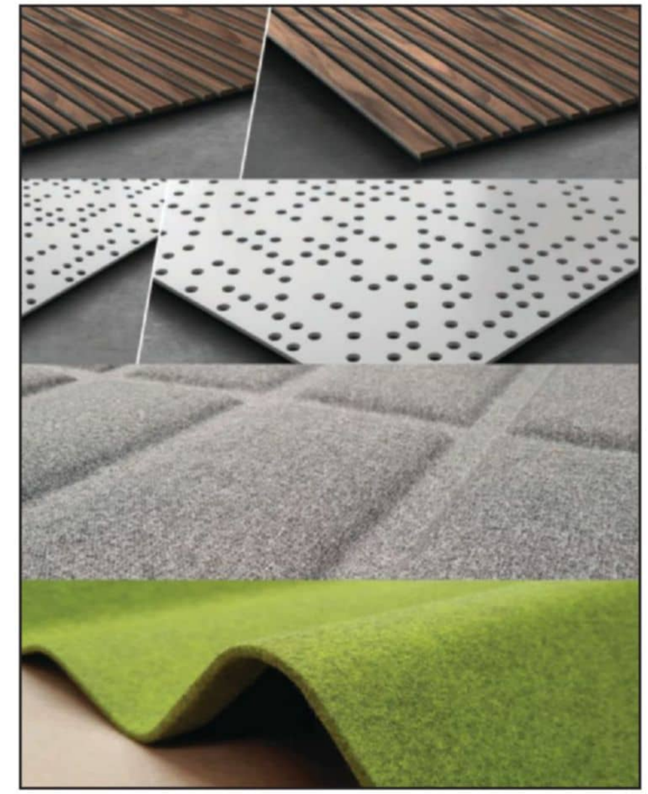
Vance Dickason has been working as a professional in the loudspeaker industry since 1974. He is the author of *Loudspeaker Design Cookbook*—now in its seventh edition and published in English, French, German, Dutch, Italian, Spanish, and Portuguese—and *The Loudspeaker Recipes*. Vance is the editor of *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.

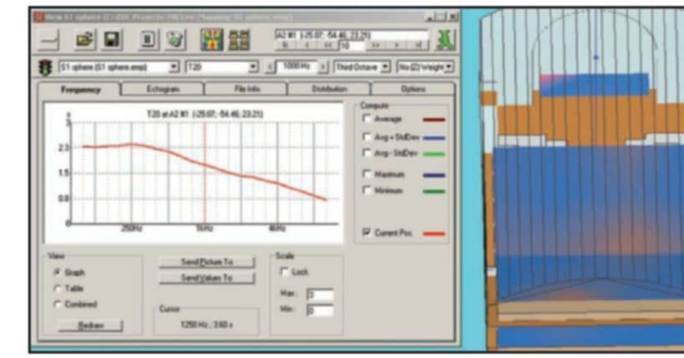
Dr. Richard Honeycutt fell in love with acoustics when his father brought home a copy of Leo Beranek's landmark text on the subject when Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical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

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

Michael Steffes has worked in high-speed signal path IC development, spanning IC design roles for the BurrBrown high-speed amplifier products, and product launch/application manager roles through Texas Instruments/Intersil. He continues to focus on high-performance ADC interface solutions, DSL/PLC line driver design and optimization, and signal path design optimization tools. In a career spanning five different major IC vendors, he has participated in the introduction of 150 industry leading amplifier products (setting many of the now standard practices in high speed amplifier characterization curves), publishing more than 100 contributed articles and application notes from 1989 onward.

Volume 51 – No. 8 August 2020





Features

Acoustics Special Features

Columns

Departments

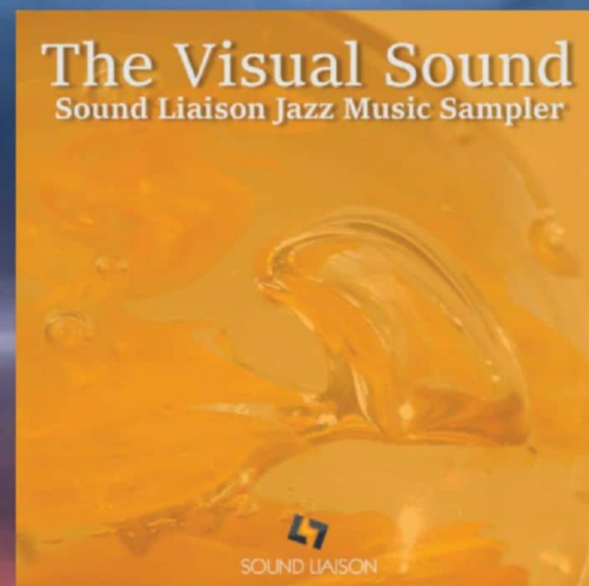
Websites

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

@audioxp_editor audioxpresscommunity
linkedin.com/company/audioxpress

Taking a Listen to Sound Liaison's DXD Music Sampler

Gary Galo explores the recording methods of Sound Liaison, a small audiophile label based in The Netherlands, and discusses whether the company's DXD sampler, The Visual Sound, actually reflects its stated goal of bridging the gap between the studio and the listeners.



By
Gary Galo
(United States)

Sound Liaison is a small audiophile label based in The Netherlands. The company name reflects its stated goal of building "a bridge (liaison) between the studio (engineer and musicians) and the people who love to listen to beautiful music using high-quality audio equipment." Sound Liaison's high-resolution recordings are made during studio recording sessions as well as special live events with an audience, held in its professional studio.

The company was founded by Frans de Rond and Peter Bjørnild. Bjørnild is a professional double-bass player with a background in both jazz and classical music. A native of Denmark, his musical education includes studies at Royal Danish Conservatory of Music as well as the Royal Conservatory (Koninklijk Conservatorium) in the Hague. De Rond, the engineer for all these recordings, is also a double-bass player, with an education in both recording technology and music from the Royal Conservatory. De Rond's goal is to "create an almost visual sound field" that comes "as close to the natural, organic sound of the instrument as possible." Bjørnild "believes that a recording should be as realistic and beautiful sounding as possible. As if, when closing your eyes, you find yourself in the best seat of the hall."

De Rond notes that until recently his approach to recording was similar to that of most other engineers. He employed multi-microphone techniques, using one or more microphones per instrument, attempting to get the greatest separation by placing the performers in separate acoustical baffles. This method, used along with multi-track recording, allows a great deal of flexibility after the session, because the engineer has complete control over the balances and the stereo perspective long after the musicians have left the studio.

The problem de Rond noted was that the musicians don't blend very well and don't sound as though they are performing in the same acoustical space. Recording with many microphones also introduces phase artifacts. To avoid these problems, he did an about-face and decided to make a number of recordings with a single stereo microphone. It can sometimes take time to find the right spot

The Visual Sound—Sound Liaison's DXD Music Sampler
Sound Liaison
Laan 1940 1945 29
1222 NG Hilversum
Netherlands
info@soundliaison.com
www.soundliaison.com

Photo 1: MCO Studio 2 in the Dutch Broadcasting Music Center is the venue for recordings made by Sound Liaison. The room is the oldest radio studio in the Netherlands. (Photo courtesy of Sound Liaison)

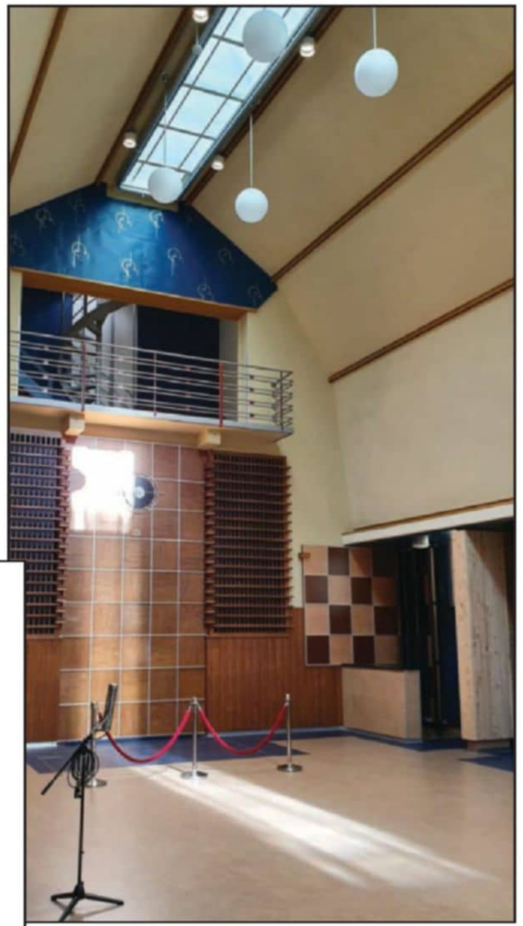


Photo 2: This is the Josephson C700S stereo microphone. This unusual stereo mic contains three condenser capsules and can be used for a variety of single-point stereo techniques, including Blumlein, X-Y, and M-S. (Photo courtesy of Josephson Engineering)

for a single-point stereo microphone and, particularly when recording a live event, there are no second chances. But, when you get it right, as de Rond notes, the result is "phase coherence, perfect imaging, great sense of depth, and superior realism." Musicians have been most enthusiastic about his efforts. He quotes bass player Clemens van der Feen as saying "I never heard my bass like this before on any recording. Finally it sounds the way I always wanted."

Making purist recordings depends not only on the skill and ears of the engineer, but also on the acoustics of the recording venue, and the microphone. The venue chosen by Sound Liaison is MCO Studio 2, part of the Dutch Broadcasting Music Center (Muziek Centrum van de Omroep) as shown in **Photo 1**.

According to the MCO website, Studio 2 is the oldest radio studio in the Netherlands. It's a large room with a floor area of 1442 ft^2 (134 m^2) and a high, vaulted ceiling. The large floor plan can accommodate 80 audience members, making it suitable for concerts and recording sessions. The studio is equipped with its own, adjacent control room and a Steinway Model B grand piano. The studio has a distinguished history, hosting many famous jazz artists over the decades, including Django Reinhardt, Ray Brown, Ella Fitzgerald, Wes Montgomery, Clark Terry, and Eric Dolphyi. De Rond notes, "I like to record the musicians in one big good sounding hall. The unique sound of Studio 2 in the MCO building here in Hilversum is ideal for that purpose."

A recording engineer looking for a single-point stereo microphone has many choices—Sound Liaison uses the Josephson C700S (see **Photo 2**). Josephson Engineering is a familiar name to readers of this magazine. Founded in 1988, the Santa Cruz, CA company started out designing and manufacturing measurement microphones. Its C550 is one of the most respected microphones available for acoustical and loudspeaker measurements.

In more recent years, Josephson has expanded into the competitive field of recording microphones, offering seven series of mics for studio use. There are two microphones in the C700 series. The C700A is a mono microphone incorporating 16 mm omnidirectional (pressure) and 26 mm figure-8 condenser (gradient) capsules. By controlling the mix of the pressure and gradient signals, any pattern from omni to figure-8 can be derived. The internal preamp is designed around a cascode FET input and a Class-A active balanced output circuit. The C700S adds a third, side-facing figure-8 capsule to the C700A. By combining the three signals, any coincident stereo pickup can be created, including Blumlein, X-Y, and M-S. The stereo image can be created on-the-fly, during actual recording, or the three signals can be recorded separately and mixed later, giving the engineer flexibility in production.

Sound Liaison doesn't sell physical media—it offers downloads in both pulse-code modulation (PCM) and direct stream digital (DSD) formats. All its recording is done in DXD PCM format (DXD stands for Digital eXtended Definition).

The DXD process was pioneered by Merging Technologies and involves recording at sampling rates higher than those used for conventional high-resolution PCM—either 352.8 kHz or 386 kHz, 24-bit (twice 176.4 or 192). Sound Liaison uses 352.8 kHz, probably because the most transparent conversion from PCM to DSD is from PCM files that are multiples of 44.1 kHz. Downloading the 352.8 kHz files provides audiophiles with a clone of their own studio masters but, as the company notes, its DSD files are not studio masters. These are conversions from the DXD PCM files. There's one exception—the album *Impromtu*, featuring bassist Tony Overwater and pianist Bert van den Brink, was recorded simultaneously to PCM and DSD using two separate recorders. For those who do not have equipment that can handle the ultra-high-res DXD files, Sound Liaison also offers PCM conversions at 192 kHz and 96 kHz.

The Visual Sound Recordings

The Visual Sound is a sampler that includes 10 small-group jazz selections from Sound Liaison's DXD albums, all mastered at 352.8 kHz. For review, I received the 10 selections in native DXD resolution as .wav files, down-sampled 96 kHz/24-bit FLAC files, and conversions to DSD 64, which is the base DSD resolution (the same as SACDs). Prices range from 9.75 to 19.75 Euros, depending on the resolution. Six of the recordings were made using multiple microphones, while the other four were made with a single Josephson C700S. The lists and photos of recording equipment used by Sound Liaison reflects its perfectionist approach, including Merging Technologies' Horus microphone preamps, A/D converters and master clocks, plus AudioQuest microphone cables, power cables, and power line conditioners. Notes for the musical selections are on the Sound Liaison website.

My main digital player is an OPPO Digital UDP-205 (reviewed in *audioXpress*, November 2017). I copy media files to an external, portable hard drive connected to a USB 3.0 port on the OPPO with an AudioQuest Carbon cable. The OPPO's HDMI audio output feeds a Monoprice HDMI audio converter with a Pangea Premiere SE HDMI interconnect. The Monoprice feeds full PCM audio resolution via S/PDIF to my Benchmark DAC3 HGC with a D.H. Labs D-75 cable fitted with Canare 75 Ω RCA connectors.

My preamp is a Benchmark HPA4 feeding a pair of Benchmark AHB2 power amps. The power amps passively bi-amp my Audio Concepts Sapphire 3/Sub 1 loudspeaker systems. I use D.H. Labs Air Matrix balanced analog interconnects and D.H. Labs Q-10 loudspeaker cables. The 352.8 kHz files were used for most of my listening. The OPPO player down-samples them to 176.4 kHz, verified by the display on the Benchmark DAC3. I played the DSD64 files on my Samsung Galaxy Note 4 phone using the USB Audio Player Pro app. The Note 4 feeds the USB input to the Benchmark DAC3 directly, using the DoP (DSD over PCM) protocol. The interconnect is a D.H. Labs Mirage USB and a RadioShack USB On-the-Go adapter.

One of the best single-mic recordings on this sampler is "Ou Es-Tu, Mon Amour?" performed by Reinier Voet & Pigalle 44, from their album *Ballade pour la nuit*. Voet is a jazz guitarist and the collaborating group consists of Karin van Kooten on violin, Jan Brouwer on rhythm guitar, and double-bassist Jet Stevens. The recording was made with a live audience in the studio, and the session photo shows the placement of the microphone and the group (see **Photo 3**).

Photo 3: This is guitarist Reinier Voet (left) and the group Pigalle 44 in concert in MCO Studio 2, during the live recording of *Ballade pour la nuit*. (Photo courtesy of Sound Liaison)

The aural image is extremely realistic, with perfect balances and a natural, but not exaggerated, stereo image. The solo violin has moved center stage for her solos, but the rest of the groups sounds exactly as they appear in Photo 3. The performers sound as though they are in the same acoustic space, and the placement of the C700S microphone captured just the right amount of hall ambience. The double-bassist uses his own pickup and amplifier, but the undoctored recording shows his own balance with the rest of the group to be spot on. These are accomplished musicians—the solos by Voet and van Kooten noteworthy for their taste and imagination, rather than virtuosic display, matched by the support they receive from Brouwer and Stevens.

I became a fan of the Dave Brubeck Quartet when I was a high school student, around 1966. The Columbia album that made the group famous,

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. Galo 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. Galo 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 United States*. 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.

Time Out, had been released in 1959, and was still considered ground-breaking because of the use of time signatures largely unheard of in jazz, plus a harmonic language that reflected the group's knowledge of 20th-century classical music. In his 1956 Columbia album *What is Jazz*, Leonard Bernstein noted that modern jazz musicians know Béla Bartók and Igor Stravinsky, and wear horn-rimmed glasses and Ivy League clothes (Sony CD SMK 60566, out of print). He was talking about Dave Brubeck, of course!

The four musicians in the Feen Brothers jazz quartet cover the same instrumentation as the vintage Brubeck Quartet: Paul van der Feen on saxophone, Mark van der Feen on piano, double-bassist Clemens van der Feen, and drummer Matthijs van der Feen. "The Duke," featured on this sampler, was Brubeck's tribute to Darius Milhaud, the 20th-century French composer and his teacher at Mills College in Oakland, CA.

Session photos show that the recording was made with the group gathered around a single C700S microphone, which provides a striking contrast to the multi-mic approach normally given to the Brubeck Quartet by Columbia's engineers. There's nothing spectacular here—just a natural balance and a real sense of the acoustic space, and a recording that captures the influence of Paul Desmond's unique, somewhat breathy, alto sax sound on Paul van der Feen's playing.

Photo 4: This is the Paul Berner band during the live recording of their album *The Bird has Flown*. The widely-spaced main stereo pair can be seen in front of the group, at the edge of the carpet. (Photo courtesy of Sound Liaison)

The Feen Brother take "The Duke" at the rather leisurely tempo chosen by the Brubeck Quartet on their original 1956 mono album Brubeck Plays Brubeck, rather than the considerably faster one they took for their live stereo remake recorded at the Newport Jazz Festival in 1958. Strangely, the notes for this selection discuss "It's a Raggy Waltz" and "Unsquare Dance" (which are not on the sampler) but make no mention of "The Duke."

Vocalist Carmen Gomes is featured in selections from two of her albums, one made with a single C700S microphone, and the other one with multiple microphones. Gomes calls her group Carmen Gomes Inc., and she is supported by Folker Tettero on guitar, Peter Bjørnild on double-bass, and Bert Kamsteeg on drums. "Summertime" from Gershwin's *Porgy and Bess* was made under studio conditions using the single stereo microphone, and is taken from their album *Don't You Cry*. Getting the right balance between a vocalist and an instrumental group can be difficult with a single-point pickup, but Frans de Rond succeeded admirably on this recording.

According to the notes, Gomes wanted a retake because she felt she had become too emotional near the end of the song. De Rond would have none of it and refused to do another take; he was right! "Summertime" crossfades directly into "Losing Hand" from their album *Carmen Gomes Sings the Blues*, all tunes that Harry Belafonte had recorded in 1958 for his album *Belafonte Sings the Blues*.

This album was recorded with a live audience using a total of eight microphones, but by this point de Rond had dispensed with the isolation baffles he once used. The main pickup was a pair of Schoeps MK5 condenser mics in an A-B (spaced) configuration. The remaining six were accent mics for each member of the group, including three on the drum set. The recording is a testament to de Rond's skill in doing a successful live mix. The main stereo pair ensures that the sense of acoustic space and depth are not lost, with the accent mics adding subtle reinforcement. By comparison, the group's single mic recording places the performers in the same acoustic space, whereas in this one the listener is slightly aware of the highlighting.

Pianist Juraj Stanik is featured in "Fool on the Hill" from his album *I Wonder*. The single C700S microphone captures a very realistic sound from the house Steinway Model B piano, with the various registers of the instrument moving subtly in the soundstage. The Steinway B is a 7′ instrument, which does not have quite the richness of tone or sustain qualities as their full-size 9′ concert grand, the Model D. The Josephson microphone and its skillful placement leave no doubt as to which instrument is in use, though the house piano is obviously maintained by an excellent technician.

The remaining five recordings were all made with multiple microphones. As with the Carmen Gomes, Inc.'s recording of "Losing Hand," the key to de Rond's success is his consistent use of a pair of main microphones in a spaced A-B configuration, always Schoeps MK5s. The MK5 is a two-pattern capsule, switchable between cardioid and omnidirectional. Sound Liaison's documentation doesn't specify which pattern was used, but there's enough low-coloration room sound that I would bet that they were used as omnis. The room sound is especially beneficial for Folker Tettero's quartet playing "Cold Duck Time." Tettero is a guitarist and the other three players include Dirk Beets on trumpet and flugelhorn, Dave Breidenbach on electric bass guitar, and Klaas van Donkersgoed on drum set.

Given the rock influence, many engineers would simply have recorded this live event with a completely manufactured soundstage, but de Rond's tasteful multi-miking, accenting the main stereo pickup, gives a plausible sense of the positions of the players as heard by the audience. The Paul Berner band are heard in "It's Only Love" from their album *This Bird has Flown*, which is an instrumental tribute to John Lennon and Paul McCartney.

Berner plays acoustic bass, supporting Michael Moore on clarinet and alto sax, and electric and acoustic guitarists Ed Verhoeff and Peter Tiehuis. The notes indicate that Verhoeff and Tiehuis are placed left and right in the soundstage. The recording captures the increase in soundstage width as the music become progressively more complex (see **Photo 4**).

Photo 5: Here the Whitmer trio records their album *Free—Soulful Piano Reflections*. For this multi-mic recording, engineer de Rond spaced the stereo A-B pair rather close. (Photo courtesy of Sound Liaison)

The Witmer Trio members include Cajan Witmer on piano, Han Slinger on bass, and Maarten Kruijswijk on drums; on their album *Free—Soulful Piano Reflections*, they're joined by violinist Vera van der Bie. For this recording, de Rond spaced his Schoeps A-B pair closer together than on some of the other multi-mic recordings (see **Photo 5**).

"Ain't No Sunshine" is most impressive for the sense of depth de Rond achieves on the drummer. In addition to front-to-back placement, the recording really captures the acoustic surrounding the various instruments in the drum set, all superbly balanced with the rest of the ensemble. Witmer is obviously well-versed in classical music; he quotes the legendary Romanian conductor Sergiu Celibidache's comments on how tempo and dynamics are relative, and depend on the sound of the hall. For these arrangements, Witmer carefully considered the sound of this group in Studio 2. Judging from this selection, he has been very successful.

The REF Trio are heard in "Born to be Blue" from their album of the same name. The group consists of Rob van Kreeveld on piano, Edwin Corzilius on double bass, and Frits Landesbergen on vibraphone. In his note about the recording, de Rond says that he first got the stereo image and depth using the main Schoeps MK5 pair. He then tried to place the accent mics in the same positions. The result is a very natural sound, with the spot mics adding the necessary clarity without drawing attention to themselves. "Basin Street Blues" was written by Spencer Williams in 1928 and made famous by Louis Armstrong. This tune is from the album *Wild Man Blues—Ruud Breuls & Simon Ritger Quintet Play Louis Armstrong*.

Bruels plays trumpet and flugelhorn in this group, with Simon Rigter on tenor saxophone, Karel Boehlee on piano, Jos Machtel on double bass, and Marcel Serierse on drums. This is the most complex multi-mic recording in this sampler. In addition to the Schoeps MK5 main pair, seven accent mics were used, including three on the drum set and two on the piano, all skillfully mixed and balanced with the main pair to create a very natural soundstage. The stereo image successfully captures the relative positions of the saxophone and the trumpet solos, with and without a mute. The sax is up front, the trumpet just behind, and the muted trumpet further back. Bruels captures at least some of the unique Armstrong sound, though he's perhaps a bit too polite!

The Verdict

There are some audio enthusiasts who claim that high-resolution digital recordings offer no audible improvement over conventional Red Book CDs. I could not disagree more, but what about 96 kHz vs. DXD at 352.8 kHz? I found the 96 kHz files to be excellent, but by comparison the 352.8 kHz originals have a bit more air and space in the sound, even with my OPPO player down-sampling them to 176.4 kHz. The DSD comparisons are interesting. I found the DSD files to be a bit "warmer" but I could not help wondering if there's some sort of euphonic coloration going in the conversion process. At times, I felt that the DSD files had a slightly larger and more precise stereo image. That may seem strange, but I can only report what I heard. It's possible that my reactions to the warmth and the imaging are due to the simpler analog filtering in DSD playback.

Sound Liason's DXD sampler, The Visual Sound, offers some of the most realistic sounding recordings I've heard, in enjoyable selections performed by excellent musicians. Most of the music on this sampler is decidedly laid-back, and audiophiles looking for an in-your-face sonic spectacular will likely miss the point. For engineer Frans de Rond and the excellent musicians heard on these recordings, it's all about subtlety and refinement. Musically perceptive listeners will appreciate the efforts of all involved in making these fine recordings.

Resource

Dutch Broadcasting Music Center,
www.mcogebouw.nl/verhuur/beschikbare-ruimtes/studio-2

Sources

Benchmark Media Systems | https://benchmarkmedia.com

D.H. Labs Cables | https://silversonic.com

Josephson Microphones | www.josephson.com/c700a.html

Monoprice HDMI Audio Converter | www.monoprice.com/product?p_id=10251

USB Audio Player Pro |
www.extreamsd.com/index.php/products/usb-audio-player-pro

The Visual Sound (Contents)

1. It's Only Love – Paul Berner Band (6:04)
2. Ou Es-Tu, Mon Amour? – Reinier Voet & Pigalle44 (4:15)
3. Summertime – Carmen Gomes Inc. (5:46)
4. Losing Hand – Carmen Gomes Inc. (5:25)
5. Ain't No Sunshine – Witmer Trio (4:51)
6. The Duke – Feenbrothers (4:38)
7. Born To Be Blue – REF Trio (5:32)
8. Basin Street Blues – Ruud Breuls & Simon Rigter Quintet (7:05)
9. Fool On The Hill – Juraj Stanik (1:57)
10. Cold Duck Time – Tettero (5:23)

Total time: 50:56
Catalog Number: SL-1036A

Fresh From the Bench

NAD M10 Masters Series Streaming Amplifier

Stuart Yaniger reviews the compact but mighty M10 Masters Series Streaming Amplifier from NAD Electronics, which combines performance, an attainable price, and powerful features including a network streamer and the latest Dirac Live room correction with Bass Control.

Photo 1: The NAD M10 Masters Series Streaming Amplifier is a surprisingly small and slick package with a ton of functionality.

By
Stuart Yaniger
(United States)
Photos by Cynthia Wenslow

Much of the joy that the hardware side of audio has given me recently is, ironically, the commodification of excellent performance. By this I mean that top-quality sound is available at attainable prices, and that features, form factor, and manufacturer reputation can now be the determining factor on the electronics end, with high electrical performance almost a given. The revolution on the loudspeaker side has been just as profound, with powered speakers using DSP and featuring a variety of inputs and controls now costing about the same as a couple of concert tickets in the nosebleed seats.

The other end of commodification is the reduction of the sheer amount of physical equipment needed for music reproduction, both for the hardware and the source material. For people who thrill to swapping preamps and cables and power amps and phono cartridges, this is anathema. But if, like me, you want your music without fuss and the sound to be great, we live in the best of times. It's easier and cheaper than ever to achieve that goal.

An Introduction to the NAD M10

Which brings me to the NAD M10 from the company's Masters Series. The article title of "Streaming Amplifier" doesn't do the integration justice: the M10 contains a network streamer, preamplification with analog and digital inputs, a pair of high-quality 100 W Class-D power amps (nCore), a subwoofer crossover (stereo, yet!), equalization, and best yet, Dirac Live room correction software. All of this in a sleek black box with a footprint smaller than an issue of *audioXpress*.

My description here will be longer than usual because, well, there's a lot to describe. Physically, this is the proverbial black box, with no physical controls other than a black Gorilla Glass touchscreen on the front (see **Photo 1**). And when I say "no controls," that includes remote control—there is none included. Instead, Bluesound's BlueOS app can be loaded onto your phone and be used to control the M10, a plus toward removing the annoyance of remote control clutter (always an issue chez nous) and the tragedy of lost remotes. People who like the feel of large rotating knobs and clicking switches will not be happy, but this is just the kind of user interface I like. If you really want to use a dedicated control, the M10 is programmable for using standard IR universal remotes.

BluOS is also meant to integrate the M10 into a home network so that the streaming capabilities can be used. The M10/BluOS is capable of addressing most of the best-known services like Qobuz, Pandora, Amazon Music HD, Spotify, Tidal, and many more, where it can stream CD-quality, HD (up to 24/192), and MQA sources. The TuneIn app is also integrated for streaming Internet radio and podcasts. The M10 can also use Apple AirPlay 2 streaming, and be synced with smart speakers (e.g., Echo and Google Nest).

The back panel (see **Photo 2**) includes USB, Ethernet, SPDIF coax and optical, and analog inputs, as well as line-level outputs for mono or stereo subwoofers. There are also preamp outputs in the event you want to use a larger external power amplifier—there's no corresponding power amp inputs, so if this option is chosen, the internal power amps can't be used for subwoofers or surround speakers. The main speaker outputs are four binding posts (unfortunately, spaced slightly too far apart for stereo banana plugs, which seems to be the custom these days), which had no trouble accommodating the 10-gauge speaker wire that I use nor the banana plugs from my dummy loads. The analog inputs and outputs are all unbalanced.

It should be noted that the M10 does not have a dedicated headphone output. The preamp outputs could, in theory, drive a dedicated headphone amp, but the main speaker outs will not be muted. That seems like an easy function to incorporate, and I hope NAD does this in future versions. The Bluetooth function can be used to drive Bluetooth headphones, which will be fine for some listeners, just not me or hardcore headphone users.

Moving on to the control panel for the M10, the touchscreen on the unit enables the user to control volume, source, balance, equalization (tone controls and Dirac Live), subwoofer configuration and crossover frequency (if selected), and display appearance (brightness, album art, VU meters, etc.). All of these and more are controllable using the BluOS app, of course, but in some instances, control at the amp itself might be useful.

One neat gimmick is the ability to look at the temperature of the power amp output devices. This seemed pretty stable, in the low 60°C range at idle, rising to just below 70° after a few hours of loud music. I'm not sure what actionable information the user can get, but nonetheless, it's interesting in a geeky sort of way.

Perhaps more useful is the line-level subwoofer output—although slopes and levels are not adjustable, the frequency can be varied between 40 Hz to 200 Hz using the BluOS app control panel (see **Figure 1**). The slopes are not specified and are not variable, nor is there a phase or polarity control. The assumption is that most powered subwoofers will have variable gain and phase functions built into them, so there's no need to duplicate these functions. Some users have noted that the subwoofer output levels are low, so it's a good idea to verify that your active subwoofer(s) have sufficient input sensitivity.

Photo 2: The M10's back panel contains all of the I/O functions, which are clearly labeled and easy to use.

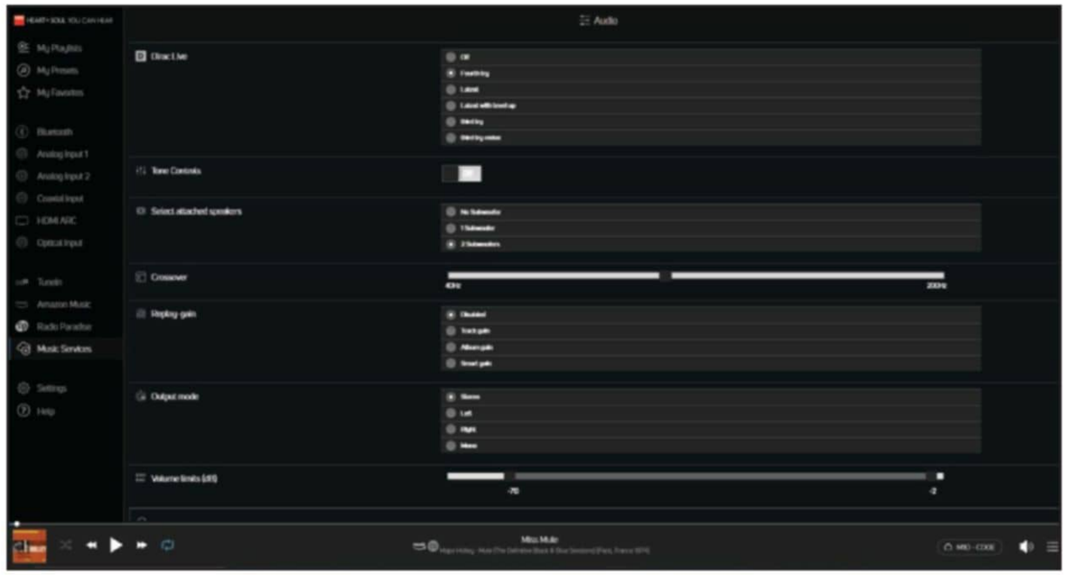


Figure 1: The control panel in the BluOS app allows the user to choose Dirac filters, crossover frequencies, equalization, and speaker configuration.

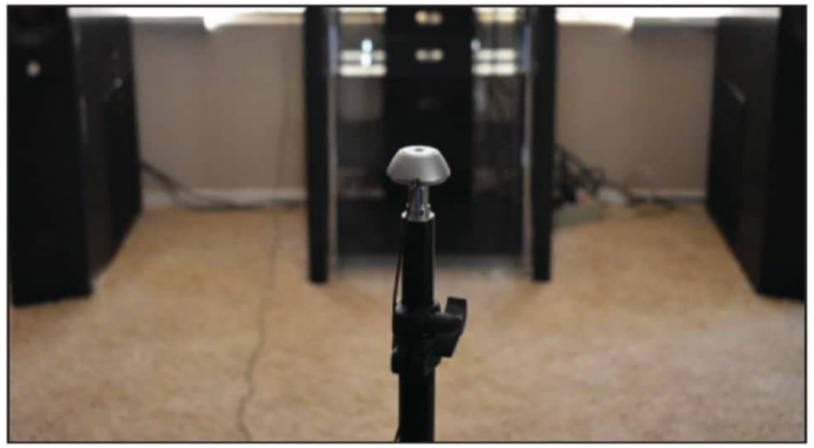


Photo 3: A measurement microphone with a calibration file is included with the M10 to set up Dirac. It mounts on any photographic tripod and points upward.

Dirac Live

This is, for me, the single most interesting feature of the M10. Dirac Live is room-correction software included with the M10 (and some other products from NAD). There are two versions. The standard one (Dirac Live LE) performs correction from 20 Hz to 500 Hz, the region where room modes predominate. For an additional $100, you can upgrade to the full-range version of Dirac Live, which works (nominally) up to 20 kHz. I did the upgrade, reasoning that if someone is in for a few thousand dollars, an extra $100 for an added nice and potentially useful feature is not a big deal. Worst case, it's the audio equivalent of undercoating in that new car; best case, it's a significant improvement.

Dirac is set up and run via a phone app or a desktop app. In my case, I used the desktop version for these tests, but I briefly ran through the phone version. Installation of the apps was simple and without incident. The app guides the user through the measurement and setup process (see the Sidebar).

Dirac works a bit differently than most other room correction software. A typical software package will measure at the listening position, determine a magnitude response curve, and then create an equalization function to flatten the response. More sophisticated packages will measure the response at several positions, then create a weighted response to equalize. Dirac takes this a few steps further.

As with the best conventional software, Dirac will take measurements at several different locations. However, rather than weighting them, as part of the calculation the Dirac software will look at frequency and phase response at each position, then differentiate between features that are constant between positions before applying correction.

That last statement is pretty simple, but in reality there's a lot of clever engineering that has to be done. There are four sorts of distortion effects degrading speaker-room performance: low-frequency standing waves (which are essentially a diffuse field and represent long delays from the main signal in the time domain); time delays between speaker drivers (which comes from the direction of the speakers and is close in time to the onset of the main signal); early reflections from cabinet diffraction and surfaces near the speakers (which also come from the direction of the speakers); and late reflections, which come from other areas of the room away from the speakers.

It's clear that using an inverse to the frequency magnitude response as a filter cannot properly fix these issues except at a single point. Even there, because the system will not be minimum phase, the arrival of energy to fix a problem in the magnitude response will be at a time removed from where the reflection responsible for the problem arrives. So one needs to look at the complex signal (i.e., both magnitude and phase) and this means correction in the time domain.

This now opens a new can of worms: linear phase filters versus minimum phase filters. With linear phase, the phase is not corrected, just the magnitude. With minimum phase, the phase can be corrected, but the possibility of audible pre-ringing becomes an issue. Dirac opted for a mixed approach, which allows correction of both frequency and phase under the constraint previously mentioned, and the correction can only be effective for errors that occur at all measurement positions. This method of phase correction can have a profound effect on the impulse response at the listening position, as will shortly be seen.

Since Dirac Live is incorporated into the M10, there's a measurement microphone and USB adapter included. The calibration file for this mic is built into the M10, and the Dirac setup software can access it. The included mic appears to be a standard inexpensive electret, but it's embedded in a metal housing that looks a bit like a truncated hockey puck, points upward, and has a screw fitting in its base that will fit photographic tripods (see **Photo 3**). Although I have instrumentation-grade microphones and the Dirac measurement software will accommodate them, I used the included mic for all Dirac measurements and setup.

In-Use Impressions

I set up the M10 in my main lab system. The loudspeakers were a pair of heavily modified NHT M3.3s, set up for biamping, via changes in the

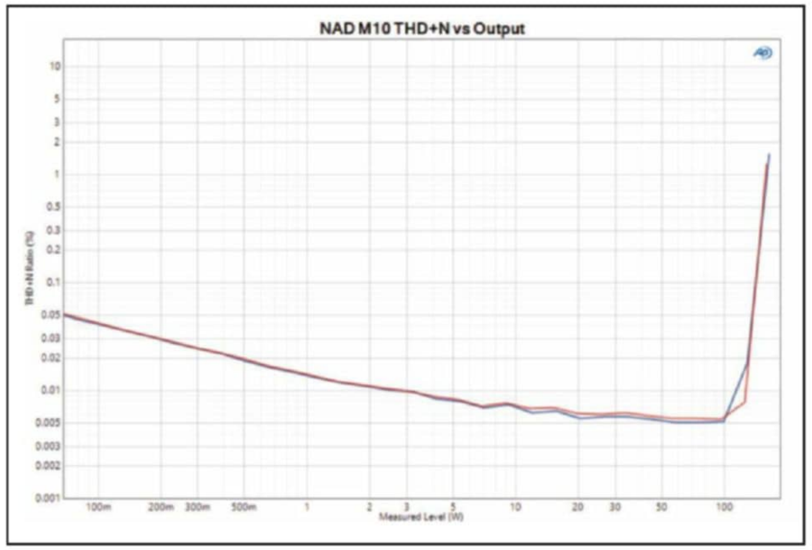


Figure 2: This plot of THD+N as a function of power shows the M10's nCore amps comfortably exceeding specifications.

crossovers and cabinets (described on my website, see Resources). Choosing the "2 Subwoofers" option from the BluOs M10 control screen, I used the top sections (midbass, midrange, and tweeter) as the "satellite" speakers, and the woofer sections as the subwoofers. The subwoofer part was driven by a Purifi EVAL1 stereo Class-D amplifier, with a variable gain stage inserted between the M10 subwoofer outputs and the Purifi to set subwoofer levels (most "real" powered subwoofers have a gain function built in); the crossover frequency from the M10 was set to 110 Hz, which is the crossover point I had previously been using in my main setup.

The real fun began when I tried integrating the M10 into our home network. This can either be done via an Ethernet connection or wirelessly. Since the stereo system was on the second floor but the network router was on the first floor, Wi-Fi was the only practical solution. Thus began at least two weeks of total frustration. Nothing worked the way it was documented, neither the desktop version of BluOS nor the iPhone app. The M10 would only intermittently appear as an option to add to the network, and following the documented instructions led to screens and options that were different than shown in the manual (and not useful). I went through several different people at NAD, Bluesound, and Lenbrook (the parent company for NAD and Bluesound) trying to get this resolved.

Although admittedly I am not a genius when it comes to networks, my wife is, and she was equally baffled. Eventually, with some guidance, we stumbled into a solution that worked for us, but it's probably specific to our network and the particular version of BluOS and NAD firmware that I have, so it is likely not going to be useful if you encounter the same issues. The one thing I can assure you is that following the documented procedure is almost certainly not going to be successful. Although I am not a big advocate of brick-and-mortar dealers, if you are planning to use the M10 (or similar devices) with Wi-Fi, I would only buy from a dealer who is willing to set it up for you and make sure it's functioning properly.

This warning notwithstanding, once we had the M10 operating on our network, the streaming, and the connection to other devices was indeed flawless. I have no inside knowledge about the product development process for NAD and Bluesound, but given the combination of the evolving feature set and documentation that lags badly, I'm guessing that they run using Agile methods rather than waterfall—this is the number one shortcoming of Agile-developed software.

With the M10 integrated into our network, I settled in for some extended listening, often with the fine company of *audioXpress* Technical Editor Jan Didden. We had spent several weeks together listening to my main lab system, which we both thought sounded pretty well. That was before the M10 and Dirac Live.

Before getting Dirac set up, the M10 sounded pretty much the same as my previous setup, which is high praise, since it replaced six large boxes and all of their associated wiring and interconnection. And the one difference is noise—even with my ear right up to the midrange or tweeter, I could hear absolutely no hiss or buzz—dead silence. Of course, the M10 also brought an enormous capability for streaming that my system had previously lacked.

Using either the desktop or iPhone version of BluOs, the integration of streaming services was terrific, very intuitive and quick. The basic look and feel of the interface is consistent among different music services and the TuneIn app. Although I mostly use Amazon Music HD, just to check I grabbed a couple of MQA-encoded songs off Tidal. They played without issues, and the "MQA" logo showed on the M10 panel so, for better or for worse (I am not a fan of the MQA concept), the detection and decoding work as intended.

Turning Dirac on after completing the measurements and setup described in the Sidebar was... a revelation. Besides the expected significant increase in bass smoothness, what was immediately apparent was a major (and I mean major) improvement in imaging and soundstaging—my listening room seemed largely to disappear, with the acoustics of the recording taking over. Although I had always thought of my speakers as imaging champs, this was truly a different level: turning Dirac Live off didn't affect the precision of instrument localization, but it absolutely reduced the amazing reach-out-and-touch-it 3D quality of images that we got with Dirac switched on. With Dirac on, the sound was completely detached from the speaker locations—an aspect that was not at all subtle and at times was almost spooky. I've used, and quite like, other room correction software, but this was something different and special.

I tend not to get into too much detail on listening impressions, but there were certainly a few recordings that really showed off the improvement wrought by the NAD/Dirac combination. One of my favorite albums is the eponymous *Redbird* (Signature Sounds), a quartet recording of some incredibly fine Folk/Americana performers done in a living room with a single stereo microphone. The tonality of the guitars and voices is incredibly natural, and there's very little compression or equalization—this is almost the essence of a minimalist recording. And I know the voices and instrument sounds since I've been up close and live with them on many occasions. It's been a reference for me over the years, so I was unprepared for what I heard with the M10 and Dirac.

First, I cued up "Ships" (a Greg Brown cover) and was suddenly in the living room with the players. They were sitting around the stereo mic, singing, and playing with great joy. You could even hear Kris Delmhorst leaning in toward the mic as the Elvis verse of the song came around to her. We sat in awed silence through the rest of the album.

The little background noises characteristic of living room recordings (passing cars, crickets, neighbor dogs, and the rustling of clothing) were more apparent, but put into perfect context, verisimilitude rather than annoyance. I can't say I heard things I hadn't heard in this recording before but it all seemed just more coherent and for lack of a better word, "right." The sound in my room honored the natural miking of this recording, especially things like the size of the guitars and tonality of the voices. Switching Dirac off, I was again listening to a very good stereo system in my room, but was no longer in their living room with them playing in front of me.

Moving to classical, we turned to Enrique Bernaldo de Quiros playing "Hungarian Rhapsody no. 12" on the outstanding recording *Conversations* (PlayClassics) in hi-res format. With the Dirac on, the piano's outlines seemed more solid, and we could easily pull the sounds of the mechanism and pedals at the appropriate points in space. I'm sure it's an illusion, but with Dirac Live engaged, the dynamics of the recording seemed to be brought out in higher contrast. Closing my eyes, the studio acoustic sounded almost enveloping, as if the speaker end of my room didn't exist.

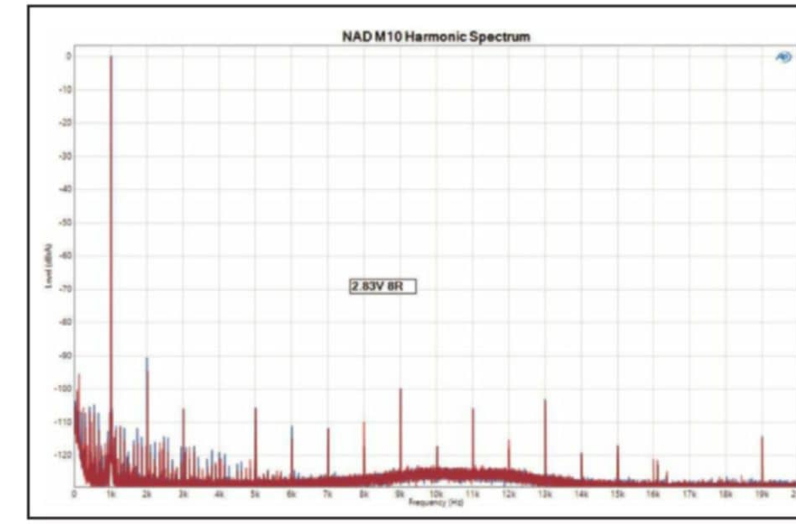


Figure 3: The harmonic distortion spectrum of the M10 at low power is dominated by small amounts of low-order distortion.

And streaming the delightful "Play Chords!" by the Milt Buckner Trio (MPS), Jo Jones' brushed drums and cymbals were placed perfectly and solidly in space, throwing a light on the conversation with Buckner's "locked hands" piano. You can get a similar effect, possibly even a bit more transparent, listening to Kenny Washington's brushes on "Brushes and Brass" from *Clark Terry: Live at the Village Gate* (Chesky), though the interrogation is from a muted trumpet instead of a piano. When Terry vocalized, it made me jump—now that is realism!

Studio rock is always a more subjective judgment, since it's not a transcription of a live acoustic event in a performance space. But still, it should sound "good"

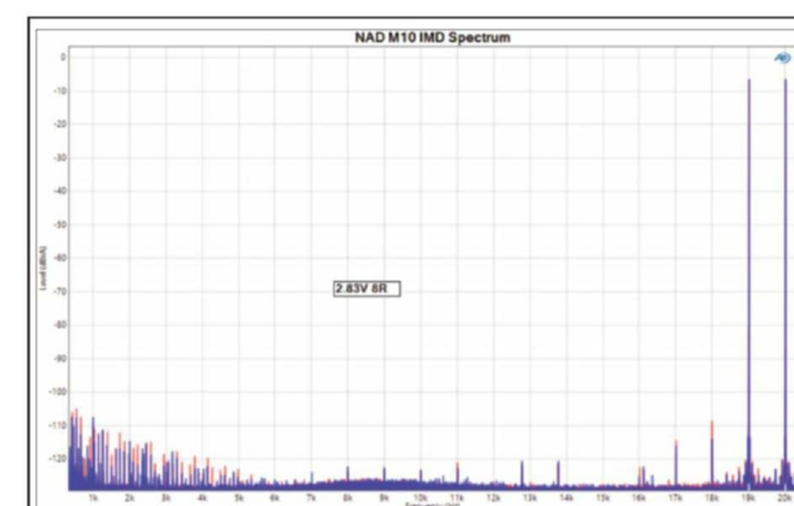


Figure 4: The 19+20kHz intermodulation spectrum of the M10 is extremely clean.

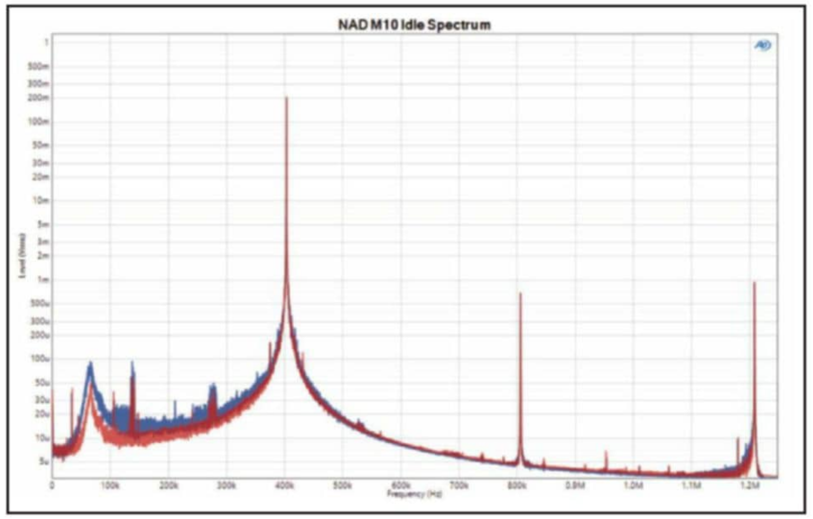


Figure 5: The M10's wideband (up to 1.2 MHz) noise spectrum indicates very low levels of ultrasonic and switching noise.

in the sense of sounding clean, instruments positioned where the engineer put them in the mix, and a nice punch and drive. One of my favorite tracks for that quality is from Okkervil River's landmark album *Black Sheep Boy* (Jagjaguwar), the song "For Real," where the electric guitar and drum come in—when things are dynamic and right, it'll make you jump! And with the M10/Dirac Live combo, yes it did. Likewise Tool's "Fear Inoculum" (Tool Dissectional/RCA) kept urging me to turn it up louder. I managed to clip the amp a few times on "Chocolate Chip Trip" turned up to 11, but it really was seriously loud, and perfectly clean up to that point.

One other thing to note—the M10 has a built-in latency of 50 ms, which can be increased to between 50 ms to 100 ms for the purpose of allowing sound and image to be synced in a home-theater setup. In my case, I only used the M10 for a music system, but for home-theater users, this is a very nice feature (though finding the documentation on it is a bit of a challenge!).

Every ointment has its fly, and in this case, it's a big and glaring one. While using the digital inputs (coax or optical), I thought I would hear intermittent ticks or dropouts. This was rather annoying, but I didn't experience this at all with streaming or analog sources. Likewise, sending data via the USB connection worked perfectly, both with a thumb drive or an SSD containing my ripped albums and home recordings. After much correspondence with Lenbrook, I got an explanation which boiled down to, "The SPDIF receiver will occasionally lose lock. We're aware of this issue and will attempt to fix it via firmware, but are delayed because of current pandemic lockdown restrictions." I hope this will be fixed shortly—my sources for coax and optical have other connection options, but in a unit at this price level, the basics should work reliably.

Some Brief Additional Measurements

To keep things to a manageable length, I am not presenting the entire suite of measurements of the electronics section, but I did want to mention a few highlights of the power amp. **Figure 2** shows THD+N at 1 kHz as a function of power into an 8R dummy load. At lower power, it's noise-dominated, with clipping setting in at more than 150 W, comfortably exceeding specifications.

The spectrum of a 1 kHz tone at 2.83 V into 8R (1 W) is shown in **Figure 3**. Getting this spectrum and much of my other data took a rather roundabout method, since the digital inputs had the intermittent loss of lock defect I noted earlier. I had to save my test signals as files, then access them via the M10's USB port. Sweep measurements had to be done via the analog inputs. Highly inconvenient! The distortion is low (better than 0.008%), but not spectacularly so, with second harmonic dominant. The 19+20 kHz intermodulation spectrum is shown in **Figure 4** and is quite clean, with all sidebands and first order products below -100 dB.

The wideband noise spectrum at idle is shown in **Figure 5**. It's dominated by the nCore amps' switching frequency of 400 kHz, but that's only at about 200 mV, nothing to worry about. All other ultrasonic noise spikes are under 1 mV.

I also checked the M10's subwoofer outputs since their slopes and filter types are not specified in NAD's documentation (that I could find, anyway). **Figure 6** shows the subwoofer outputs (line level) and the main outputs (speaker level) with the crossover frequency set to 110 Hz. The response resembles a fourth-order (24 dB/octave) Linkwitz-Reilly response. The more jagged traces are the corresponding electrical outputs with the Dirac correction switched on (see the Sidebar).

Wrap Up

The M10 is mainly wonderful, and at the same time, slightly disappointing. The disappointing part includes failure to accomplish the simple task of taking digital inputs correctly, something that doesn't plague even the cheapest devices that I own.

About the Author

Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently works as an R&D director for a construction products company in Mesa, AZ. His professional research interests have spanned theoretical physics, electronics, chemistry, spectroscopy, aerospace, biology, and sensory science. One day, he will figure out what he would like to be when he grows up.

The streaming and analog input functioned perfectly, so this was particularly annoying by comparison. NAD claims that a fix is in the works, and it may be available by the time this review reaches print. Setting the M10 up on a network using Wi-Fi was an adventure in frustration—any

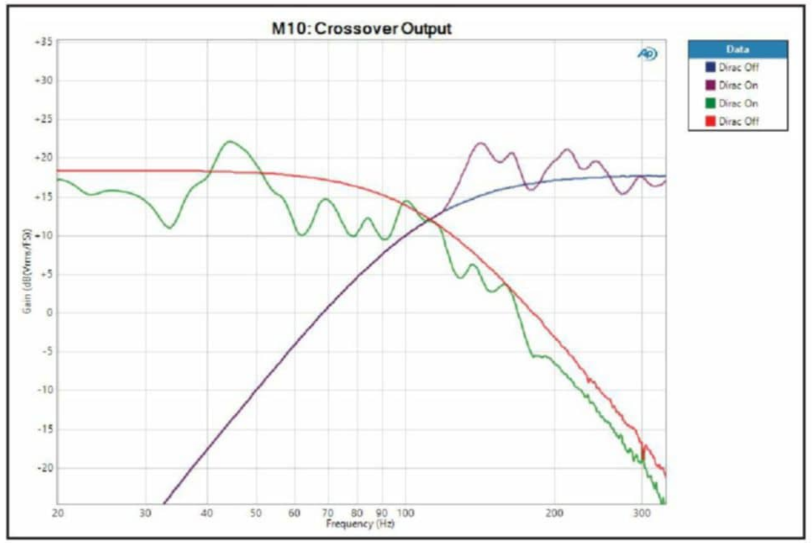


Figure 6: The subwoofer crossover of the M10 provides 24 dB/octave Linkwitz-Reilly filtering. The electrical output with Dirac Live turned on shows the applied correction response.

Resources

A. Bahne, L. Brännmark, and A. Ahlen, "Audio Precompensation Controller Design with Pairwise Loudspeaker Channel Similarity," US Patent 9,426,600 (2016).

Bluesound BluOS, https://bluos.net

L. Brännmark, "Digital Room Correction: Benefits, Common Pitfalls, and State of the Art," Audio Engineering Society (AES) Sweden Lecture, 2014, https://www.dirac.com/live-home-professional-audio-info

L. Brännmark and J. Ägren, "Reducing the Phase Difference Between Audio Channels at Multiple Spatial Positions," US Patent 10.284,995 (2019).

L. Brännmark, M. Sternad, and M. Johansson, "Sound Field Control in Multiple Listening Regions," US Patent 8,213,637 (2012).

M. Johansson," On Room Correction and Equalization of Sound Systems," presented at 123rd Audio Engineering Society (AES) Convention, 2007, www.dirac.com/live-home-professional-audio-info

NAD M10, https://nadelectronics.com/product/m10-bluos-streaming-amplifier

S. Yaniger, "Biamping the NHT M3.3," https://syclotron.com/biamping-the-nht-m3-3

S. Yaniger, "Sonarworks Reference 4 Studio Edition Loudspeaker and Headphone Correction System," *audioXpress*, July 2018, https://audioxpress.com/article/fresh-from-the-bench-sonarworks-reference-4-studio-edition-loudspeaker-and-headphone-correction-system

purchase of the M10 ought to be conditional on the dealer getting the unit set up on your home Wi-Fi network. Ethernet would likely be simpler if that's a practical solution for you. Documentation of features and menus is spotty and has not kept up with software and firmware updates, but on the positive side, the feature set is remarkably wide and most are quite easy to use.

Also on the positive side, the integration of streaming services, the amplification, the control via BluOS desktop or phone app (which were easy and intuitive to use once the networking issues were sorted out), the variable subwoofer outputs, and most of all the Dirac Live room correction software make this a compelling unit. The slick packaging and good looks add to the appeal.

Since I was able to find workarounds to the serious problems of badly functioning digital inputs (and I have my fingers crossed for a firmware fix), the M10 became the centerpiece of my main listening system, replacing an entire rack of gear. My stereo has become a music appliance with absolutely superb sound.

Given the breathless prose in so many equipment reviews, one might think that there's a constant stream of step-level improvements in sound quality for home audio; in reality, major change only happens every decade or two. The remainder is either minor refinement or achieving the same functions but in smaller and less expensive form factors. But in this case, the change was actual, real, and significant.

For the first time in a long time, I found myself seriously wondering, "Could this be my last stereo system?" After a couple of months living with the M10, I couldn't decide whether to be elated that this quality of sound and source flexibility is available in such a small and simple package or to be depressed that my rack full of complex and impressive-looking gear was now empty and obsolete. Although "lifestyle system" is used pejoratively by some audio enthusiasts, in this case, the product delivers the audio performance as well. There was nothing sonically to quibble about using analog, USB, and network inputs.

Subject to my caveats about network setup and digital inputs, the M10 is really a game changer. After getting it set up and doing the Dirac calibrations, I've got the best sound I've ever experienced in my home, and have been delighted with the ease of use. The versatility is unprecedented in my experience. This review was, I confess, difficult to write because rather than set up mikes and make measurements, I kept getting distracted by listening to music. I think that's the bottom line.

Room Correction Measurements Using Dirac Live

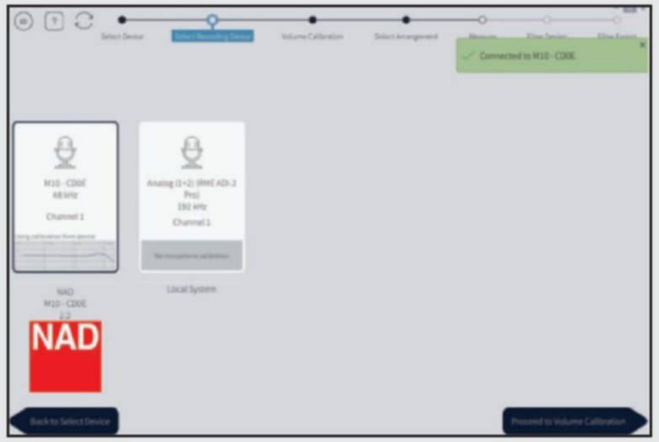


Figure 1: The M10 enables the user to select either the included microphone or an external mic and interface.

Dirac has tried to make the measurement and calibration process as simple and flexible as possible. The app takes the user through a series of screens to achieve the calibration. First, the user can choose whether to use the included mic (see **Figure 1**) plugged into the M10's USB port or a separate mic/preamp with its own calibration file. I opted for the former, since that's what most users will do.

Once the M10 and mic source are chosen, the user has the options of a studio or home setting for the loudspeaker arrangement. Within the "Home" option, one has the further choice of a wide or narrow seating arrangement (see **Figure 2**), with the former requiring more measurement locations and potentially not having the same degree of correction. Since I'm a selfish man who usually listens to music in solitary, my choice was the single seat, "tightly focused" imaging. The Dirac/M10 combination allows up to five different filter settings to be stored, so if you like, you can do measurement/calibration for both single and multiple seats, then select the appropriate filter for any given listening situation.

Once the speaker and seating arrangements are chosen, the app has the user set testing levels using the app's pink noise source, and running through each speaker in turn. With the overall levels set, the measurement process then begins. For the chosen seating arrangement, the Dirac app shows the mic measurement positions (see **Figure 3**).

There's two small problems. First, the provided test mic needs to point upward and mounts on a photo tripod. That's great except for measurement at the actual listening position, where the tripod and the chair cannot occupy the same space at the same time! This is less of a problem with a more conventional measurement mic, which can be placed on a boom and extended over the chair. I did several workarounds here, but it would be nice to have some manufacturer suggestions.

The second problem is that the positions to the sides, above and below the listening height are not really specified (i.e., there's a line drawing with circles, but no actual distances). This is left to guesswork. The ranging process used by Sonarworks is still the best I've seen for setup, but giving the user actual distances between points would be a nice second-best.

If I can't be accurate, I at least want to be precise! So we did two things for precision—marking and measuring listening position and measurement locations on the floor with tape (see **Photo 1**), and making marks on the mic tripod to accurately reset

Figure 2: Dirac Live can be configured for a narrow or wide area, depending on desired seating arrangement. It can also be configured for studio use with desktop monitors.

Figure 3: Once the area and seating positions are chosen, Dirac Live indicates the positions for the acoustic measurement.

the height at each location (see **Photo 2**). These are easy things to do for repeatability, but I sure would like to know the distances assumed for the mic positions during the filter calculation.

Other than this quibble, the measurement process was easy and only took 15 minutes or so. At each point, the Dirac system does a sine sweep for each speaker, then a final verification sweep before prompting you to move on to the next position. Once that's complete, the filter can be generated (it's apparently calculated at Dirac rather than by your local device), then loaded into one of the slots of the M10 and saved. A useful feature is that you can choose the target response for the filter to be something other than flat—a common adjustment is an overall downward response with increasing frequency (e.g., the KEF and Harman curves). The response curve is adjusted graphically by moving points on the target curve

Photo 1: To ensure mic placement was consistent from side to side, we measured and marked spots on the floor. Note my remarkably empty equipment rack!

Photo 2: Putting marks on the tripod shaft enables consistent and repeatable changes in mic height for the Dirac measurements.

Dirac enables you to see the results of the measurements for each speaker, corrected and uncorrected. Even better, you can also see the uncorrected and corrected impulse response for each speaker.

Figure 4 is a screenshot of the measured uncorrected and corrected subwoofer response. The response swing without correction (thin trace) is almost 15 dB; after correction, this is improved to within a 5 dB window (thick trace). **Figure 5** is the equivalent responses of the main speakers before and after correction—the effect is a bit subtler but still very evident.

Even more interesting is the comparison of the impulse responses. **Figure 6** shows the uncorrected (upper) and the Dirac corrected impulse response (lower). What's striking is that the reflections at about 3 ms and 4ms from the main impulse are only slightly diminished, but the cleanness of the main impulse and phase coherence is markedly improved.

To get a better overall picture of what Dirac was doing in my system, I set up my usual acoustic measurement system of an Audio Precision APx525 analyzer, APx1701 transducer interface, and a PCB Piezotronics 376A32 1/2" condenser mic, with the mic placed at the listening position. The M10 was fed Farina chirp (log sine) test signals via its analog inputs, and these were converted

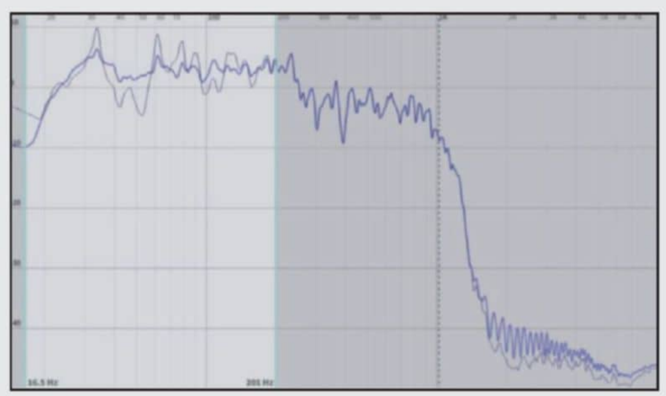


Figure 4: Dirac displays the measured and corrected frequency responses of each of the subwoofers, in this case, the left one. Note the greatly improved bass smoothness.

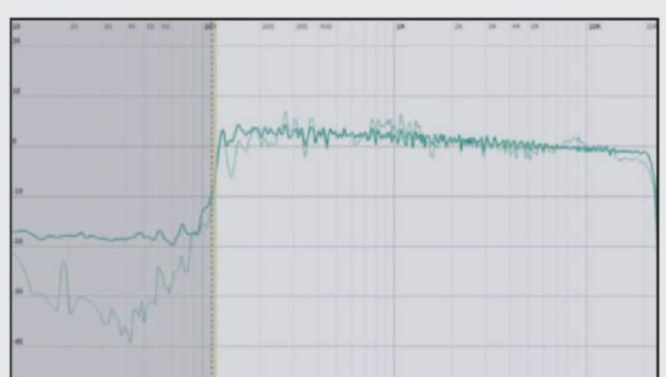


Figure 5: The frequency response of the left main speaker before and after Dirac correction shown in the Dirac window also shows improved flatness of frequency response.

to frequency and impulse responses.

Figure 7 shows the ungated (in room full range) frequency response of my speakers with Dirac turned off, fed a stereo signal. It was taken at the listening position as a power average of nine measurements with the mic moved a few inches horizontally and vertically to achieve a spatial average. The peaks and dips are pretty typical for speakers in an untreated or (in my case) lightly treated room without quasi-anechoic gating.

Switching the Dirac on gave me the frequency response shown in **Figure 8**. This is significantly smoother, and absolutely sounded that way. I noticed during the measurement that the individual responses at different measurement mic positions were nearly coincident, so something was going on beyond mere equalization. And of course, that "something" is the time domain/phase correction.

At the listening position with a stereo signal, the measured impulse response is shown in **Figure 9**, both without (a) and with (b) the Dirac filter turned on. The reflections seen earlier appear in this measurement as well, but it's very clear that, consistent with the single speaker results from Dirac, the initial impulse for the stereo pair with subwoofers is much sharper and cleaner immediately following the main spike, and even the reflections look more coherent. This is a striking demonstration of the Dirac's power in the time domain.

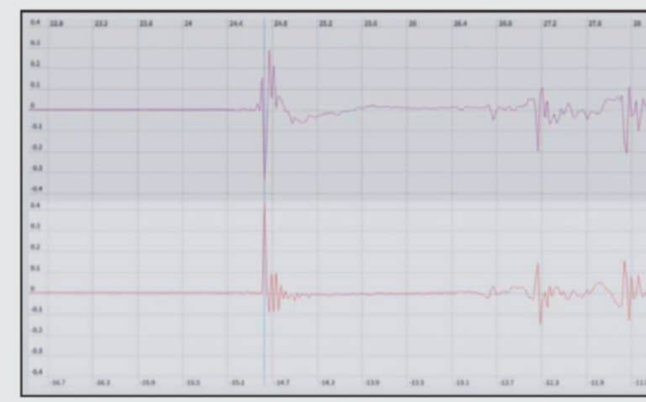


Figure 6: Comparing the ungated in-room impulse response of the left speaker uncorrected (upper) and the same impulse with Dirac Live correction (lower) shows a remarkable improvement in phase coherency over the first few milliseconds.

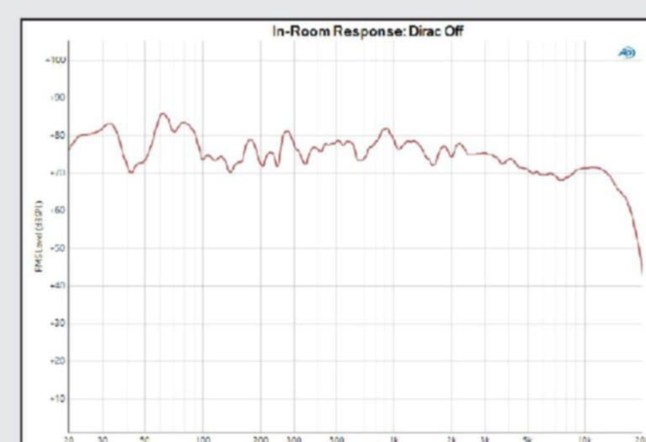


Figure 7: The ungated full-range frequency response of a mono signal driving both speakers taken at the listening position has typical bumps and dips from room interaction.

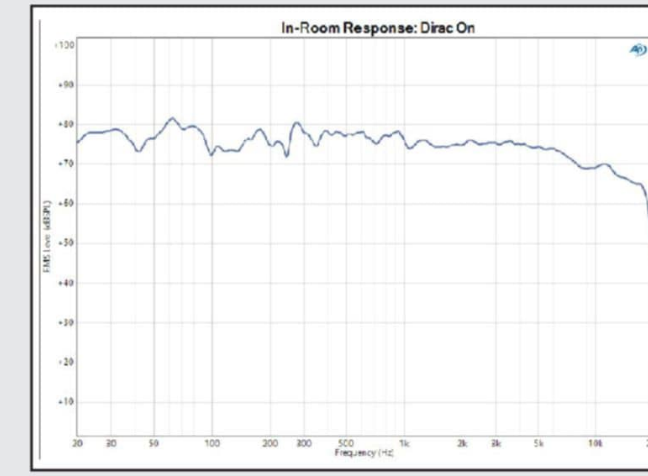


Figure 8: Taking the same measurement as in Figure 7, but with Dirac Live switched on, shows a significant improvement in the flatness of the response.

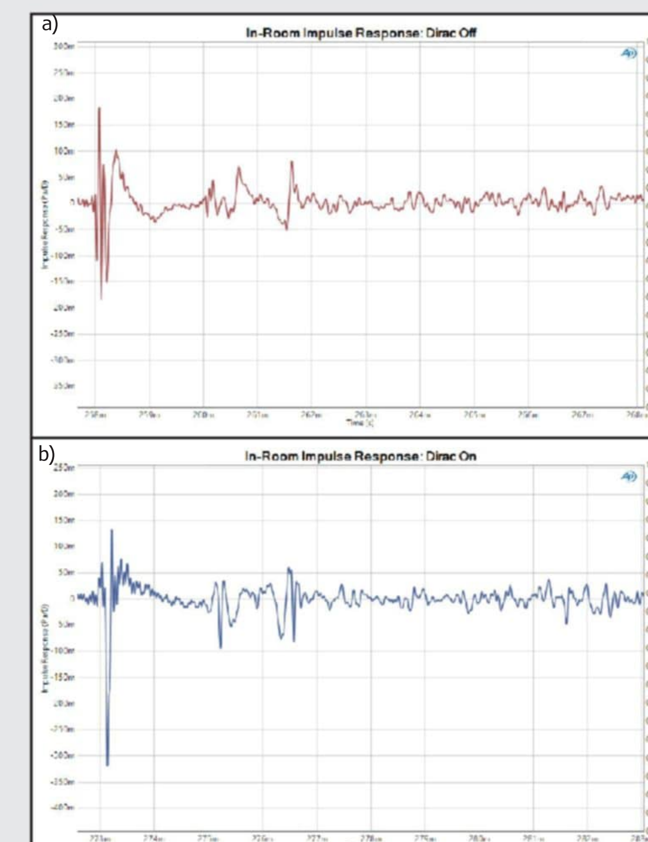


Figure 9: The impulse response of a mono signal driving both speakers taken at the listening position shows a marked increase in coherence when comparing uncorrected (a) with corrected by Dirac (b).

Testing Acoustical Systems

By
Richard Honeycutt
(United States)

There are a number of parameters that sometimes require testing during the design, optimization, renovation, and use of acoustical and electroacoustical systems. For this article, Richard Honeycutt primarily focuses on acoustical measurements.

Photo 1: The Fogg Lecture Hall was an acoustical disaster in 1895 when Wallace Sabine began his work.

Acoustical measurements include reverberation time (RT), early decay time (EDT), echoes, objective clarity (C80), objective definition (D50), acoustical strength (G), speech intelligibility (STI, STI-N, STI-PA, and %ALcons), sound pressure level (SPL, SPL actual, LMIN, LMAX, LPEAK, LEQ), and background noise.

The Fogg Lecture Hall of Harvard University, built in 1883, exhibited a serious lack of speech intelligibility. In 1895, junior physics professor Wallace C. Sabine was assigned the formidable task of taming the beastly acoustics. He began by observing that the root of the problem was the slow decay of sound in the room, allowing speech syllables to be obscured by the acoustically reverberated syllables of previous speech. Sabine defined the time required for a sound at the level of normal conversation to decay to the level of inaudibility—a 60-dB decay—as the reverberation time (RT) of the room. He reasoned that the RT of a room was determined by the cubic volume of the room and its absorptivity, as shown by the well-known Sabine equation:

$$RT = \frac{0.161V}{\Sigma S\alpha}$$

where V = the volume of the room in cubic meters, and the denominator of the fraction signifies the sum of the products of the area (S) and absorption (α) of each material.

Understanding that the absorptivity varied with frequency, Sabine knew he must measure the RT throughout the speech frequency range.

Today, we would measure RT by using a known signal source—perhaps a swept sine wave or pink noise source—to acoustically excite the room. We would feed the signal from our measurement mic to a computer, where software would perform a Fast Fourier Transform (FFT) and give us the RT in various octave or one-third-octave bands. Of course, such instruments were not available to Sabine, so he used organ pipes to create specific pitches, blowing into them to establish a constant sound level, then abruptly stopping the breath. He used a stopwatch to measure the time between the cessation of sound from the pipe and when the tone was no longer audible.

Use of RT Measurements

RT measurements are used today for a variety of reasons: to verify that a room performs acoustically as designed, to evaluate the acoustical effects of changes in room design or finishes, and to identify causes of acoustical faults leading to complaints from users of the space. In my consulting practice, I have worked with cathedrals having enormous cubic volume and virtually no acoustical absorption, 1920s-era cinemas that have been repurposed as civic centers, and numerous small- to medium-sized halls. The cathedrals are always challenging as regards to taming excessive RT, especially in the low to middle frequency range. The old cinemas usually have the opposite problem: while they functioned well as movie houses, and still are quite good as lecture halls, their "dry" sound makes them generally undesirable for orchestra and choir performances.

My services were once sought to evaluate issues with people attending events in one middle-sized auditorium: they "couldn't hear" the person talking. In actuality, "couldn't hear" meant "couldn't understand." As illustrated in **Figure 1**, the RT in the 1000 Hz and lower octave bands was well above the target value (about 1.5 seconds at 500 Hz) for this particular auditorium, and the excess RT increased as frequency became lower.

While this presented a significant speech-intelligibility problem, it would also have caused music to sound bassy, and perhaps to raise complaints about the pipe organ being too loud (because pipe organs produce a lot of sound energy at low frequencies). By having made detailed measurements of RT vs. frequency at various locations in the auditorium, I was able to recommend a solution that would decrease the low-frequency RT, particularly on the main floor, where the problem was greatest.

Calculating RT

Before the advent of computerized acoustical measuring instruments, a common method of estimating RT was to excite the room using an impulse (handclap, balloon burst, or starter's pistol discharge) and measure the time required for the reverberant sound to fade from audibility. This method could not discriminate the RT at various frequencies, and so would not have shown up the problem in the auditorium I just recounted.

Sabine based his work on the assumption that the sound reflections in the room were diffuse (every surface having an equal probability of being struck by the same number of sound wavefronts), so the RT would be the same throughout the room. This explains why the Sabine equation has no variables corresponding to the positions of the sound source or the observer. In practice, the "Sabine conditions" are not met in many rooms.

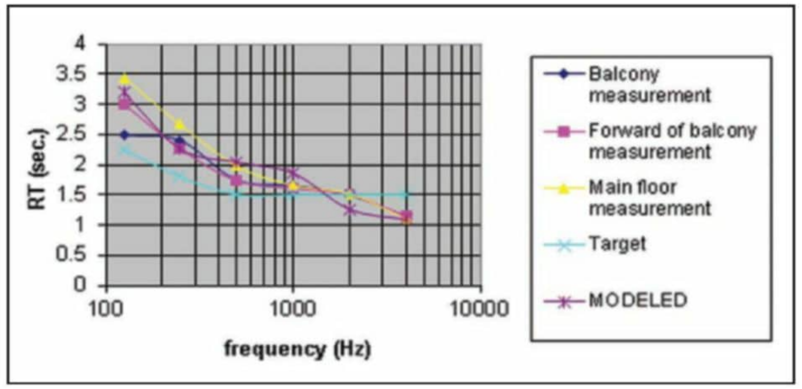


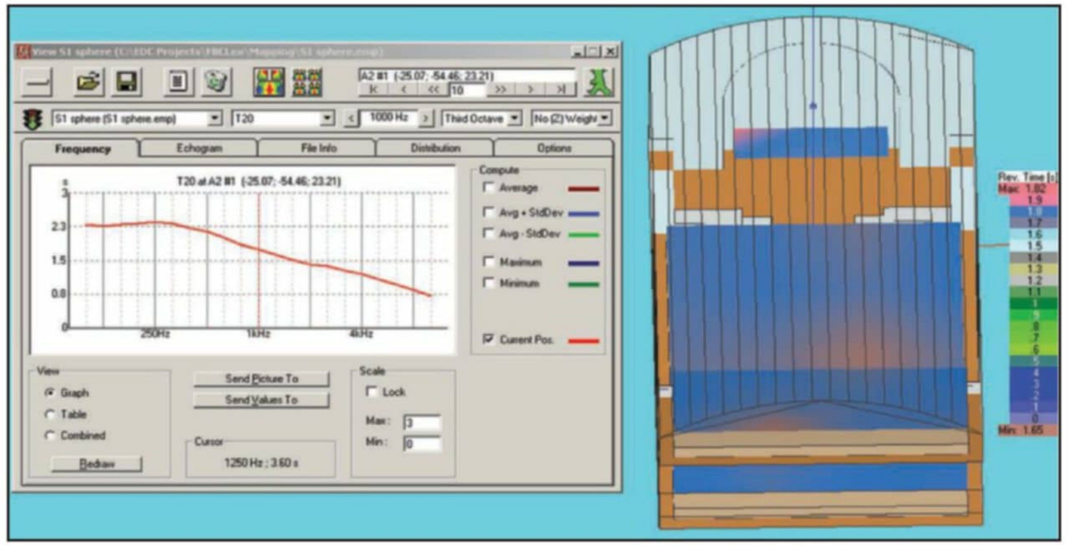
Figure 1: RT in this auditorium was far too high at low frequencies.

Examples involve balconies. The RT measured in balcony seats is often lower than on the main floor, as shown in Figure 1. Under-balcony-area RT is often even shorter than balcony RT, especially if the depth of the under-balcony area is significantly greater than its height.

Traditionally, an omnidirectional source is used for RT measurements (see **Photo 2**). Initially, this sounds like a good idea. In fact, the directionality of an installed speaker is often used as a sales tool. ("Because it's directional, it doesn't excite the reverberant field as much as an omni speaker, which sprays sound all over the place.") So spraying sound all over the place seems like a good thing in

Photo 2: This dodecahedral speaker radiates almost omnidirectionally.

Figure 2: This graph shows RT vs. frequency and a map of RT in various positions in an auditorium. An omni speaker was used.



measuring RT. However, if you compare the EASE models of the same room with a directional main speaker and with a spherical ("omni") speaker, you'll see almost no difference in predicted RT, either overall or in any specific places (see **Figure 2** and **Figure 3**). Measurements in this room verify the results predicted by the model. I should mention that speaker directivity does have other beneficial effects, even if it does little to help control RT.

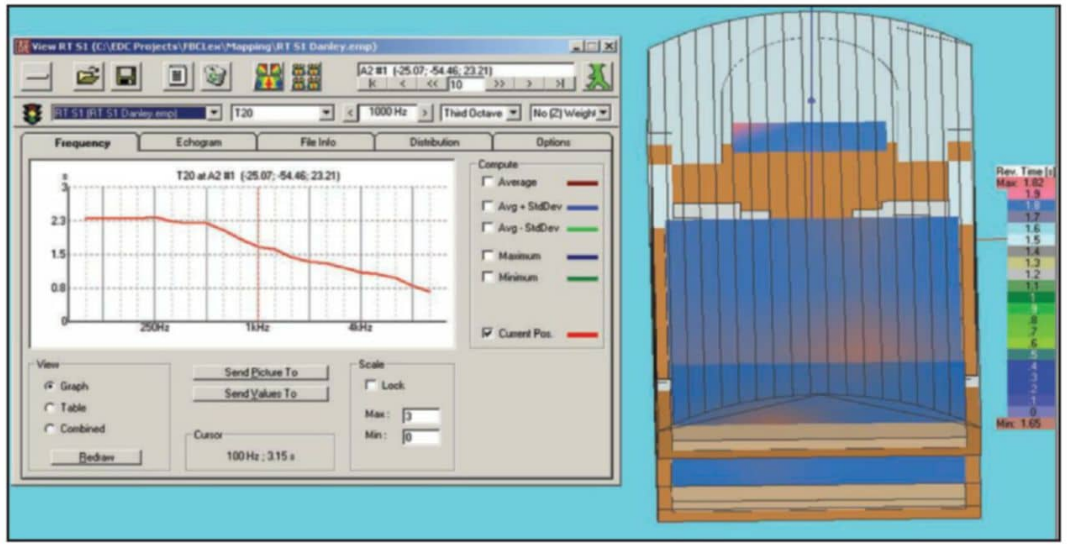
The ideal location for the sound source in RT measurements is near the center of the room and well above the floor. Asymmetrical placement can skew the results.

Measurement of Other Acoustical Parameters

Early decay time (EDT) is six times the time required for a sound to decay 10 dB. In almost any room, the EDT is shorter than the RT. In fact, it has been found that listener perception of the "liveness" of a concert hall (and perhaps the speech intelligibility) is more closely correlated with EDT than with RT. Since EDT is more strongly affected by the placement of acoustical absorption than is RT, the two can to some extent be varied independently. **Figure 4** shows the variation of EDT with frequency at one particular position in an auditorium. EDT is automatically provided in the results of most computerized RT measurements.

Echoes are different from reverberation, although both are manifestations of reflected sound. The effect of an echo is to repeat a sound, while the effect of reverberation is to lengthen it. ("clap-clap-clap-clap vs. cla-a-a-p"). The most common method of investigating echoes is to use an impulsive sound and listen for repetitions. A way to measure the delay of an echo is to play an anechoic of an impulsive sound (perhaps bongos with one strike every 5 seconds) and digitally record the resulting sound at the location where the echo is heard. If the .wav file is loaded into software such as Adobe Audition, Audacity, or another .wav editor, the time between echoes can be read from the computer screen. This time can be used to help locate the surfaces from which the echo is being reflected.

Figure 3: The graph of RT vs. frequency and a map of RT at various positions with a directional (horn) speaker shows performance almost identical to that shown in Photo 3.



Objective clarity (C80), and objective definition (D50) quantify the ability of a venue to support clear musical and speech articulation, respectively. These measures assist in identifying acoustical issues that result in "smeared" music and "blurred" speech. They are also provided by RT measurement software.

Acoustical strength (G), often known as "room gain," is the SPL boost provided by the room, compared to what a listener would hear at the same distance from the source outdoors. A sufficiently high value of G helps unamplified sources to project well in the hall. Reflecting shells positioned above choirs or orchestras help increase G. In one project, I was asked to recommend improvements to a choir's ability to be heard in an auditorium.

The original measured G, mapped onto the floor plan, is shown in **Figure 5**. The mapped values correspond to the average G from 125 Hz to 4 kHz. The ratio of maximum-to-minimum G was 5.6 dB. After a choir shell was installed, the G almost doubled, and its minimum value (back of the room) was only 2.7 dB lower than the maximum value (front of room), resulting in much improved choir projection into the seating area (see **Figure 6**). Speech intelligibility is specified in various ways: STI, STI-N, STI-PA, and %Alcons. Specialized instruments are used to measure intelligibility. However, the main determinants of speech intelligibility are RT and background noise. D(50) is used by some consultants as a guide to the speech intelligibility of a venue.

Sound pressure level (SPL) is measured for a variety of different reasons, including hearing protection, ensuring that adequate levels are provided for hearing speech and music, and privacy. Parameters important for useful SPL measurements include the measurement length (slow, fast, impulse, peak, or other settings on a sound level meter), averaging, and minimum/maximum levels during the measurement period.

Details about equipment for acoustical testing are available in the Sound Control articles published in *audioXpress* from November 2014 through January 2016.

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

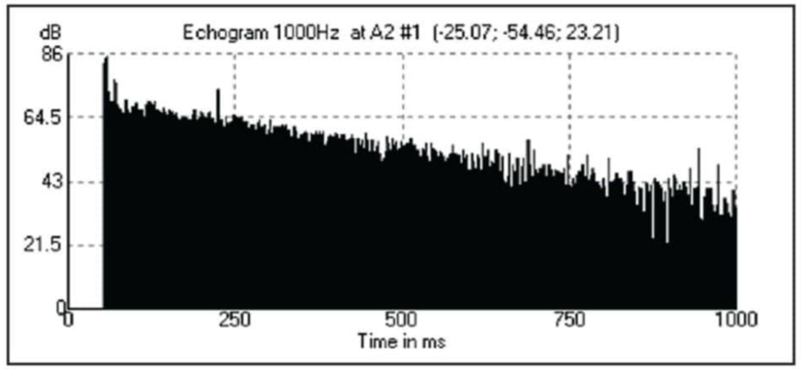


Figure 4: This echogram shows the EDT vs. frequency for one position in an auditorium.

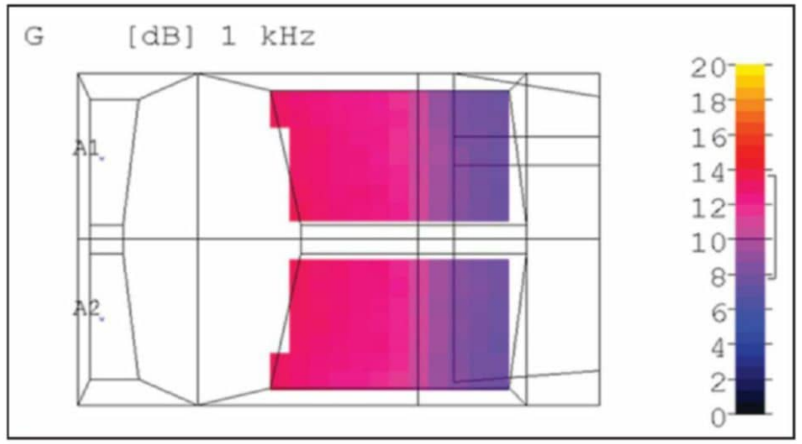


Figure 5: The acoustical strength varies quite a bit in the original condition of the auditorium.

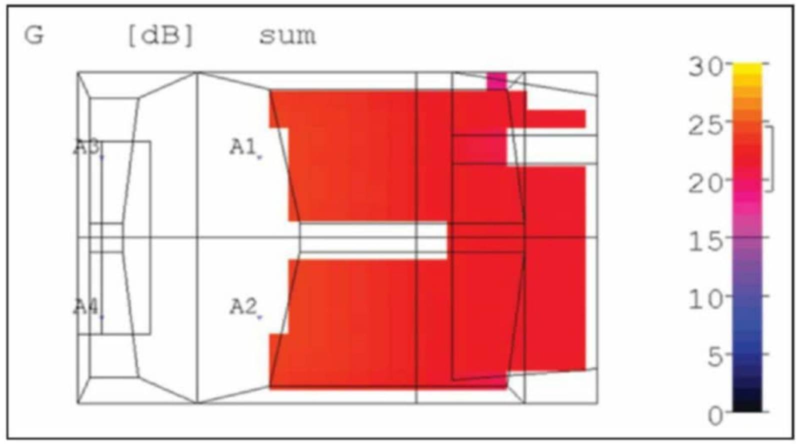


Figure 6: After installation of a choir shell, G improved and became much more uniform.

Acoustic Solutions Round-Up 2020

From Acoustic Comfort to Productivity

By
J. Martins
(Editor-in-Chief)

Acoustic science is where everything starts, determining how sound and audio systems behave in practice. *audioXpress* has made acoustics an integral part of its mission.

For this year's Acoustic Solutions Round-Up, we have taken a more diversified approach. First and foremost, we focus on home environments, listening spaces, and adapting spaces for communication and remote collaboration applications—which makes sense as we continue to deal with the global pandemic and we realize that investments in this area are worth considering long term. Second, we highlight a few useful solutions for acoustic measurement and acoustic testing in product development, given that the need to evaluate and profile spaces is one of the very first steps toward dealing with an acoustical solution.

Kvadrat Soft Cells

A Textile Approach for Acoustics

Kvadrat has been a leading textile innovator since 1968 when the company was founded in Denmark. The brand is globally renowned and its high-quality textiles and textile-related products are specified for public spaces and domestic interiors by architects, designers, and consumers. Kvadrat Soft Cells is a Kvadrat-owned company that creates fully customizable, high-performance acoustic panels characterized by enduring aesthetic values, and offers end-to-end project support, delivered by its global network of acoustic specialists.

The strong appeal of Kvadrat Soft Cells is also the company's unique ability to combine aesthetics with technological innovations in acoustic solutions. Reflecting this, Soft Cells acoustic panels set the benchmark for sustainability, flexibility and durability.

Kvadrat Soft Cells Standard are large-size panels that are effective on large surfaces for sound conditioning from 200 Hz to 4000 Hz.

Soft Acoustic Panels

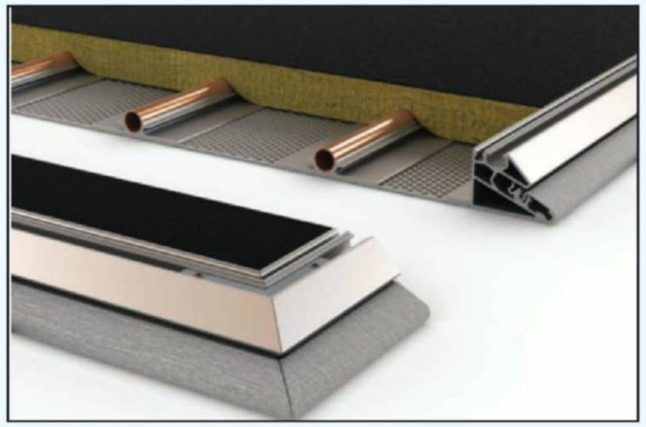
Soft Cells are acoustic panels that deliver up to Class-A sound absorption. They can be mounted both on walls and ceilings and are available in a wide choice of Kvadrat textiles and fully customizable, so they can be seamlessly integrated into any design scheme. Strong references for Soft Cells are installations in a variety of places—from The Royal Danish Library and Oxford University to Rolls-Royce and Microsoft corporate buildings.

Acoustics consultants recognize that textiles add acoustic value, add comfort to a space from its tactile surfaces, and are aesthetically appealing. Textile panels can also be a valuable solution to many acoustic challenges in interiors, promoting productivity.

To manufacture the acoustic panels, Kvadrat Soft Cells also embraces an eco-friendly approach. The structural Soft Cells frames are made with a minimum of 50% recycled aluminum and include a patented tensioning mechanism, which ensures they are unaffected by temperature or humidity for many years.

Within the Soft Cells range there are several different products, which all have in common the fact that, from the front, they all appear as minimalistic textile surfaces with impeccable finishes. Underneath, the products reveal a sturdy multilayered construction using rigid aluminum frames and perforated laminates, combined with soft materials.

As an example, Soft Cells Standard panels rely on two layers of tensioned textile to control sound. Typically, this version offers Class-C sound absorption, as per ISO 11654.



This shows the construction detail of the Kvadrat Soft Cell range of Weave Radiant textile panels.

Soft Cells Broadline panels incorporate acoustic padding behind a textile layer. Typically, this model offers Class-A broadband sound absorption, as per ISO 11654.

Soft Cells Lowtone panels have a specially developed glass textile membrane behind a tensioned front fabric layer, and deliver excellent acoustic performance concentrated in the low- and mid-range frequencies—from 125 Hz upward.

All these panels can be applied in large surfaces such as corporate offices and are valuable resources for acoustic treatment of any house interior.

And Soft Cells' catalog even includes clever solutions such as the patented Weave Radiant Textile panels that control thermal comfort. The panels achieve this by incorporating water pipes, which heat and/or cool a radiant surface located behind a layer of tensioned textile.

More recently, the company also developed a range of sound absorption panels that feature LEDs behind the front textile layer of a Kvadrat Soft Cells frame. This project, developed in cooperation with Philips, delivers ambient lighting and the ability to add dynamic content in rooms with optimal acoustic conditions, meeting all building regulations and safety norms. The Luminous textile panels are available in custom or standard sizes, and are connected over Ethernet, allowing to manage the dynamic content and connecting the system to, for example, a building management system.

Modular Acoustic Wall Panels

Elements by Kvadrat Soft Cells is a simple, modular system of acoustic panels that help improve sound quality in any type of interior spaces. Built with the proven Soft Cells Broadline technology, the patented panels can be configured to provide Class-A sound absorption and contribute to increased well-being, motivation and productivity.

These panels are available in 15 sizes and a vast selection of textiles and colors, providing sound conditioning effective from 125 Hz to 5000 Hz. This is an ideal solution for spaces with limited free wall surface for acoustic regulation, and for lowering overall reverberation.

www.soft-cells.com

Acoustics First

New ArtDiffusor Nouveau Array

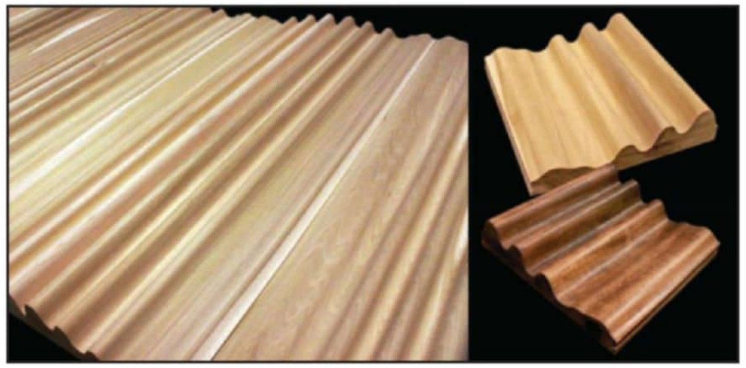
Acoustics First continues to expand its already significant catalog of acoustic solutions, with a special care for the diffuser panel category. The company's latest addition in this area is the ArtDiffusor Nouveau, a solid-wood quadratic diffuser by design, allowing the creation of large surface arrays. As the company explains, these diffusive boards expand its ArtDiffusor family of products and are the logical extension of the existing ArtDiffusor Trim line. In other ways, Acoustics First adds, they are also the logical successor to the Art Diffusor Model W.

These robust wood diffusive planks (2.75″ thick × 11.25″ wide, nominal) can be installed side by side to create 2′ × 4′ quadratic diffusers in both convex and concave configurations. Also, for architectural type applications, the boards can be installed in larger arrays. For larger applications, many pieces may be abutted to create a rolling pattern with both convex and concave quadratic effects in a single array.

Typical installations examples include its use as back wall diffusion in studios, recording and listening environments, architectural diffusion with installation over or behind a stretch-wall, chair-rail up or down, and so forth. The standard size of a plank is about 2.75″ × 11.25″ × 48″, however, longer planks can be provided if required.

This latest line of building materials from Acoustics First is stocked as natural unfinished Poplar wood, which can be finished in multiple ways, providing an excellent choice where great design aesthetics are a must. The Art Diffusor Nouveau panels are versatile diffusors with a robust acoustic performance and provide numerous options for customizing and optimizing acoustics for architectural and critical listening applications.

www.acousticsfirst.com



The ArtDiffusor Nouveau is an acoustically diffusive wood panel that allows for aesthetically pleasing, customizable installations.

Acoustical Solutions Update

GIK Acoustics

Impression Series Panels Offer Expanded Options for Acoustic Treatment

Acoustic treatment manufacturer GIK Acoustics can make a difference in room treatments with its decorative range of acoustic panels in the Impression Series—the latest product from the Atlanta, GA,-based company. As the name implies, the Impression Series delivers a visual statement to enhance any room, but more important, they are ideal for creating an even balance of low-end absorption without losing high-frequency presence. The rigid plates not only give the panels a stylish appearance but also help reflect/scatter high-frequency content in the room giving the proper balance needed. The panels use acoustic foam inside for sound absorption in the low-to-mid frequencies and the rigid plate cover delivers sound diffusion in the high frequencies. All panels are wall mountable using adhesive and there are additional mounting accessories and hardware options.

Impression Series panels are available in a variety of designs, from Bubbles and Checkerboard to Gatsby Arches, and more exotic looking designs, in square (24″ × 24″), narrow (12″ × 48″), and rectangle (24″ × 48″) sizes, with three different levels of absorption (2″, 4″, and 6″ thick). Top plates are available with different finishes.

Because the Impression Series panels perform well in any space, they are useful tools for taming room acoustics, improving speech intelligibility, and enhancing productivity in households. For more critical installations, including home recording and listening spaces, the Impression Series also offers a Corner Bass Trap model that GIK Acoustics adapted from its existing low-frequency control range. The Impression Series Corner Bass Trap has the same size and shape of the Tri-Trap Corner Bass Trap and still adds the scattering effects of the series' plates to effectively improve high-frequency balance and ambience. Designed to fit into any corner, they are made of a solid core of eco-friendly absorption material, wrapped in fabric, and finished with the same semi-reflective plate of the Impression Series.

In those more critical designs, users can also add the similar Alpha Series plate designs, which feature a mathematical equation to create one-dimensional or two-dimensional patterns, to ensure the reflected sound is truly random. In contrast, the Impression Series plates do not feature a mathematical pattern, but still offer scattering, which is effective at distances further than 3′ to 4′ from the panels and provide more design flexibility.

www.gikacoustics.com

Acoustic Sciences Corp.

IsoThermal Upgrade for TubeTraps

The Acoustic Sciences Corp. (ASC) TubeTrap is a hollow, cylinder-shaped bass trap that comes with a built-in treble range specular diffuser centered on the chrome dots. The result of work by Art Noxon, the company's founder, this patented corner-loaded bass trap/treble diffuser is one of ASC's most iconic designs. Now, the company is introducing the IsoThermal upgrade (patent pending) for the TubeTrap that doubles its bass absorbing power below 60 Hz.

The TubeTrap functions as an acoustic RLC series-parallel circuit. The surface of the cylinder is the acoustic Resistor and the internal air cavity is the acoustic Capacitor, similar to an electronic series RC circuit. The front half of the resistive surface R is covered with an acoustic inductor (L), which backscatters upper treble. The back half remains treble absorptive. The entire surface is bass absorptive.

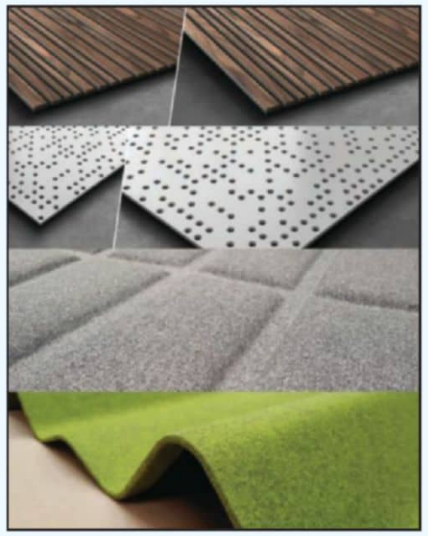
The air cavity is an air spring that allows air flow through the resistive walls of the TubeTrap in response to external acoustic air pressure variations, thereby absorbing acoustic bass energy. Adding the heatsink matrix into the air chamber is similar to adding dielectric (non-polarized) material into the air gap of an electronic capacitor. In both cases, the effective size of the capacitor is enlarged, which accordingly increases the current circulation in the resistor. This upgrade converts the adiabatic air chamber into an isothermal air chamber.

Adiabatic air warms and cools with variations in acoustic pressure while isothermal does not. The specific heat ratio of regular air is 1.4 but isothermal is close to 1.0, reducing its stiffness by about 40%, which allows more air movement in the resistor, increasing the resistive (I2 R) power loss. This effectively doubles (1.42 = 2.0) its absorption coefficient below 60 Hz. Above 60 Hz, efficiency rolls off because the pressure changes are faster than the thermal time constant in the matrix. Absorption efficiency is also increased by 30% throughout the muddy bass range 80 Hz to 250 Hz.

www.acousticsciences.com

Artnovion

Feature Walls with Acoustic Treatment Panels



The new Siena, Lyra 16.32, Altay and Tejo acoustic panels

Origin Acoustics, the US distributor for Artnovion—the manufacturer of high-quality, affordable acoustic products—introduced new acoustic treatment solutions that are aesthetically appealing and effective in both retail and residential environments. The new Siena Panelling, Lyra 16.32 Panelling, Altay Absorber, and Tejo Absorber solutions all have a simple discreet installation system that allows for quick mounting and easy maintenance.

Siena is a hybrid acoustic panel, combining a high-performance acoustic core with calibrated QRD diffuser. Hybrid panels decrease reverberation time while also improving sound field distribution—the perfect combination for wall-to-wall solutions to decrease reverberation time and improve speech intelligibility. The Siena panels are made from select sustainable wood and a recycled foam core and are available in four wood finishes and four lacquered finishes.

Lyra is a hybrid acoustic panel, combining a high-performance acoustic core with a binary sequence perforation pattern, tuning the absorption spectrum of the panel to focus on mid-range frequencies and efficient phase diffusion. This versatile panel has myriad applications, focusing on improving speech intelligibility, sound field distribution, and reducing general reverberation time. Lyra is available in three different perforations, each tuned to a certain frequency range, and four wood finishes and four lacquered finishes.

Altay is made from recycled PET and carefully wrapped in a recycled PET fabric. The molding process takes inspiration from traditional furniture making techniques, allowing the company to carefully fashion the product without the use of any additional chemical adhesives or high temperature presses—making an ecologically sustainable product constructed entirely from recycled waste. Altay panels are high-frequency absorbers, designed for spaces where speech intelligibility and noise control is a critical factor—ideal for home working spaces and conference areas. Altay is available in a range of select fabrics and finishes and is easily adhered to any surface.

Finally, the new Tejo Absorber is also made from recycled PET and carefully wrapped in a recycled PET fabric. Tejo panels were inspired by Artnovion's flagship Douro diffuser, creating acoustic performance in the minimum thickness possible. Tejo features variable air cavities behind the panel, boosting the performance of the sleek acoustic core. This high-frequency absorber was designed for spaces where speech intelligibility and noise control are critical factors and is available in a range of select fabrics and finishes.

www.originacoustics.com | www.artnovion.com

Artnovion

Effective Home Kits

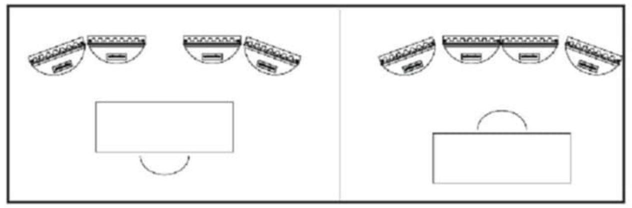
Artnovion is now offering a range of solutions for productive home offices, using some of the company's most advanced acoustic solutions.

The Mobile Home Office Kit uses only four mobile wall panels equipped with high-performance acoustic absorbers that can reduce the room's reverberation time and lower background noise—creating an acoustic buffer also from the rest of the room or house. This sort of environment not only benefits productivity and focus but also improves sound quality and intelligibility in conference calls and web-meetings.

Users who need to record audio can also expand this type of approach with Artnovion's Helen Mobile Wall 3.1. This portable solution, inspired by the company's pro audio range, allows users to create a more controlled acoustic environment for mixing or recording, solve challenges with existing doorways, and creating configurable rooms and variable recording spaces. The high-quality Helen absorbers are also a great solution for listening areas, to help control first reflections. The modular panels with an integrated frame system are quickly assembled using the company's patented FixArt and Tube Fixing System.

Finally, using Artnovion's advanced Siena or Eiger bass traps (see article documenting low-frequency absorption research by Jorge Castro, Nathaniel Bailey, Bruno Fazenda, and Kelvin Griffiths, published in *audioXpress*, August 2018), the company also created a Mobile Wall Bass Trap kit to help with low-frequency control in a solution that will be compatible with home environments.

Mobile Home Office and Mobile Wall Bass Trap kits from Artnovion



These modular, stackable, and portable acoustic panels are a flexible solution for acoustic control.

Installing these tunable membrane bass traps on freestanding supports enables users to place and move them anywhere in a room, avoiding screws or glue onto surfaces. Each kit can be assembled in two height configurations, by stacking two or three Bass Traps.

www.artnovion.com

NTi Audio

Room Acoustics Optimization Solutions

The Room Acoustics Reporter PC software from NTi Audio, in combination with the DS3 Dodecahedron Speaker and the XL2 Sound Level Meter, provides the perfect solution for acoustics evaluation of a room, performing reverberation time optimization calculations and providing comprehensive reporting according to ISO 3382 and DIN 18041 standards. Useful for anyone working on analysis and evaluation of room acoustics, the XL2 and the Room Acoustics Reporter software even allow simulation of acoustic treatment options with absorption data from manufacturers.

The reverberation time of a room is one of the most important parameters in room acoustics. The intended use of the room determines the reverberation time requirements. In a classroom, for example, depending on the size of the room, reverberation time should be between 0.5 and 1.0 seconds in order to achieve good speech intelligibility. This is often not the case, particularly in old school buildings, due to a lack of absorbing materials. In a concert hall, a reverberation time of between two and three seconds is desired to create a warm and rich listening experience.

For standard-compliant reverberation time measurements, a sound source with omnidirectional characteristics is needed. The sound energy must be radiated uniformly in all directions. The current building acoustics standards also require a flat frequency spectrum in the relevant frequency range. The DS3 Dodecahedron Loudspeaker fulfills these requirements while delivering high sound power and thus provides a perfect excitation signal for reverberation time measurements.

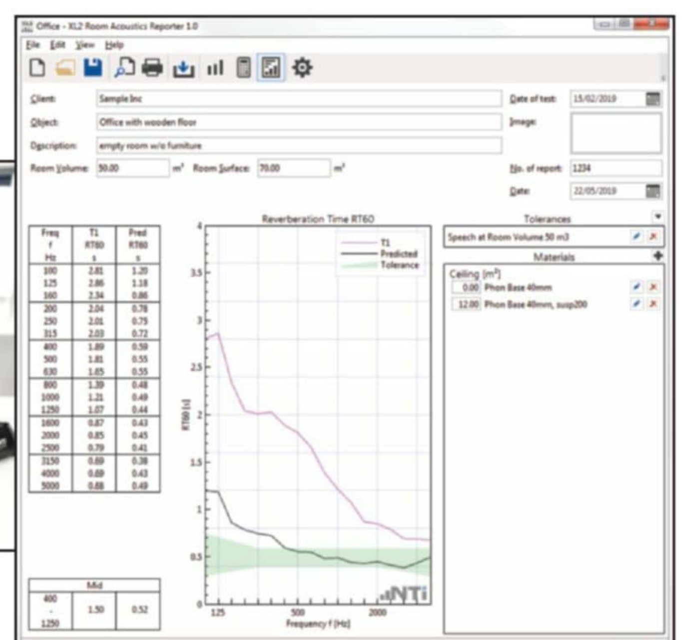
The NTi XL2 Sound Level Meter provides reverberation time measurements in octave bands from 63 Hz to 8000 Hz. Since many manufacturers of acoustic absorbing materials specify the material properties in octave resolution, the manufacturer datasheets and values measured by the XL2 fit well together. If a reverberation time measurement is required in one-third octave resolution, the "Extended Acoustic Pack" Option can be installed onto the XL2 at any time. The XL2 automatically triggers on an impulse sound source or a gated pink noise. When recording the reverberation times, it is possible to switch between T20 and T30; these two

types of analysis differ in the level reduction required to trigger a measurement.

Room Acoustics Reporter

The Room Acoustics Reporter is a PC software for creating reverberation time measurement reports. The RT60 test result can also be compared immediately with a tolerance band, which is specified according to the volume of the reverberant space and purpose of the room (e.g., in accordance with DIN 18041). If the measured reverberation time exceeds the tolerance, the software allows for simulation with additional absorbers. The corrected reverberation time is calculated immediately using the Sabine reverberation formula and shown in the graph. This makes it easy to verify what quantity and type of absorbers have to be brought into the room in order to meet the room specifications.

For the simulation calculation described, the sound absorption coefficients of absorbers from various manufacturers are available



NTi Audio's Room Acoustics Kit solutions combine the DS3 Dodecahedron Speaker set with the XL2 Acoustic Analyzer and the Room Acoustics Reporter software.

for download and import into the Room Acoustics Reporter. The list of absorbers is constantly updated as soon as new absorber data is made available by the respective manufacturers. Also, users can define their own materials for a simulation calculation.

The Room Acoustics Reporter software can be installed on any number of computers. A Room Acoustics Option must be installed on the XL2 as a permanent license – alternatively the cost-effective annual subscription service "Room Acoustics Reporter 365" may be enabled. The software, including sample measurement data, is available for download for registered XL2 users. Detecting reverberation times in one-third octave resolution requires the Extended Acoustic Pack Option on the XL2.

www.nti-audio.com

Acoustic Geometry

Sound Absorption In Creative Shapes, Configurations

Acoustic Geometry recently introduced its AG Geometric Wood Panels range. Cementitious wood fiber panels are created by combining wood fibers, Portland cement, and water, making them an environmentally friendly, cost-effective, sound absorbing, creative solution for any space requiring acoustic treatment and a unique design. Panels are available in nearly any size and shape, creating an effective, lower cost, and versatile solution for covering large surfaces in walls or ceilings and offering an effective sound absorber with multiple finishing options.

Acoustic treatment projects can benefit from the wide range of elemental squares, rectangles, circles, trapezoids or hexagons—the basic, standard options—and evolve with endless combinations for all sorts of patterns.

AG Geometric Wood Panels are Class-A fire-rated with low Volatile Organic Compounds (VOC) and can be mounted to walls or ceilings as a surface finish for optimal reduction of echo and reverberation creating an acoustically comfortable space. Standard panels are delivered with a clear lacquer finish but can be custom painted to complement the design aesthetic and vision for a space. A state-of-the-art manufacturing process ensures that panels are delivered with consistent thicknesses and dimensions with clean, square corners, producing a seamless result when applying the panels in a side-by-side custom design installation. As an eco-friendly, high-performing, and cost-effective acoustical solution, AG Geometric Wood Panels are positioned to differentiate new projects in commercial, residential, and educational acoustic applications.

www.acousticgeometry.com

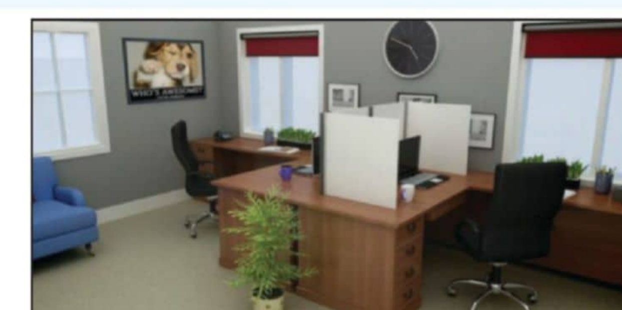
Acoustic treatment panels are available in many shapes, sizes and colors, great for large surfaces and ceilings.

Acoustic Geometry

SoundBlox Partition Package

A very useful solution, designed originally by Acoustic Geometry for offices and workspaces, is now being offered as a valuable partition solution for home offices, conference areas in the home, and creating study focus zones.

The SoundBlox Desk Partitions were a recent award winner in Acoustical Treatments category for commercial applications, but can also be a useful tool to create personal space that blocks out surrounding noises. The modular solution is now available in packages that include polyester sound absorbing partitions and hardware essentials, allowing anyone to set up multiple acoustically absorbent workstations for distance learning and working from home. The PolyMax acoustical partitions are non-allergenic, non-toxic, and contain no chemical irritants or formaldehyde. Panels can be printed with custom graphics to blend into any décor.



SoundBlox Desktop Partitions are perfect for personal workstations.

www.acousticgeometry.com

Acoustical Solutions Update

Sonitus USA

Make a Difference by Acoustically Treating Your Room

Sonitus USA, a division of MSR Acoustics, was founded by two industry veterans and leading experts in home entertainment and listening rooms and offers an expanded catalog of acoustic treatment solutions for any type of space and acoustical challenge. Behind the philosophy of Sonitus USA is Anthony Grimani, who has a career in the audio industry spanning three decades, several audio/acoustic patents, has held positions at Dolby and Lucasfilm THX, and is a highly respected expert in home-theater acoustics and design and consultancy. The second person is Manny LaCarrubba, who developed the principal acoustics technology of the Grimani Systems CinemaOne Audio Ensemble, the Conic Section Array waveguide. LaCarrubba's work in this area started in the mid-1990s, he then founded Sausalito Audio to commercialize patents related to this work.

The vast experience of Grimani and LaCarrubba provide Sonitus USA with a focused acoustics approach and pre-engineered room tuning systems, developed with trial and error in more than 1,500 home cinemas, listening rooms, and pro studio installations.

In the company's catalog, we can find all the required tools for those "recipes," combining diffusers, absorbers, and bass traps along with easy placement guidelines.

As the company explains in those guidelines, there is typically two to three times more reflected sound energy reaching our ears than direct sound from the speakers. "We hear the sound coming from the speakers because our brain takes directional cues from the direct sound. But all the reflected energy that's bounced off the walls sometimes distorts the beautiful direct sound. It stands to reason we'd want to do something about that, by controlling the reflected energy so that it works in concert with the direct sound."

And it's not OK to kill all the reflected energy! Research shows that only about 15% of the wall and ceiling surface area should be absorptive (plus a carpet or rug covering at least two-thirds of the floor). Absorptive materials should be evenly distributed around the room and of sufficient thickness and density to work over a broad range of frequencies.

Another 15% to 20% of the surface area should be covered with scattering materials to randomize sound and trick our brain into thinking that the room is bigger! Scattering materials should also be distributed throughout the room, with 2D hemidisc scattering toward the front of the room and 3D hemisphere patterns toward the back and on the ceiling.

Using the vast range of first-approach panels from Sonitus USA, it is possible to create proper listening rooms at home. Note the use of the Bigfusor II Massive diffusors.

Finally, the company focuses on low-frequency control. One of the most difficult disciplines in acoustics, to which Sonitus USA is able to respond effectively using different techniques including membrane absorbers, Helmholtz resonators, or a combination thereof.

As an example of the company's range of acoustic panels, the Bigfusor II Massive is a professional-grade, solid-wood (48 lbs) 3D diffuser, designed to randomly scatter sounds in multiple directions. Because of its curved design, besides diffusion, the reflection direction can be precisely controlled. It effectively randomly scatters sounds coming its way making the sound even over a wider area naturally and it provides a balanced approach when used with other 2D diffusers and various absorbers. It is available in white, ebony, natural, and mahogany finishes, and ships with its own wall mount system.

www.sonitususa.com

Odeon A/S

Version 16 of the ODEON Room Acoustics Software

Danish company Odeon A/S, the pioneer of ray tracing and hybrid models that combine particle and ray tracing used to simulate and measure room acoustics, announced a new version of its ODEON Room Acoustics software. The new version 16 of ODEON is packed with new features and enhancements, covering all four available editions: Basics (B), Industrial (I), Auditorium (A) and Combined (C). The update includes advanced tools for simulating and measuring room acoustics in a broad range of venues, from concert halls and opera houses to industrial facilities.

CAD compatibility has been greatly improved in ODEON 16, enabling users to import geometries with direct support for a large range of file formats. At the same time, the ODEON SketchUp plug-in has been further improved to allow for stable and fast import of complex SketchUp models.

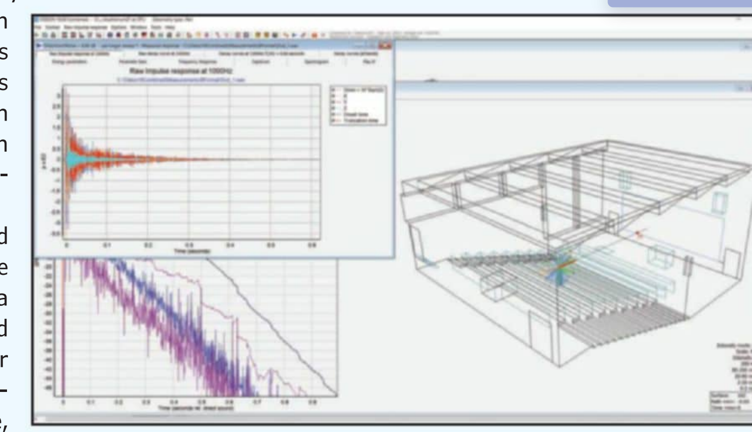
The improved measuring facilities include a new enhanced sweep measurement tool that allows better signal-to-noise ratio by customizing speed of sweep as a function of time, according to measurement hardware and room acoustic conditions. In editions A and C, measurements can now be performed using a first-order Ambisonics probe, for calculation of parameters like LF80, for studying intensity components of decay in 3D Hedgehogs and—not the least—for listening to 3D auralizations of measurements using a virtual dummy head.

A sweep spectrum correction can be applied in order to obtain similar decay ranges in all active octave bands. If a measurement has obtained poor decay range in some octaves, then ODEON can calculate corrections to the sweep based on that measurement. The result is a sweep where less measurement time is spent on the octaves with high decay range and more time is spent on the octaves where poor decay range was obtained.

But among the main highlights for ODEON 16 are the improved Reflectogram and 3D reflection-paths, active-sources display and improved Decay Curve, featured in editions A and C. The Reflectogram in the Single Point Response display enables users to have the selected reflection or all the zoomed reflections displayed in 3D, automatically updated when zooming or scrolling, translate the reflection currently visible. The reflection(s) can be colored according to time of arrival or to reflection order. The color scale can be selected by the user to be Auto, Music, Speech, or Studio.

The Decay curve tabsheet of the Single Point Response has been enhanced to display direction of arrival of intensity components, 3D hedgehog graphs, and allow tracing intensity components in the 3D View. The Decay curve for the selected band in the Single Point Response can be displayed next to the 3D View. This enables users to watch the direction of intensity peaks/slices inside the room model.

Also, many additions in the interface make it easy to edit positions of sources and receivers, read distances in models, manage multiple versions of materials in rooms etc. Last, a new 3D Matrix tool derives room acoustics parameters from all source to source combinations in the room, making it easy to study interaction between musicians in a concert hall or co-workers in an open-plan office.



The Play it! tool (in the last tabsheet of the Measured Impulse Response) makes it possible to listen to the measured impulse response as well as to listen to a signal convolved with the same impulse response. And when viewing a measured (or simulated) first-order B-Format impulse response, ODEON decodes the four-channel B-format file into a binaural impulse response, allowing play back of the impulse response over headphones as well as binaural playback of a signal convolved with this binaural impulse response.

Finally, a Material Archive functionality has been added to the Materials List. This enables users to restore previous material configurations (without losing the current one). ODEON keeps a list of the last 10 material configurations ordered by date and with an optional description.

ODEON is a very comprehensive software for room acoustics and version 16 adds a new level of usability. Given a 3D-model and materials (surface properties), any acoustics can be predicted, illustrated and listened to. Sound reinforcement is easily integrated into the acoustic predictions. The software, known for its easy workflow, fast calculation times, and high accuracy supports a valuable iterative (trial and error) design process, trying out different materials, sound sources, result options, parameters, and more. This is possible because of the fast calculations based on the strong algorithms of the software.

www.odeon.dk

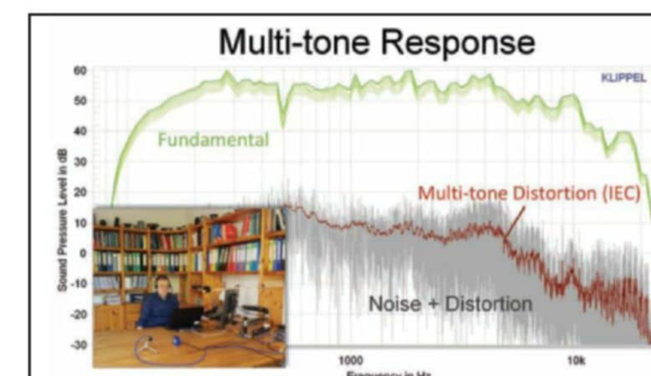
Klippel GmbH

Acoustical Measurement of Sound Systems

The Klippel Analyzer System has received another minor software update for both R&D and QC applications. The update is free for any users of dB-Lab major version 210 or QC 6, respectively. dB-Lab is Klippel's software that hosts the various modules of the Klippel Analyzer Systems as well as the Klippel Controlled Sound Technology.

The update provides new relevant tools for output-based testing of contemporary DSP-enhanced speakers, headphones, and other audio systems, according to IEC 60268-21. As demonstrated by Dr. Wolfgang Klippel in his free Klippel Live webinar series "Acoustical Measurement of Sound System Equipment according to IEC 60268-21," multi-tone-based testing plays a key role for the critical evaluation of SPLmax, compression and other important parameters of today's audio products.

In order to meet those requirements, Klippel released two new software modules: Multi-tone Measurement (MTON) for lab applications and QC Multi-tone Distortion (MTD) for efficient end-of-line testing.This all comes along with an entire batch of new test templates using the new MTON and other long-established measurement modules such as Transfer Function Measurement (TRF), 3D Distortion Measurement (DIS) or Near Field Scanning System (NFS) that specifically focus on taking advantage of the new IEC standard



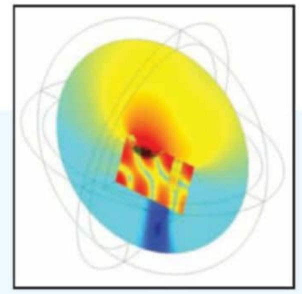
and helping users perform compliant measurements without hassle.

An informative webinar series is available on demand at www.klippel.de/know-how/education/webinars.html and is highly recommended for any professional working with DSP, audio systems or loudspeakers.

This update to dBLab 210 is rounded up with a full revision of the Suspension Part Measurement (SPM) and many smaller feature updates and minor bug fixes for tools like Rocking Mode Analysis (RMA) or Live Audio Analyzer (LAA).

www.klippel.de

Acoustical Solutions Update



Mvoid Technologies

Acoustical Virtual Product Development

Mvoid Technologies GmbH, the German company specializing in virtual audio product development, has released the latest Mvoid Methodology version 2.3. Mvoid Acoustical Virtual Product Development methods use a combination of standard modeling, simulation, and measurement software, combined with virtual reality and 3D visualization and binaural auralization tools for Automotive and Professional Audio applications.

Methods of acoustical virtual product development have been used for many years because virtual models reduce development time and optimize quality at the same time. For companies in consumer and professional audio, the question arises as to whether and how they can integrate virtual technologies into their product development and production processes.

The earlier these processes are used in the product development process, the greater the benefits. A sweet spot can be found by exploring the design space, which requires building, testing, and evaluating many models, however, the number of real prototypes is limited by time and costs.

In the past, adverse acoustic influences had to be manually adjusted or eliminated by acoustic experts, using physical prototypes. Today, these effects are quickly recognized and optimized in the virtual space. With a digital model, it is possible to explore the full design-space and become certain where the solution is positioned, providing a solid basis for a product roadmap. With the appropriate Virtual Reality tool and an auralization process, products can even be made audible before the first prototype exists.

Depending on the phase of the development cycle, there are different approaches. The most common approaches consider the evaluation in pre-development, evaluation of the acoustic concept and feasibility studies, and evaluation of the acoustic concept in detail engineering. A more complete description is detailed at: www.mvoid-group.com/developing-consumer-and-pro-audio-products-with-virtual-methods-to-improve-quality.

Mvoid is a pioneer and expert in virtual acoustics with the focus on simulation of all hardware in the audio chain. The company uses a process with advanced multiphysics simulations and combines modeling of all aspects of an audio system, culminating in real-time auralizations that allow actual listening evaluation and validation of design decisions.

With the latest Mvoid Methodology version 2.3, and its extension of a process level 6, introducing measurements of audio systems, Mvoid adds a new level of design and system validation. The Mvoid methodology for tier level six focuses on measurements of physical audio prototypes and products using a multi-channel acoustic measurement system fully integrated into VRtool, the company's own virtual tuning and auralization solution, where acoustic results can be auditioned over headphones. With this combined approach, Mvoid guarantees a smooth transition from virtual prototypes to physical prototypes, including allowing for the creation of a digital twin of an audio system for direct evaluation.

www.mvoid-group.com

Acoustics Vendor and Services Directory

acouStaCorp
www.acoustacorp.com

Acoustic Fields
www.acousticfields.com

Acoustics First
www.acousticsfirst.com

Acoustics in a Box
www.acousticsinabox.com

Acoustic Manufacture
www.acousticmanufacture.com.pl

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

Acoustic Geometry
www.acousticgeometry.com

Artnovion Acoustics
www.artnovion.com

Aural-Aid
www.auralaid.com

Auralex
www.auralex.com

ClearSonic
www.clearsonic.com

Crossley Acoustics
www.crossleyacoustics.com

Delta H Design
deltahdesign.com

Eccentricity of Wood
www.woodblocks.design

GeerFab Acoustics
https://geerfab.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

Isinac Acoustic World
www.isinac.com

Jocavi Acoustic Panels
jocavi.net

Media Specialty Resources - MSR Acoustics
www.msr-inc.com

Overtone Acoustics
www.overtoneacoustics.com

Performance Media Industries, Ltd.
www.pmiltd.com

Pinta Acoustic
www.pinta-acoustic.com

Primacoustic
www.primacoustic.com

Prosocoustic
www.prosocoustic.com

Ready Acoustics
www.readyacoustics.com

RealTraps
realtraps.com

Resolution Acoustics
www.resolutionacoustics.com

RPG Acoustical Systems
http://rpgacoustics.com

Silent Source
www.silentsource.com

Sonex
www.sonex-online.com

Sonitus Acoustics
www.sonitusacoustics.eu

Taytrix
www.taytrix.com

Vibrant Technology
www.vibetech.com

Vicoustic
www.vicoustic.com
www.focalnaimamerica.com/our-brands

VocalBooth
www.vocalbooth.com

Wenger Corp.
www.wengercorp.com

WhisperRoom Sound Isolation Enclosures
www.whisperroom.com

Acoustic Praxis

The Elephant in the Lab

By
Jim DeGrandis
(United States)

When specifying acoustic materials, it is common to consult the lab reports to analyze the performance information so we can calculate their effects on reverb time, clarity, privacy, articulation, transmission, and more. But with more advanced tools evolving to assist the design, modeling, and simulation of spaces, is this basic reporting giving us all the information we need? This article looks at some of the challenges we face moving forward—and why they will need to be addressed.

The Status Quo

Currently, a lab report summary will generally provide some very useful information for calculating and determining a material's impact on the acoustics of a space or its contribution to an assembly. These frequently come in the form of coefficients or some simplified numeric value, distilling all the measured results into a digestible, and hopefully useful, output. This could include whole octave or one-third octave frequency bands for absorption, in sabins or coefficients or both—maybe with a nice noise reduction coefficient (NRC) value to sum it all up. There are similar outputs for sound transmission, impact insulation, and other acoustic characteristics of materials and assemblies. We can plug these values into some equations, and voilà... we then know exactly how a space will sound with the inclusion of these materials.

Well, not exactly. For the professionals who cringed while reading that last sentence—you know the perils of absolutes. How many of you are running the calculations and absolutely guarantee the 20 Hz to 20 kHz performance of an "as built" environment? There are many things that can go wrong during the build, but there is a more fundamental problem: Do we even have enough information to calculate exactly how it will perform?

Not exactly. Why? Tests are performed in controlled environments, and therefore, the results are a best-case scenario. Test results are measurements of performance in a very specific scenario, which are distilled into values that are used to calculate performance. So, we are using information obtained in a known, controlled environment and applying it to different environments—where variables such as room volume, room geometry, source location, and more, interact to create a vastly different acoustic space requiring vastly more complicated calculations.

More importantly, you don't get all of the available data from a test. Remember that tests report very specific, processed data. Maybe you get one-third octave sabins, or one-octave Sound Absorption Coefficients, or a Transmission Loss graph... but, while all of this information is useful in its own way, we should take a step back and look at the big picture. Is having an NRC enough information to provide an exact prediction of how a material will perform at all frequencies, in all environments? Of course not.

Are one-third octave sabins or Sound Absorption Coefficients enough to do the same? Unfortunately, also "no"—but that's only part of the problem.

The Unspoken Truth—Repeatability & Reproducibility

Theoretically, if you take a single sample of material to different labs—tested with the same test—you should receive exactly the same result. This misconception is called Reproducibility, and is the first unspoken truth... or perhaps... the elephant in the lab. Deviation from lab to lab is the rule. Labs that are accredited, voluntarily participate in round-robin evaluations, where they all run the same test on the same material. Those results are then graphed, and the range of values is defined and analyzed.

Notice, I said range... variability and deviation are the rule. Why?

The answer is related to Repeatability. It's assumed that if you run the same test, in the same lab, you'll always get the

Sample Test Report - Absorption

Sound Absorption Data

The measured Sound Absorption [in units of area] and Sound Absorption Coefficients of the test specimen at the preferred one-third octave band center frequencies are tabulated below.

1" Composite, "A" Mount

1/3rd Octave Band Frequency	Sound Absorption (m^2)	Uncertainty (+/-)	Sound Absorption Coefficient	Uncertainty (+/-)
125 Hz	0.1	0.6	0.02	0.1
160 Hz	0.8	0.5	0.13	0.08
200 Hz	1.5	0.4	0.25	0.06
250 Hz	3.1	0.2	0.53	0.04
315 Hz	4.5	0.2	0.76	0.03
400 Hz	5.3	0.2	0.91	0.03
500 Hz	5.7	0.2	0.97	0.03
630 Hz	6.5	0.2	1.11	0.03
800 Hz	6.5	0.2	1.1	0.03
1000 Hz	6.5	0.2	1.1	0.03
1250 Hz	6.3	0.2	1.07	0.03
1600 Hz	6	0.2	1.02	0.03
2000 Hz	5.8	0.2	0.99	0.03
2500 Hz	5.7	0.2	0.97	0.03
3150 Hz	5.7	0.2	0.99	0.03
4000 Hz	5.8	0.2	0.99	0.03
5000 Hz	5.9	0.2	1	0.03
NRC	0.9			

During the test, environmental conditions in the reverberation chamber were 26.7C and 68.1% relative humidity. The precision values [±] tabulated above represent 95% probability that the true mean value lies within the stated range.

This example report shows the 1/3rd Octave Center Frequency output from a standard ASTM C423 test. Take note that the Uncertainty (+/-) range for the Sound Absorption Coefficient is 0.06 @ 200 Hz - this indicates that there is a 95% chance that the true value falls between 0.19 and 0.31 (0.25 +/- 0.06).

Also, observe that the lower frequencies have a *higher* uncertainty due to the variability in test result performance at those frequencies.

same result. This isn't true. It's just the way it is, and is another reason why there are issues with the absolute accuracy of measurements and material performance databases.

If you completely read a lab report, it will sometimes state the accuracy or Certainty of the measurement. It will many times provide an Uncertainty (±) value, which indicates a range where you presume the value falls. Now, why do I say presume?

This certainty will have a Confidence Percentage based on the variability of different measurements, like the round-robin. This confidence is usually 95% or 95.45% (2-Sigma when graphing the results). This means if an accredited lab runs this test, it will, with about 95% confidence, get a value that falls within a certain range, around the stated value.

So, take this example: Let's say you get a Sound Absorption Coefficient of 0.25 at 200 Hz, but your Uncertainly (±) Value is 0.06. You can be 95% confident that the true value falls somewhere between 0.19 and 0.31—with a 5% possibility that it won't.

If the controlled, best-case-scenario, laboratory environments are only this accurate, perhaps we shouldn't assume absolute accuracy in our real-world predictions.

What is the Future?

With all of these inherent problems in measuring and reporting data, how can we improve things? There will always be Uncertainty issues, Reproducibility & Repeatability issues, and issues with accuracy. Accepting this fact and trying to minimize it is a good start.

The International Organization for Standardization (ISO), ASTM International (formerly known as American Society for Testing and Materials), the Audio Engineering Society (AES), and other organizations are constantly evaluating and refining their current methods to increase accuracy by improving repeatability and reproducibility, which narrows the Uncertainty value. There is also research by other institutions to develop new methods that may prove to be more accurate or provide more information.

About the Author
Jim DeGrandis is a Research & Development Engineer at Acoustics First Corp., a member of the Acoustical Society of America (ASA), and he works with ASTM International on researching new acoustic testing methods. DeGrandis frequently lectures about acoustic phenomena, simulation, and architectural acoustic design.

More Data!

Remember the one-third octave data report? What happens if your problem frequency isn't exactly one of those values? Or even worse, what if you are looking at low or high frequencies that aren't even reported?

Depending on the test, it may say something like "measurements are made in the ISO-Preferred one-third octave bands from 100 Hz to 5000 Hz...," which means that frequencies outside of that range aren't measured or represented.

There are many reasons for this—you need large labs for accurately testing low frequencies, and simple things (e.g., humidity) have a huge effect on high-frequency propagation. However, there are people demanding more accuracy, and for good reason.

Simulation and Modeling

Simple hand calculations and spreadsheets are giving way to higher performance acoustic models and simulations. The input into these models is material data derived from these standard test methods.

With various materials in an environment and incomplete or inaccurate test data for those materials, there are many variables that are limiting these simulations.

The future of acoustic material testing will address the colossal task of creating more accurate and granular data at a wider frequency range. This data will inform the models and simulations, allowing them to more accurately represent the performance of acoustic spaces.

Perhaps with the advent of inexpensive storage and bandwidth, all of the raw measurements will be made available to the end-users for analysis. There is great value in that wealth of data, and it may contain the key to closing the accuracy gap.

Eating the Elephant

A great metaphor (which I'll paraphrase poorly and credit to Michael Vorländer) is:

"Tackling all of the obstacles in creating accurate acoustic simulations is like trying to eat an elephant, which can only be approached one way—one bite at a time..."

"...and having accurate material performance data is only one bite."

Bon Appétit.

The Internal Life of Vacuum Tubes

Every once in a while, Jan Didden ventures into the world of thermionic devices or accessories for those creatures. A few years ago, he designed a high-voltage delay unit and shared its details in a 2019 *audioXpress* article.[1] The unit will delay the high-voltage for a tube amp until the heaters are at operating temperature. Many people built it and like it, but others said that it would be totally unnecessary. As so often happens in audio, arguments pro and con raged without a clear resolution. So Didden thought this was something he needed to get his head around. He found several sources, which he thought were credible, and this article is the result of his investigations, fed by those sources.

By
Jan Didden
Technical Editor, audioXpress

Author's Note: Before I begin, I want to detail some of the resources I used for this article. Among my sources for this article, is Noël van Moselvelde, who was a guest at the Audio Discussions #4, which was sponsored by Audio & Techniek [2]*. van Mosselvelde had been responsible for tube reliability and quality control at the tube factory of Philips in Eindhoven, The Netherlands, and was considered an authority on tube reliability and life span. Morgan Jones, author of* Valve Amplifiers [3] *and* Building Valve Amplifiers [4]*, should not need an introduction in these circles—his books on tube design and tube amplifiers should be household items for anyone interested in the matter. I also obtained information from Robert B. Tomer's* Getting the Most Out of Vacuum Tubes [5]*, a book from 1960 that has recently been reprinted by* audioXpress' *parent company, KCK Media. A mini review of that book can be found in the sidebar to this article.*

We all know that there are tubes with directly heated cathodes, and those with indirectly heated cathodes. Indirectly heated cathodes have a separate heater structure, often inside a cylindrical cathode, to heat the cathode. The cathode, generally a nickel tube, is covered with several chemical substances like barium and strontium carbonate. These chemicals are chosen for their ability to emit free electrons at a relatively low temperature, to limit the (losses from) heater power. Combining heater and cathode with all the various compounds in a directly heated cathode structure is, in the end, less effective to emit free electrons than an indirectly heated construction, and will require higher heater power.

The indirect heating is a complex process with chemical reactions and the "number" of free electrons is limited by the end products of the reactions, and when depleted, the tube is at the end of its life. The other mechanism here is increasing interface resistance between the cathode emissive surface and its nickel structure, which will bias the tube.

Heater Voltages

Heaters have a nominal voltage specification, such as 6.3 V (or the 12.6 series) for many dual triodes. How critical is that voltage? What happens when we put it at 6 V? Or 6.5 V? The first obvious effect from too-high heater voltage is that the cathode gets hotter and the emissive material will be reacting faster and evaporating quicker, lowering the tube's effective life span. Lowering the heater voltage, and thus, the cathode temperature too much will shift the tube's characteristic down to a lower saturation current and increase distortion. That may not be a problem with low power (pre)amp tubes, but it might be an issue with power tubes.

Also, lower temperature cathodes are more sensitive to what is generally called "poisoning" by gas ions present in the tube. Higher (nominal) temperatures provide some protection against poisoning—although it also increases noise. The upshot is that it is inadvisable to lower the heater voltage too far in an attempt to extend tube life even with low anode currents. Nevertheless, lowering heater voltage of a nominal 6.3 V tube to 6.1 V is said to increase tube life by 50% (see **Figure 1**). [3,5]

DC heater power can easily be provided with modern rectifiers and integrated voltage regulators, and it is usually done to avoid power line hum and noise being injected in the amplifier. But DC heater power in itself is detrimental to tube life. At switch-on, the large inrush currents and the mechanical spiral construction of the heater always cause a very small mechanical movement of the heater structure inside the cathode. With a DC supply, these movements are always in the same direction and of similar magnitude, causing a kind of scraping. Over many switch-on cycles, this scraping can lead to weakening of the isolation between the heater and the cathode, causing crackling sounds and eventually failure. So unless you have an issue with hum, AC powering the heater is better for tube life.

Most heaters today are run from a voltage source, (6.3 V or 12.6 V are the most common) but sometimes tube heaters are run in a series loop, being current-driven. The older Plxx/PCLxx TV-tubes, for instance, could be run in series directly from the mains, saving a heater transformer. But be careful: Not all heaters have the same thermal mass and/or resistance, so they heat up at different speeds. Unless you carefully balance the loop circuit, it is possible that one heater could heat up more slowly than another, meaning its resistance stays lower than another, and the voltage across the fast-heating one might rise considerably beyond its design value, decreasing tube life.

Last, there is the mains voltage to consider. Both in Europe and the US, mains voltages have steadily risen over the last few decades. In Europe they have risen from 220 V to 230 V or even 240 V, in the US from 110 V to 120 V or even 125 V. That means that, when using vintage power- or heater supply transformers, heater voltages could be, as a matter of routine, 5% to 10% over the nominal value. Needless to say, this will significantly lower tube life expectancy. Use the transformers specified for contemporary mains voltages, or keep the heater voltage in check with small series resistors, if needed.

Although heater failure is far from the most common cause for tube failure, it is still a potential issue and should be considered in tube amplifier design.

Dissipation and Thermal Management

You might think that when the tube has an appreciable dissipation, such as a power tube in Class A, that anode dissipation would also help to heat the cathode so maybe you need less heater power. In principle that is true, but the effect of anode temperature rise on cathode temperature rise goes by the fourth root. So a 100° anode temperature rise effects only has a little more than a 3° rise on the cathode.

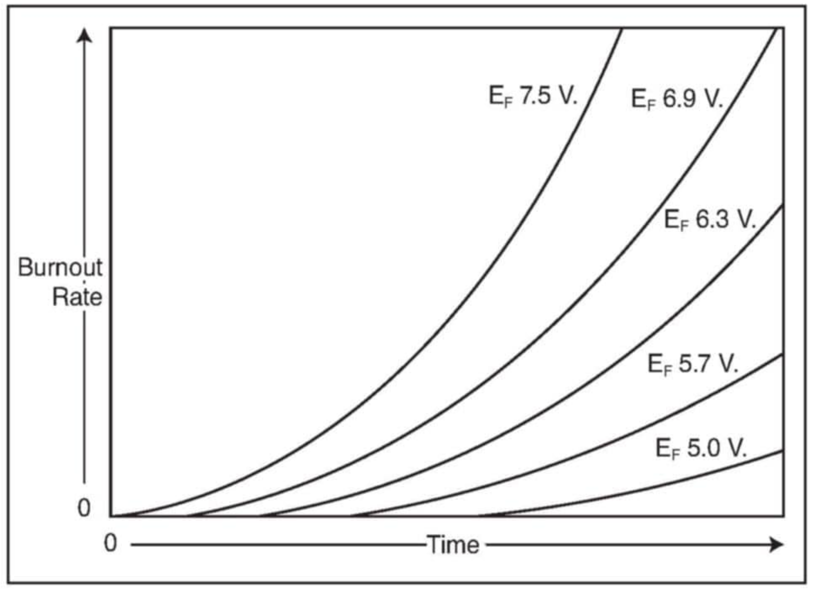


Figure 1: Increased heater voltage over the design value decreases tube lifetime.

On the other hand, cooling the tube or making sure that the environment in which it works doesn't get too hot has a positive effect on tube life. There are two factors at work here. One is the glass envelope temperature: Glass will always emit gas molecules, which lower the internal vacuum of the tube and may also increase cathode poisoning. Glass out-gassing rises exponentially with temperature so limiting the glass envelope temperature will have a large positive effect on tube life. Higher glass temperatures also lead to higher temperature gradients at the base pins or top cap, increasing the risk of micro-cracks and leakage (see **Figure 2**).

The other is the anode temperature. A very hot anode can lead to water vapor being released from the mica parts of the tube structure, and this will poison the cathode as well. Making sure that an operating tube has adequate air circulation around it promotes tube life significantly.

Heat It, Then Run It

A point of much discussion is whether it would be a good thing to delay the application of high voltage until the heaters have had time to heat the cathode. There are several mechanisms at work here. In normal operation, the heated cathode emits a cloud of electrons that bunch together at the cathode structure and this cloud is known as the space charge. With anode voltage applied, electrons are attracted to the anode and leave the space charge, which is supplied with new electrons from the cathode. But when the cathode is cold, the space charge is minimal and is rapidly depleted when the anode voltage is present. Normally, some electrons collide with stray gas molecules, producing ions that are attracted to the cathode. Ions are much more massive than electrons and when they strike the cathode, they can damage the emissive structure and cause small craters—further decreasing the emissive capacity. In normal operation, the space charge will repel these ions but as mentioned, with a cold cathode the space charge is minimal and has much less capacity to limit the ion bombardment on the cathode.

Another phenomenon with a cathode not yet at operating temperature concerns unequal cathode temperature. As the cathode heats up, the temperature will not be uniform and the hotter areas will allow more current than the cooler areas. The current that flows will tend to concentrate in small hot areas, and that will increase the current density to higher values than will be the case at normal operation. That higher current density can locally damage the cathode creating miniature craters in the structure, which overall will have a negative effect on life expectancy.

All these factors suggest that the cathode should be at the operating temperature before the anode voltage is applied. [1] This is especially important when the equipment uses solid-state rectifiers, in which case the anode voltage will be present almost immediately at turn-on. Although there are many amplifiers out there with solid-state rectification that work reliably for many years, it is ultimately a factor in tube reliability deterioration, so a delayed high voltage is a good measure, especially with those more expensive tubes!

Sometimes you read advice that says to keep the heaters on at say half power, to avoid the cold starts, and to avoid high heater inrush current at cold start. Tomer recommends that to significantly extend tube life, although there is of course the issue of higher stand-by power consumption, especially with large installations. [5] But make sure that you don't actually run the tube with anode voltage and half heater power at the same time—that will surely decrease tube life!

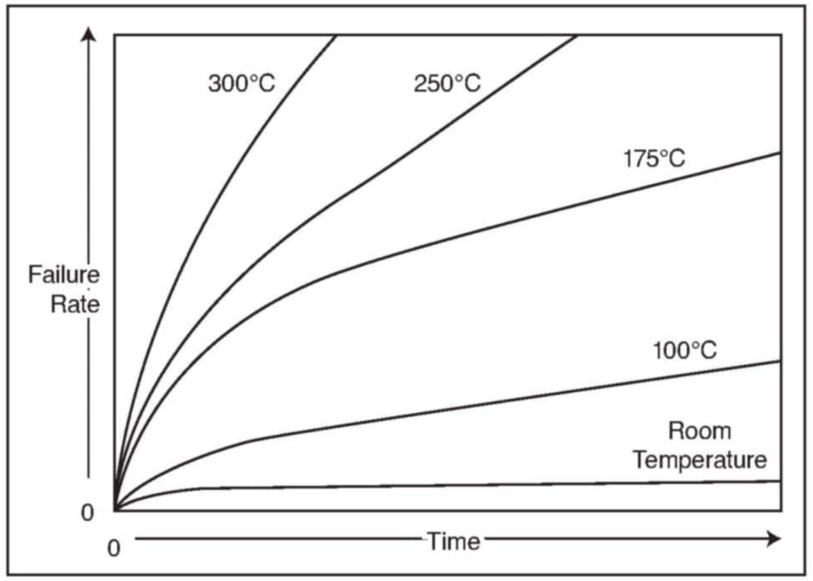


Figure 2: Insufficient air circulation heats up the tube envelope and drastically decreases tube lifetime.

About the Author

Jan Didden has written for *audioXpress* since the 1970s and he is the magazine's Technical Editor. He is retired following a career with the Netherlands Air Force and NATO. He worked in logistics, air defense, and information technology. Retirement has provided him with the time to finish all the audio projects that have piled up for decades. He writes about them on his website linearaudio.nl. Didden is also the publisher and managing editor of the twice-yearly bookzine *Linear Audio*.

The reverse situation—a heated cathode but no anode voltage to draw cathode current—is equally ill advised. In such a situation, an oxide is formed between the nickel cathode and its emissive layer, generally an oxide of barium and/or strontium. This oxide eventually becomes high resistance, decreasing emission. But it also increases the tube's shot noise, which can be considered in series with the input signal. The process is called cathode poisoning. In the days of picture tubes, such poisoning was often removed (or at least attempted to be removed) by briefly running the tube with higher-than-normal heaters and a higher-than-normal anode current, in a process called rejuvenation. But such rejuvenation is generally short-lived.

Superficially, you'd think that a directly heated cathode would be the best solution, as there is no need for a fiddly construction of heater filaments being coiled and rolled up inside a nickel tube. But consider the following: The necessarily fine heater wires have relatively low thermal mass and if you heat them with AC, their temperature would be modulated, ever so slightly, by the line frequency. That carries the risk of hum induction at twice the line frequency as the heating and cooling cycles are related to V^2. The heater wires may also, depending on construction details, produce both electromagnetic and electrostatic fields that might modulate the electron stream toward the anode, again risking hum generation.

The solution for these problems is, indeed, coiling the heater inside a tubular cathode structure. It increases the thermal mass of the assembly, limiting thermal mains modulation, and it provides a degree of electrostatic and electromagnetic screening. Thus, the indirectly heated cathode is born.

Heater-Cathode Voltage Difference

In some circuits, such as Series-Regulated Push-Pull (SRPP) or cascode designs, there can be a large potential difference between (one of) the cathodes and its heater. This is always a negative factor, even without a hard short causing flash-over. The isolation between the heater wire and the cathode will be lowest with the high temperature of normal operation, and there will always be a certain amount of leakage between the two. The maximum cathode-heater voltage specification is related to how much leakage is allowed, so it is somewhat arbitrary. In any case, higher cathode-heater voltages and increased leakage lead to higher noise levels.

When you heat with DC, the polarity of the voltage difference between the heater and the cathode is important. There are two (conflicting) requirements. If the DC heater is negative with respect to the cathode, electrons emitted by the heater (and there are always a number of them) will flow to the cathode, upsetting the anode-cathode current balance. If you make the heater positive with respect to the cathode, electrons emitted by the cathode on the inside of the cathode tubular structure will flow to the heater. This will transport material from the cathode to the heater and can lead to a metallic "bridge" between the two and in some cases to a short. But even without this effect, it will, in general, increase cathode-heater leakage and tube noise. For the highest reliability with DC heater power, the general advice is not to make the heater positive with respect to the cathode; yet,

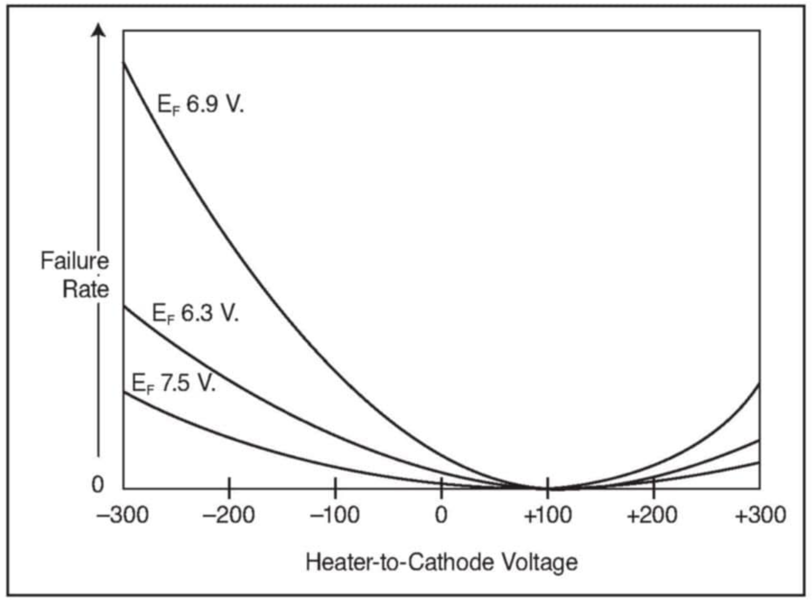


Figure 3: Increasing Vkf above approximately 90 V decreases tube lifetime.

some RCA data books recommend that you make the heater positive with respect to the cathode, ideally by 40 V or so, presumably to minimize noise. So as with everything in life, there are trade-offs. With an AC heater, the polarity between the heater and the cathode reverses with the mains frequency and that appears to ameliorate some of the effects mentioned earlier. In any case, unless you have insurmountable hum issues, it is best to run the heater on AC.

One way to avoid or limit any of the mentioned parasitic currents is to return the heater to ground, or whatever point is selected, via a resistor, to make the leakage (or in case of a short, the shortage) current loop high impedance. The old picture tube datasheets often recommended grounding (or biasing) the heater through a 1M resistor, but in audio amps it can be (much) smaller (see **Figure 3**).

Coming back to the maximum allowed voltage between cathode and heater, obviously you want to keep it below the datasheet value to avoid flashovers or shorts. But even with voltage differences as low as 30 V and heaters on AC, there can be ionization of remaining gas molecules inside the tube, deteriorating the isolation between cathode and heater—leading to shorts in the long run. If there's enough current capacity in the circuit, you can get a heater that is "stuck" to a cathode, resulting in a defective tube. As mentioned earlier, there will be an increase in leakage between the cathode and the heater and increased tube noise long before there is an actual short. It seems a long-term effect though, so only of importance if you want to eke maximum lifetime out of your tube. Although stories abound that a mild mechanical shock to such a tube can separate the heater from the cathode, I have not been able to confirm that.

Grid Current

Power tubes are often operated with positive Vgk, which cause grid current to flow. As long as the tube is designed for that and the drive circuit is low impedance, that'll be fine. (If the drive circuit is not low impedance, the grid current will cause distortion of the drive signal).

Sometimes, you might want to use small signal tubes (e.g., the ubiquitous dual triodes), with very low supply voltages, 50 V or less, and consequently very low Vak. In those cases the Vgk gets very low too, and even when it doesn't actually become positive, grid current can start to flow at Vgk more positive than -1 V. You can do that, if you keep the grid current low, but you need to condition the tube for that. The grid surface has all kinds of foreign material, and when grid current starts to flow, this material will be "burned off" so to speak. That means that the characteristics will change during the initial period with grid current. This is one case where burn-in is useful!

Fixed vs. Autobias

Seemingly having no relationship per se to tube life, the choice for biasing method nevertheless has an impact on the probability of tube failure. If a circuit fault develops in a fixed bias circuit, tube current may run away unchecked, leading to tube destruction. With auto-bias, there is always the cathode resistor, providing some amount of negative feedback to limit tube current.

A similar issue can arise with fixed screen bias. If in such a circuit the anode voltage collapses due to some failure, the screen will probably start to run red hot, causing destructive gassing, warping of elements, and finally tube failure.

Summary

The above is by no means an exhaustive treatise on tube reliability factors, but it gives an overview of often-discussed issues. If you want to know more about factors determining tube life, Tomer's book would be a good read. Audio Discussions #4 also has many tips and tricks related to tube life throughout the book.

To wrap up the most important points:

- High voltage should preferably be switched on only after the cathode has reached operating temperature, and this is especially important when using solid-state rectifiers.
- To avoid higher anode voltages at switch-off, it is recommended to first switch off the high voltage, then after discharge of the high voltage capacitors, switch off the heaters. This may depend on the actual equipment, possibly thermal inertia of the anode structure is enough to discharge the high voltage capacitors while the anode current peters out. Here is one case where excessive rectifier filter capacitance is a disadvantage.
- The preferred heater supply is AC, unless there is an insurmountable hum issue. Heater voltage should be nominal or a bit lower like -5% for maximum tube life. Consider running the heaters at 50% in standby, never turning them off, to avoid turn-on current surges and insure quick turn-on of the equipment.
- Make sure that an operating tube has sufficient air circulation around the envelope to avoid high temperatures. The tube will get hot, but the surrounding air should not, and should be able to flow unobstructed.
- Keep an eye on heater-cathode voltage differences—even below the specified limit, leakage and noise may increase.

If you do switch the heaters completely off, consider limiting heater turn-on surges with an NTC resistor or something similar. But consider where to mount it—an NTC under the chassis may generate enough heat to cause reliability issues on itself.

When debating the use of fixed or auto bias for the grid and the screen, also consider the impact on equipment reliability.

Author Acknowledgements: I wish to thank Stuart Yaniger, tube aficionado and prolific contributor to audioXpress*; and Morgan Jones, purveyor of fine books on all things tube, for their advice and for keeping me on the straight and level. Valuable tips and information were also provided by Menno Vanderveen of ir. bureau Vanderveen, and by Guido Tent of Tentlabs and Grimm Audio. I thank Guido Tent for making Audio Discussion #4 (Reference 2) available. Any remaining errors in this article can safely be attributed to my stubbornness.*

References

[1] J. Didden, "Tube High-Voltage Delay," audioXpress, October 2019, https://audioxpress.com/article/you-can-diy-a-high-voltage-delay-for-tube-amplifiers-the-sequel

[2] "Audio Discussions #4, *Audio & Techniek*, Rotterdam, The Netherlands, 1991 [The journal *Audio & Techniek* no longer exists, but some of the participants in this discussion are still active in audio: Guido Tent, of Tentlabs and Grimm Audio; Eelco Grimm of Grimm Audio; and Peter van Willenswaard of Audio Magic, to mention a few. Quoted with permission.

[3] M. Jones, *Valve Amplifiers*, 4th Edition, Newnes 2012.

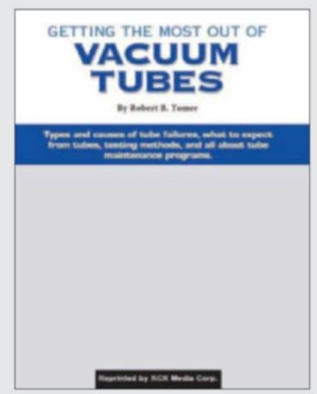
[4] M. Jones, *Building Valve Amplifiers*, 2nd Edition, Newnes, 2014.

[5] R. B. Tomer, *Getting the Most Out of Vacuum Tubes*, Howard W. Sams & Co., 1960, reprinted by KCK Media Corp., 2019, available at www.cc-webshop.com.

Book Review

Getting the Most Out of Vacuum Tubes

Review by Jan Didden



I ordered this book, *Getting the Most Out of Vaccum Tubes*, from KCK Media Corp. while I was researching tube service life issues. It is an unusual book, not discussing what you can do with tubes, how to use them in circuits, operating parameter calculations, or that sort of thing. Instead, it is solely devoted to factors that decrease useful service life or cause catastrophic failures. Its subtitle, aptly, is "Types and Causes of Tube Failures, What to Expect from Tubes, Testing Methods, and All about Tube Maintenance Programs."

Yes, and you'd be surprised how the wrong type (or too much) maintenance activity on tube equipment can actually damage individual tubes or decrease their service life. One example: I know of audiophiles who regularly "re-seat" their tubes in an attempt to make sure the pin contacts are OK. Robert B. Tomer, the book's author, advises against that: If it works, leave it alone. Re-seating will put stress on the pin-glass interface, leading to micro-cracks and air leaks, hastening tube demise. A corollary: When inserting tubes for the first time, use a pin straightener first, rather than forcing the pins to straighten themselves by wriggling them into the socket.

Tomer talks about different types of failures. The most obvious is of course catastrophic failure, the tube just stops working. But degenerative failures, where emission slowly decreases, leakage increases and the cathode gets depleted are just as serious although often not identified until in an advanced state. Then there are subjective failures—the tube happily continues to work but develops hum, noise, or microphonics. This book is about how to avoid those failures or at least how to postpone their occurrence.

A chapter I found particularly interesting is about the spread in characteristics of real tubes vs. the datasheet values. It is very insightful, to read how the datasheet values are set, and what sort of variation to expect from the tubes you just bought.

There also is a chapter on tube testers, showing that there are many different types and that the measurements are not always correct or the same between testers. He tested 100 tubes with several commercial tube testers of the day. Tested in the lab, 95% of the tubes were better than the datasheet indicated, but one Tube Tester reported 50% of the lot as below the datasheet values!

One factor you wouldn't expected to be related to tube life: fixed- vs. auto-bias. Fixed bias has a greater chance of failure because if anything in the circuit goes wrong, there is no degenerative mechanism like a cathode resistor to counteract possible runaway current. Do you take that into account in your design? Being aware of it lets you make the trade-off. Finally, two chapters discuss "Predicting Tube Performance" and "Methods for Lengthening Tube Life."

So there you have it. An unusual book, but in an era where much-sought-after tubes can cost several hundred dollars each, it is a book that should be on the shelf of anyone who is serious about tube equipment. And it is available for less than the price of a medium-size pizza.

Getting the Most Out of Vacuum Tubes
By R. B. Tomer
Howard W. Sams & Co., 1960
Reprinted by KCK Media Corp., 2019.
160 pages
$14.95 | www.cc-webshop.com

The APR-17 Aperiodic Loudspeaker (Part 1)

Starting the Build

By
George Ntanavaras
(Greece)

Photo 1: The completed APR-17 loudspeakers

In this two-part article, George Ntanavaras shares the details of his latest project—the APR-17 loudspeakers. The design features a compact, two-way, stand-mounted loudspeaker consisting of a 6.5" (17 cm) mid-woofer and a 1" dome tweeter. Its main feature is the "aperiodic loading" of the woofer, which is implemented with an open vent filled with resistive material on the enclosure's front baffle.

Aperiodic is a word of Greek origin and means "not periodic" or "without a period." According to the *Oxford Dictionary* aperiodic is "a potentially oscillating or vibrating system that is damped to prevent oscillation or vibration." This comes from the fact that the resonance frequency (period is the inverse of the frequency) of an aperiodic system is so well damped that no resonance frequency seems to occur.

Probably the most famous aperiodic loudspeaker is the legendary Dynaco Model A-25. It was introduced in 1969 and was one of the most popular loudspeakers ever produced, with more than 1 million units sold. It was highly acclaimed by audiophiles of the day, and received positive reviews in consumer magazines as well as in the high-end audio press. It was a two-way, two drivers loudspeaker with the 10" extended excursion woofer aperiodically loaded. The woofer was matched to a special designed soft dome tweeter driven by a high-pass crossover filter at 1500 Hz.

Dynaudio from Denmark was also another supporter of this concept. For many years, it manufactured a device called "Variovent," which was a contained tightly packed fiberglass stuffing held in place by a plastic grill and frame. According to the description provided in the datasheet, "the Variovent is flow resistant, damping the resonance like a DC-resistance in the oscillating circuit, which results in a more precise bass response and better woofer quality. The impedance maximum at the resonance point is reduced by at least 50% compared to a sealed cabinet. Consequently, the amplifier is able to give more power in the lower range. The oscillation of the cone after a strong pulse is aperiodically damped, producing a more clear and well-defined bass response."

Another Danish firm, Scan-Speak manufactured a similar device. According to the description, it was an air flow resistant, "aperiodic" damping device, which is very useful in sealed box applications. It worked by reducing the maximum impedance at the box resonance by at least 50%. This resulted in clearer, better defined bass, with more amplifier power and control into the lower frequencies. It also allowed the use of a driver in a smaller than optimum sealed enclosure. The vent was mounted to the cabinet back wall in a 110 mm diameter hole. About 60% of the cabinet volume was filled with damping material, with an open path from the woofer's rear side to the vent device.

The above words for "precise bass response," "better woofer quality," and "better drive for the power amplifier" were the triggers for me to initiate an investigation into this loudspeaker. My resulting pair of the APR-17 loudspeakers is shown in **Photo 1**.

The Drivers

Before I started to design the loudspeaker, I had to select the drivers. For the woofer, I selected the Visaton W170S4, a 6.5" coated paper cone driver with a ±10 mm excursion and a Q_{TS} suitable for a closed box. According to the manufacturer, the Thiele-Small (T-S) parameters are the following:

f_s =36 Hz, Q_{TS} =0.41, and V_{AS} = 38 ltr

For the tweeter, I selected the SEAS 27TBCD/GB-DXT-H1499, an aluminum/magnesium alloy dome tweeter with a DXT acoustic lens technology to optimize the directivity. It has a fine mesh grid that protects the diaphragm and a rear chamber with acoustic damping that allows the tweeter, which has a low-resonance frequency, to cross over at low frequencies.

I "broke in" the woofers by operating them for a few hours with a very low-frequency signal (around 6 Hz) at a voltage level that caused their cone to move considerably. Next, I measured the T-S parameters using the LIMP software. **Table 1** shows the results. The average values of the parameters of the two woofers as shown in the column were:

f_s= 40.7 Hz, Q_{TS}= 0.49, and V_{AS} = 30 ltr

Parameter	W170S4 #1	W170S #2	Average
F_s	41.93 Hz	39.55 Hz	40.74 Hz
R_e	3.18 Ω	3.22 Ω	3.20 Ω
Q_{TS}	0.49	0.49	0.49
Q_{ES}	0.59	0.58	0.59
Q_{MS}	2.98	3.03	3.01
M_{MS}	11.92 gr	11.84 gr	11.88 gr
R_{MS}	1.0550 kg/s	0.9700 kg/s	1.01 kg/s
C_{MS}	0.001208 m/N	0.001368 m/N	0.001288 m/N
V_{AS}	28.19 ltr	31.90 ltr	30.05 ltr
Sd	129 cm²	129 cm²	129 cm²
Diameter	12.8 cm	12.8 cm	12.8 cm
Bl	4.10 Tm	4.04 Tm	4.07 Tm
Sensitivity (2.83 V/1 m)	91.38 dB	91.20 dB	91.29 dB

Table 1: The table shows Thiele-Small parameter measurements for the Visaton W170S4 woofers used in this project.

Experimenting with the Aperiodic Loading

I made a prototype enclosure with a net volume of 19 ltr to experiment with the aperiodic loading. I wanted also to make comparisons between the closed and the aperiodic loading box so I made the front panel of the box removable and I build two front panels. **Photo 2** shows the prototype box with the aperiodic vent. Later, I realized that it was much easier to block the aperiodic vent very tightly with a piece of solid wood.

D. B. Weems in his book *Great Sound Stereo Speaker Manual* recommends a vent of about

Photo 2: I used this enclosure for the initial tests.

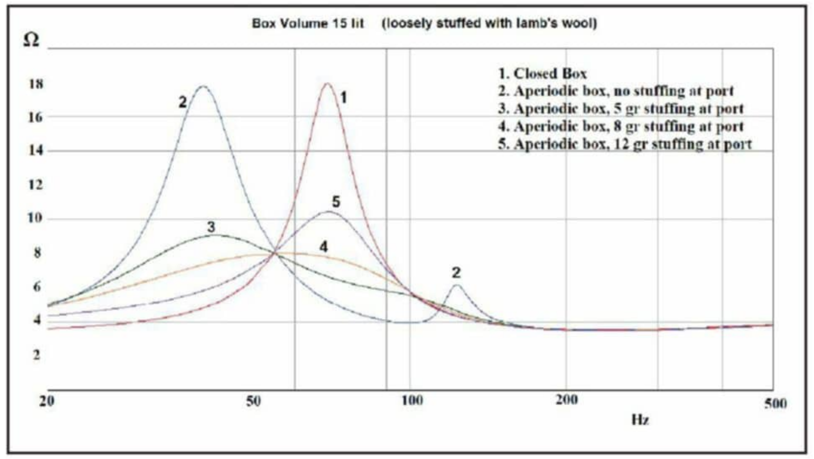


Figure 1: Impedance measurements for the 15 ltr test enclosure

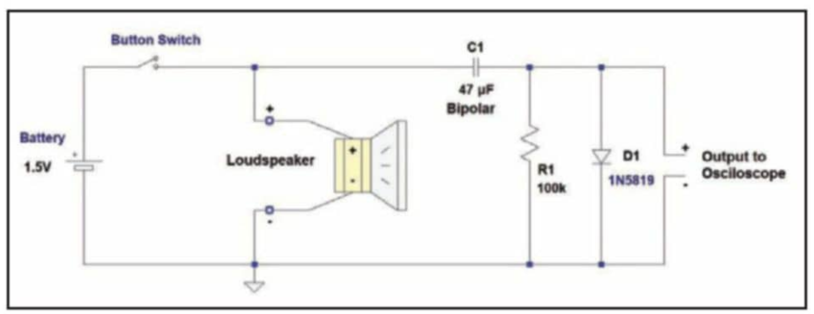


Figure 2: I used this circuit for the damping measurement.

Photo 3: I made this front plate for the adjustment of damping.

10 square inches per cubic foot of enclosure volume, which equals to 64.5/28.3 = 2.3 cm²/liter. For the A25 loudspeaker, Dynaco uses a vent area of around 37 cm² for a net box volume of 22 ltr (1.7 cm²/ ltr), while the A10 loudspeaker uses about 15 cm² for a net box volume of around 10 ltr (1.5 cm²/ltr). For the prototype box, I used a larger vent area of 76 cm².

I found that I could change the active volume of the box by putting solid wooden blocks inside the enclosure. For this project, I used Visaton lamb's wool as damping material for the aperiodic vent.

I tuned the damping of the aperiodic vent using two methods. First by measuring the impedance of the woofer. **Figure 1** explains how I did it. Five impedance measurements are shown with the woofer installed in a 15 ltr enclosure, which was loosely stuffed with lamb's wool.

Curve 1 is the impedance with the box totally closed. The resonance frequency is at 69.5 Hz with a maximum impedance magnitude of 18 Ω.

Curve 2 shows what happens with the aperiodic vent open and without stuffing. The resonance of the box drops to 40 Hz and the resonance of the open vent appears at 122 Hz.

Curve 3 shows what happened with 5 grams (g) of stuffing material to the vent. The vent resonance disappears, the box resonance increases a little at 42 Hz and the impedance magnitude drops from 18 Ω to 9 Ω.

Curve 4 shows that 8 g of stuffing in the vent seems to be the right amount for the aperiodic loading of the box. No clear resonance frequency of the box is shown and the magnitude of the impedance drops a little more at 8 Ω.

Curve 5 shows what happens when the vent is over stuffed with 12 g. The impedance is similar to that of the totally closed box but the maximum magnitude is much lower at about 10.5 Ω.

For the second method, I used the circuit shown in **Figure 2**, which is based on an article I found in a very old issue of *Radio Electronics* magazine (see Resources). The circuit operates as follows: when the switch is closed, the speaker cone is displaced due to the current flowing through the voice coil. When the switch is opened, the cone generates a voltage while it returns to the rest position. **Photo 3** shows the front panel of the device that I built for this test.

I used the "Soundcard scope" software running on a PC (see Resources) to record the voltage. The program receives data from an external USB sound-card.

Figure 3 shows six different measurements that I took for the tuning of the aperiodic vent. The first measurement on the top left side is with the

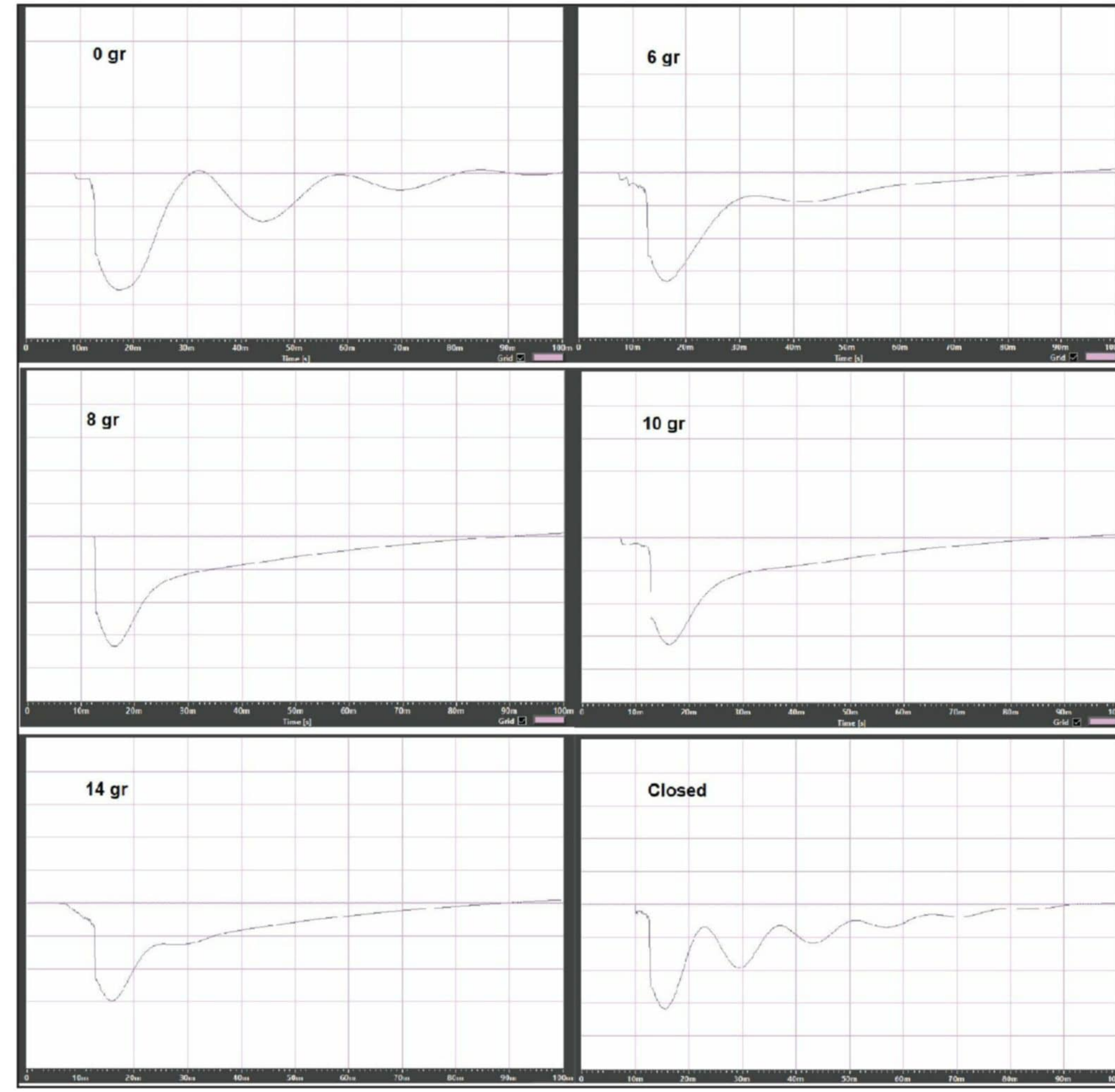


Figure 3: These are the oscilloscope traces of the woofer voltage with different aperiodic vent damping.

aperiodic vent open and without damping material. In the next measurements, the quantity of the damping material is increased from 6 g to 8 g, 10 g and 14 g. The final measurement at the bottom right is with the aperiodic vent closed. The first and the last measurement show a clear oscillation of the recorded voltage. Again the measurement with 8 g damping material in the aperiodic vent shows a very smooth response with the voltage returning to zero without any oscillation.

Next, I measured the impedance for different net volumes of the aperiodic test box in comparison with the same closed box. **Figure 4** shows the measurements for two different volumes of 15 ltr and 10 ltr. The resonance frequency of the 15 ltr

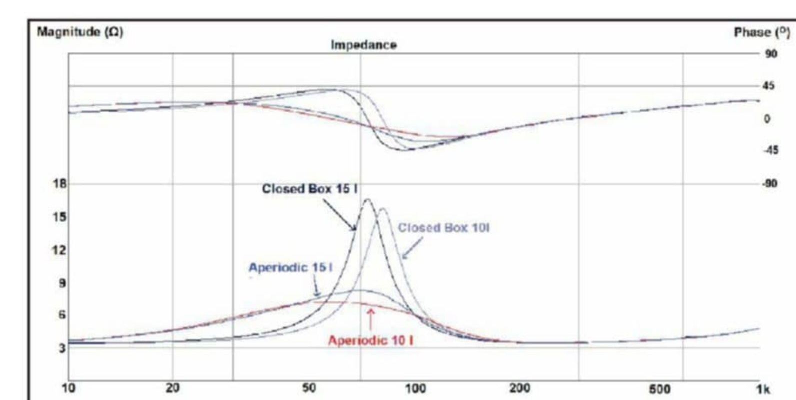


Figure 4: I took these woofer impedance measurements for the closed and the aperiodic test box.

closed box is 73 Hz, with the maximum impedance at 16.6 Ω and the phase variation around the resonance frequency between ±40°. The total Qt of the loudspeaker is 0.88 (Q_{ES} = 1.1, Q_{MS} = 4.4). When the box is aperiodically loaded, no clear resonance frequency is shown, the maximum impedance drops to 8.3 Ω in the region of 70 Hz and the phase variation is only +20°/-30°.

With the box volume decreased to 10 ltr, the resonance frequency of the closed box increases to 91 Hz with the maximum impedance at 15.7 Ω and the phase variation around the resonance frequency between ±40°. The total Qt increases also to 0.95 (Q_{ES} = 1.21, Q_{MS} = 4.49).

When the box becomes aperiodically loaded, no clear resonance frequency is shown, the maximum impedance drops to 7.2 Ω in the region of 55 Hz and the phase variation is only +20°/-25°.

Then I wanted to measure the low frequencies response. As every loudspeaker builder knows, this is very difficult to perform at home. The near-field technique proposed by D.B. Keele circa 1973 is the commonly accepted way to get low-frequency response measurement without an anechoic chamber. First, I placed the microphone very close to the woofer's cone and then at the center of the aperiodic vent's opening. To get the total near-field response, I added them with the proper weighting according to the radiating surface of each area and taking into account the phase response. The result was not very consistent, probably because the woofer cone and the aperiodic vent were very close and the measurements were contaminated by cross-talk.

Joseph D' Appolito published an excellent article in *audioxpress* (June 2012 issue) about measuring loudspeaker low-frequency response. In this article, he describes another technique, the microphone-in-the-box, which was proposed by R. H. Small in an Audio Engineering Society (AES) paper, circa 1971. Small showed that, at low frequencies, there is a simple relationship between the sound pressure level (SPL) at a distance from an enclosure and the internal pressure within the enclosure, regardless of the number of radiating surfaces. To determine the low-frequency response, only one measurement of pressure inside the enclosure is needed. The measurement microphone is placed near the geometric center of the enclosure away from walls and interior baffles.

I used a Panasonic WM-61A cartridge modified according to Siegfried Linkwitz's instructions as described in detail on his website (see Resources). With this modification, the WM-61A can measure

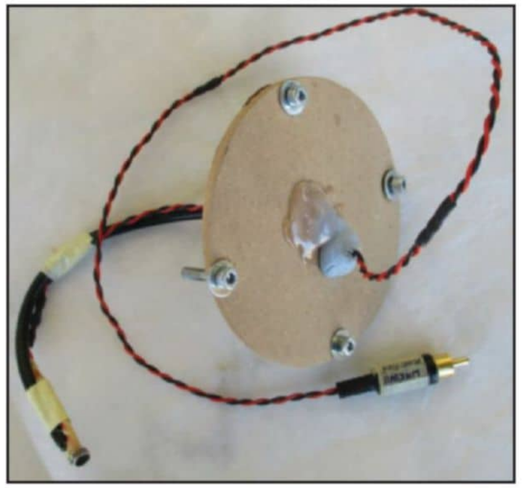


Photo 4: The microphone-in-the-box construction

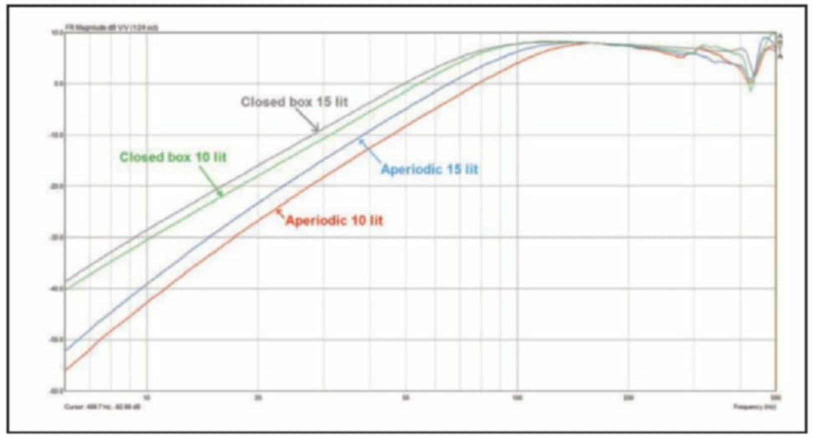


Figure 5: These are the frequency responses measurements for the closed and the aperiodic test box.

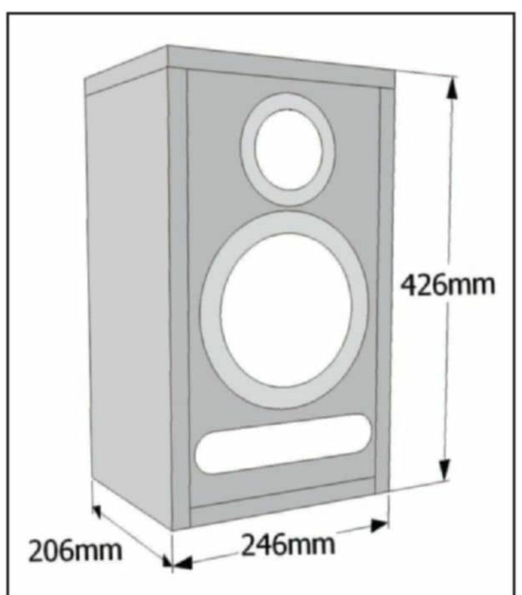


Figure 6: I used these dimensions to build the APR-17 loudspeaker.

Photo 5: The internal braces of the box

Photo 6: Here is a close-up shot of the aperiodic vent.

the very high pressure levels existing inside the loudspeaker box without significant distortion. I cut a circular piece of MDF with dimensions similar to the SEAS tweeter and I put it in its opening on the front panel of the box. The microphone was supported at the end of a flexible wire. **Photo 4** shows the details of this construction. The exact point of the microphone inside the box is very critical, but after some experimentation by moving the flexible wire that supported the microphone, I found a point close to the center of the box that gave very reliable results. This was also confirmed by comparing the microphone-in-the-box measurement with the near-field measurement with the box completely closed.

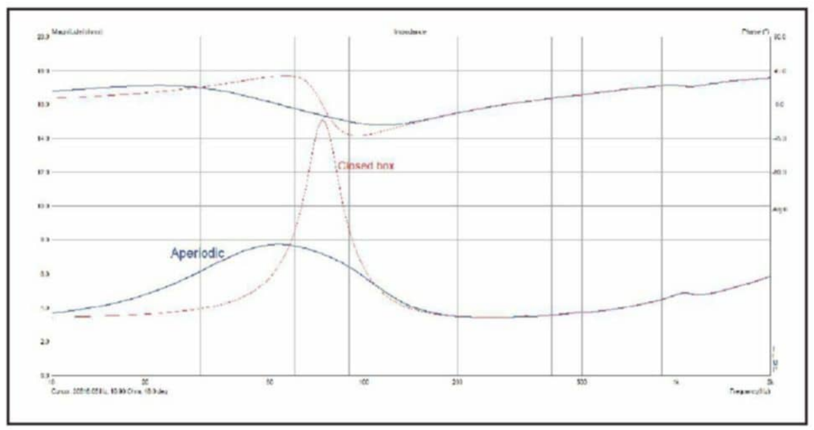


Figure 7: This graph shows the impedance of the APR-17 woofer without crossover.

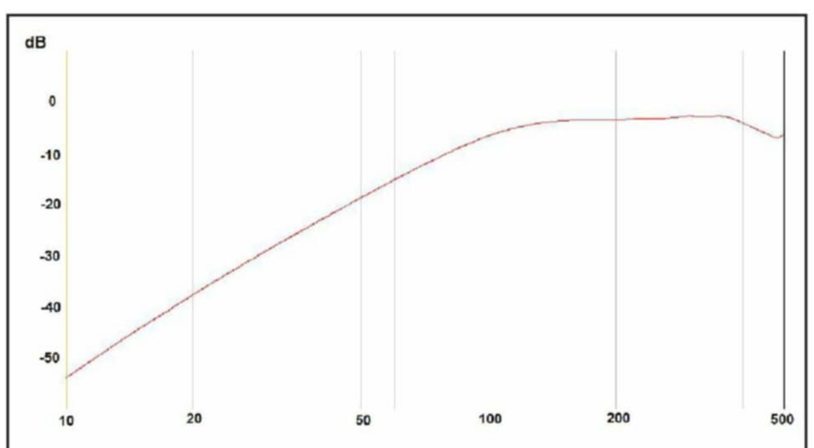


Figure 8: Here is the low-frequency response of the APR-17 woofer without crossover.

About the Author

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

Figure 5 shows the measurements of the low frequencies response. Four curves are shown, for a box with a volume of 10 ltr and 15 ltr, when the box was closed and with aperiodic loading. All curves were normalized to have the same response at the region of 150 Hz.

The low-frequency response of the closed box falls off with a slope of 12 dB/octave, while the aperiodic loaded box starts to fall off with around 12 dB/octave in the range 90 Hz to 45 Hz and then with a slope of 16 dB/octave at the lower frequencies. Later, as I will describe in Part 2 of this article, I discovered another method to measure the low-frequency response that gave a slope of 18 dB/octave.

The low-frequency response of the aperiodic loaded box is always lower than that of the corresponding closed box. For example, at 50 Hz, with the 0 dB reference at 150 Hz, the response is -7.5 dB down for the 15 ltr box, -9 dB for the 10 ltr closed box, -12 dB for the 15 ltr aperiodic box, and -15.9 dB for the 10 ltr aperiodic box.

Construction of the Final Box

I chose the net volume for the final loudspeaker box to be 12.5 ltr. If I had to make this decision again, with the experience I have now from the entire project, I would choose the net volume to be around 8 ltr to 10 ltr. This probably would have made better use of the aperiodic loading.

Taking into account the volume of the woofer, the tweeter, the crossover, and the internal braces, I increased the total internal volume of the box to 14 ltr. The dimensions of the box as shown in **Figure 6** are: height 42.6 cm, width 24.6 cm, and depth 20.6 cm. The total external volume of the box is 21.6 ltr.

I used 18 mm birch plywood for the construction of the box. All the pieces were cut to the required dimensions at a local shop making the assembly of the box relatively straightforward. The sides were glued together with a good quality wood glue. The holes for the drivers were opened with a router.

With experience from my previous projects, I designed the bracing to eliminate the box vibrations as much as possible. I used a main vertical internal brace and three horizontal braces. One of the two horizontal braces was placed between the two drivers and separates the internal of the box in two parts. Several openings in all braces assure that the box's entire volume is seen by the woofer. The top image of **Photo 5** shows all the braces that I used before they were glued to the box, while the bottom image shows them glued inside the box.

I sanded the outside of the box and I applied several layers of a wood imitation varnish for a nice finish. **Photo 6** shows a closer look at the aperiodic vent.

I loosely filled the inside of the box with 150 g of Visaton lamb's wool to attenuate resonances—otherwise they could be transmitted through the cone of the woofer driver. I kept the path between the back of the woofer and the opening of the aperiodic vent clear. I dampened the aperiodic vent as previously described with 8 g of Visaton lamb's wool.

Low-Frequency Response Measurements of the Final Box

Figure 7 shows the woofer's impedance measurement when it is directly driven without crossover. Two curves are shown—one with aperiodic loading and another with the box closed. The resonance frequency of the closed box is 72.2 Hz with a total Qt = 0.84 (Q_{ES} = 1.04, Q_{MS} = 4.28). The maximum magnitude of the impedance is 16.5 Ω, while the phase variation around the resonance frequency is ±42°. With the aperiodic loading, the impedance shows no clear resonance while the maximum impedance drops to 7.8 Ω in the region of 55 Hz and the phase variation is only ±25°.

Figure 8 shows the woofer's low-frequency response measurement with aperiodic loading and driven directly without crossover using the Microphone-in-the-box assembly.

Next Month

In the second part of the article, I will continue with the design of the crossover and the measurements that will enable us to evaluate the loudspeaker.

Resources

J. D' Appolito, "Measuring Loudspeaker Low-Frequency Response," *audioXpress*, June 2012. https://audioxpress.com/article/measuring-loudspeaker-low-frequency-response

ARTA & LIMP software, www.artalabs.hr

S. Linkwitz, "Modification on Panasonic Back Electret Microphone Cartridge," Linkwitz Lab, www.linkwitzlab.com/sys_test.htm#Mic

R. H. Small, "Simplified Loudspeaker Measurements at Low Frequencies," *Journal of the Audio Engineering Society*, 1972.

Soundcard scope software program by C. Zeitnitz, www.zeitnitz.eu/scope_en

D. J. Tomcik, "Missing Link in Speaker Operation, Part 2", *Radio Electronics*, January 1955.

D. B. Weems, *Great Sound Stereo Speaker Manual with Projects*, TAB Books, 1990.

The Early Development of Hollow-State Mixers

In this article, our columnist reminisces about the development of the audio mixer and he details how we arrived at today's complex modern-day devices.

Photo 1: The Victor Orchestra is shown recording acoustically.

By
Richard Honeycutt
(United States)

When I train an operator to use an audio mixer in an auditorium or other venue, invariably the first question I am asked is, "How will I ever learn all those buttons?" Today's mixers provide numerous functions beyond simply combining audio signals, and this impressive capability does indeed come with the cost of complexity.

Photo 2: This Western Electric "ring-and-spring" microphone was an early model used in making disc recordings. (Image courtesy of https://en.wikipedia.org/wiki/History_of_sound_recording)

But in the early days, audio mixers were much simpler. In fact, before 1960, most of the functions we expect in an audio mixer today simply were not available. Let's go back in time and review the early development of the audio mixer.

Early Recordings

The first audio recordings used no electronic devices, not even mics or mixers. The performers gathered around the large end of a horn and played/sang. The horn matched the low acoustical impedance of the airborne sound wave to the necessary high impedance of the cutter head used to inscribe the track onto the Edison cylinder or disc (see **Photo 1**).

Not until 1925, when Western Electric began to market microphones, recording amplifiers, and electrical cutter heads, did electronics become involved in the disc recording process (see **Photo 2**). Even then, recordings were made using the single-mic technique, so no mixers were needed.

The first AM broadcast, engineered by Reginald Fessenden and transmitted on Christmas Eve 1906, included both music (Fessenden playing "O Holy Night" on violin) and speech (Fessenden reading from the Bible).

Commercial radio broadcasts including both music and speech probably began at 9 PM on August 27, 1920, when Sociedad Radio Argentina aired a live performance of Richard Wagner's opera *Parsifal* from the Coliseo Theater in downtown Buenos Aires. Information is scanty, but it is likely that this broadcast used a single microphone for both speech and music.

Early Mixers

Radio broadcasts using multiple microphones debuted in the 1930s—about the same time that electrical sound reinforcement systems appeared, and the need for mic mixers was answered by two companies: Western Electric and RCA. The WE 22C mixer shown in **Photo 3** was designed for speech sources, which were the primary uses for which a mixer would be needed at that time. The RCA BC-5 mixer shown in **Photo 4**, also designed for speech sources, was often modified for recording sound for movies.

Figure 1 shows the functional block diagram of the WE 25B mixer, a successor of the 22C. Like most mixers of the time, its construction was modular. The components labeled "REP COIL" were 1:1 isolation transformers, "repeat coils" in telephone jargon. The pre-mixing amplifiers were two-stage pentode circuits, transformer-coupled at input and output. The main amplifier used three pentode stages. Although the 25B had two output channels, they were used to feed separate systems (e.g., a broadcast line and a control room). Stereo was not yet a commercial reality. The "audition output" is what would be called a "cue output" in later broadcast mixers. It would be fed to an amplifier and speaker to allow the operator to cue and hear records in the control room without affecting the broadcast. In modern live sound mixers, this function is called Pre-Fader Listen (PFL).

Transformer-Balanced Circuit Use

The emphasis on using transformer-balanced circuits in early mixers came from experience in telephone engineering. The electrically and electromagnetically induced hum on a two-wire telephone line was not only detrimental to conversations, it could literally be life-threatening. Both kinds of induced AC-line-frequency voltages increase with the length of the exposure: unlike a "long" (maybe 150′) mic line in an outdoor concert, a telephone line could involve a parallel exposure with a power line that was hundreds of miles long. Although electrostatic shielding reduces the effect of electrical induction, it does little to mitigate electromagnetic induction. So telephone engineers were very careful about limiting induced AC voltages. The most effective way of doing this is to use balanced lines.

A microphone feeds its output to two wires, and the signal at the receiver is taken from one wire to the other; this is called a "normal-mode" signal. The voltage induced from power lines is

Photo 3: This Western Electric Model 22C "portable" four-channel mixer was designed for broadcast and sound-reinforcement use with speech signals.

Photo 4: This RCA Model BC-5 was also made for broadcast use. (Image courtesy of www.worthpoint.com/worthopedia/rca-tube-console-bc5-re-built-studio-1814847982)

equal and in-phase: a "common-mode" signal. If the series impedances of the two conductors between the source and load (mic and receiver in a telephone system, or mic and mixer in a sound system) are not equal (unbalanced), and/or if the shunt impedances (line to ground) are unequal, common-mode voltages will be partially converted to normal-mode voltages. So the life hazard comes from high common-mode voltages, and the communication interference comes from high normal-mode voltages. Several methods exist for avoiding high common-mode voltages, and these involve "inductive co-ordination" between telephone and power companies.

Avoiding common-mode-to-normal-mode conversion involves well-balanced audio circuits. The only practical way of balancing the input and output circuits of audio equipment in vacuum tube days was the use of transformers: a good signal transformer could achieve as much as a 90-dB Common-Mode Rejection Ratio (CMRR). Even though network feeds and remote feeds (inputs), and studio-to-transmitter links (outputs) were the only connections between telephone lines and broadcast mixers, the use of balanced circuitry became ubiquitous, in order to prevent hum.

In addition to hum considerations, input and output transformers were used for impedance control. Most mics had impedances of 50 Ω to 150 Ω (seldom more than 600 Ω), and the input impedance of a hollow-state preamp is usually about a megohm. Thus a "step-up" transformer with an output-impedance: input-impedance ratio of 100 or so could be used, giving a noise-free voltage gain of about 10. (Truly matching the input impedance of a mic can create frequency-response anomalies, and also reduce the usable output voltage of the mic by 50%, so the often-heard mention of "impedance matching" in audio equipment is seldom accurate.)

Even if a cathode-follower output stage is used, properly driving the typical 600 Ω impedance of a telephone line (which became the de-facto standard impedance of "line-level" audio electronics) was difficult. So output transformers came to the rescue. Even in the early days of solid-state mixers, transformer-balanced inputs provided a higher CMRR than transformerless input stages could.

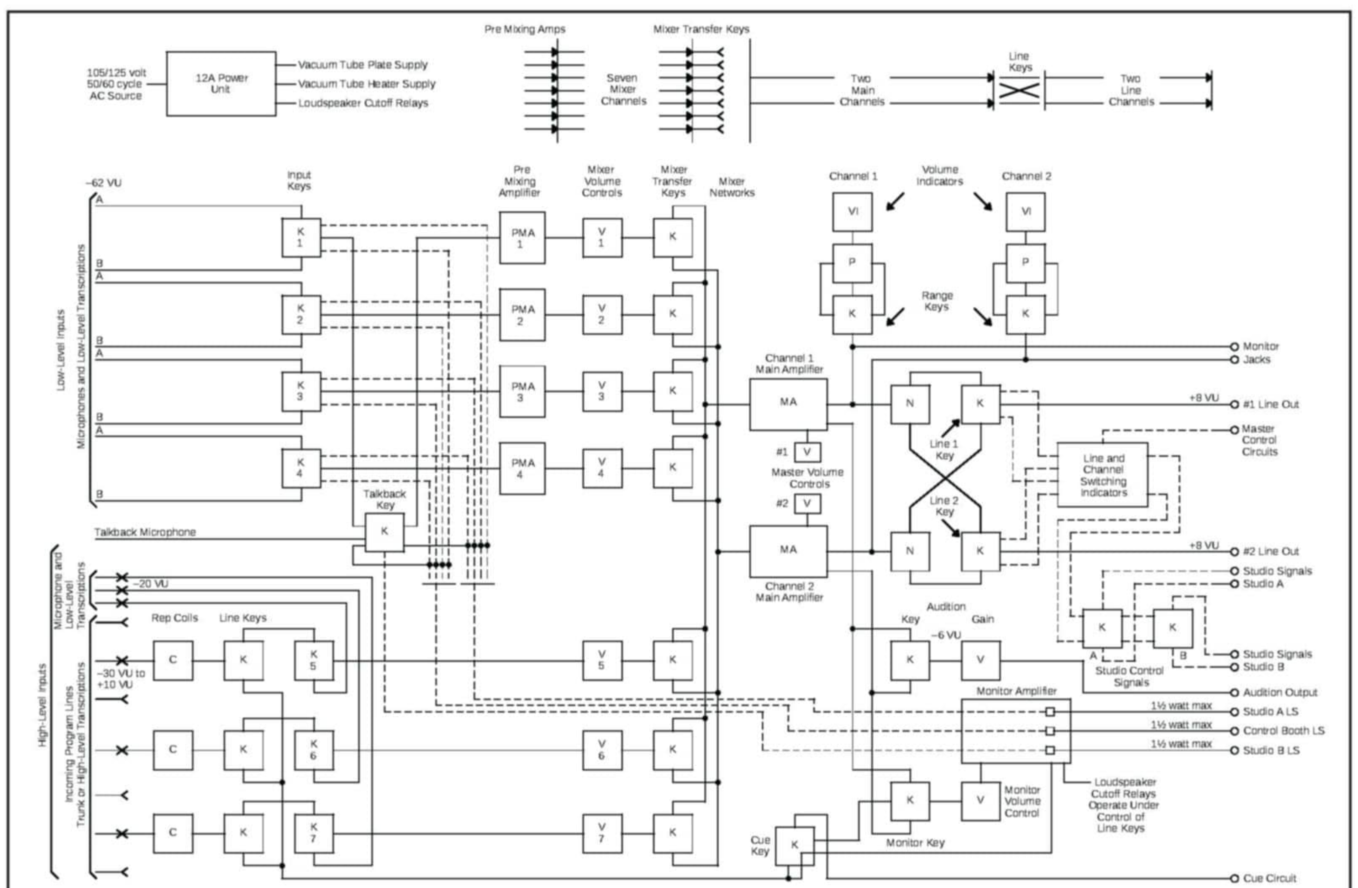


Figure 1: This functional block diagram of the WE 25B mixer is similar to that of most mixers of its era.

Mixers in Movies

By 1957, other manufacturers had entered the broadcast mixer market, and a new market for mixers specifically designed for movie making had developed. **Photo 5** shows a Dualux mixer by Gates Radio Co. that was targeted toward the broadcast radio market, but was also used by about 20 recording studios. Much like its smaller predecessors, this 12-channel model had a three-position switch above each "fader" (variable attenuator). Its left- and right-hand positions assigned the channel to one of two output lines; center was "off." Above the channel assign switches was a row of source assign switches for selecting a particular mic, tape, or remote input to be fed to the channel. The Dualux had inputs for seven mic preamps, two turntables, a network, four tape recorders, and four remote lines.

Since radio stations of the time often remotely broadcast events such as sports and news reports, in which vibration and wind noise could be a problem, the Dualux included a three-position high-pass filter with cut-off frequencies near 100 Hz, and varying response slopes, as shown in **Figure 2**. With the filter in position 0, the Dualux had a very flat, extended response, as required by Federal Communications Commission (FCC) proof-of-performance standards for FM broadcasting (50 Hz to 15,000 Hz).

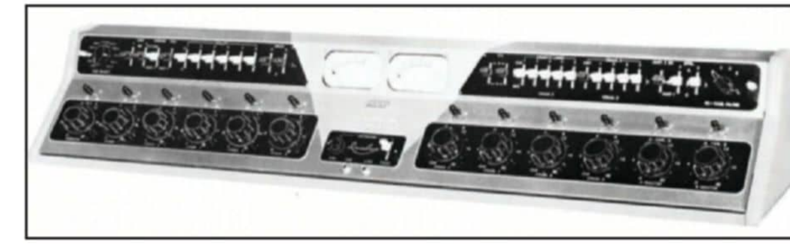


Photo 5: This 1957 Gates Dualux console was one of the later hollow-state broadcast audio mixers.

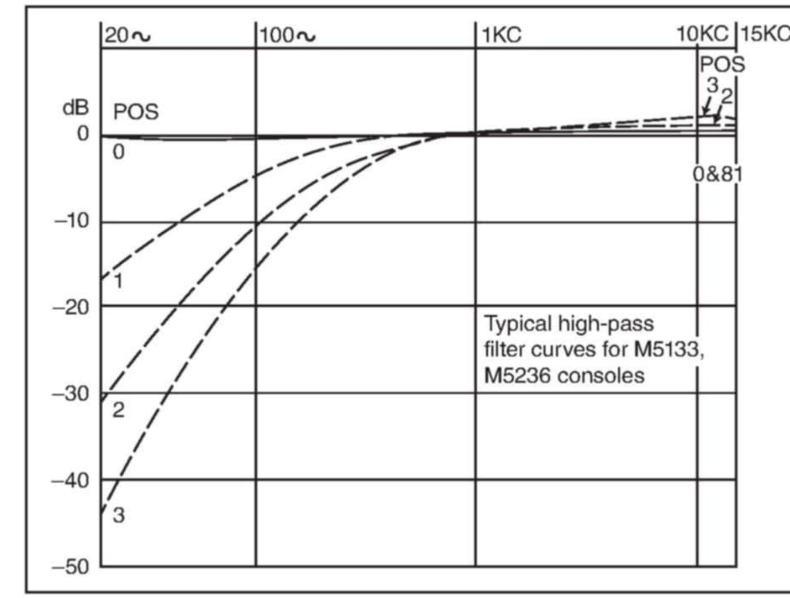


Figure 2: The Dualux had a front-panel high-pass filter with three different frequency responses.

Mixers for Live Sound

In the late 1950s, Altec Lansing introduced its model 1567A, a small five-input hollow-state mixer that would become ubiquitous in the live-sound equipment lineups of the 1950s and 1960s (see **Photo 6**). This model used plug-in input and output transformers, allowing its use with unbalanced line inputs through simple jumpering of the transformer sockets. Using three 12AX7 dual triodes as input and mixing amplifiers, and a 6GC7 as the output amplifier, the 1567A could either be rack-mounted for fixed installations or mounted in an optional carrying case for portable applications.

Photo 6: The Altec-Lansing 1567A was a small portable hollow-state mixer that was quickly adopted for all sorts of venues.

Another major player in the hollow-state audio mixer industry was Collins Radio Co. The Collins Model 212-G1 10-channel console, introduced in 1960 (see **Photo 7**), was the main broadcast mixer used in the first radio station for which I worked in the early 1960s. One of its predecessors, the 212B, was the recording mixer in the first recording studio

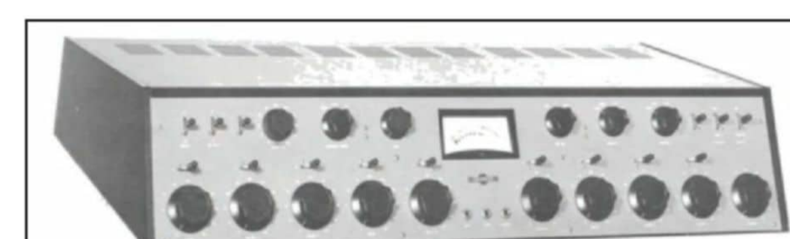


Photo 7: The Collins 212-G1 broadcast console was highly respected in its day.

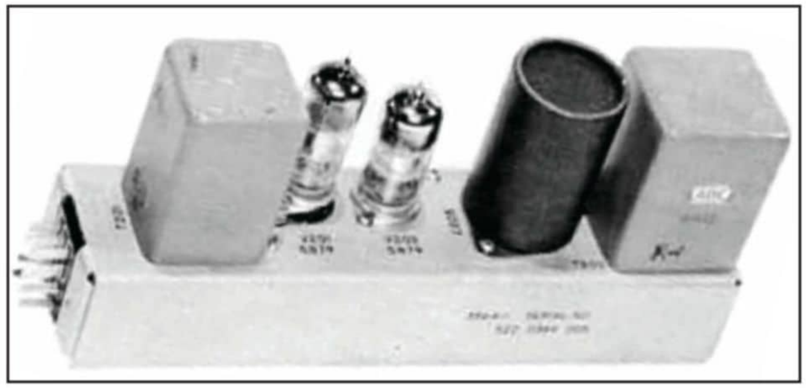


Photo 8: This two-stage plug-in preamplifier allowed easy servicing.

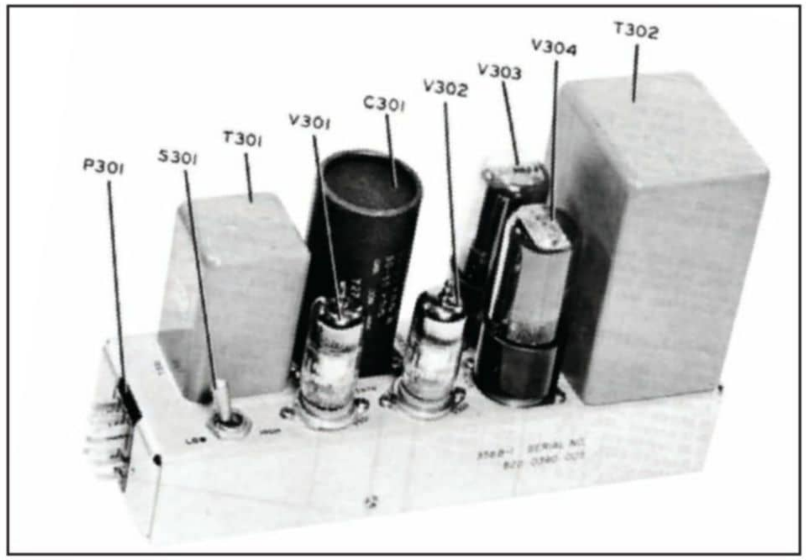


Photo 9: The Collins model 356B-1 main/program amplifier used in the 212 mixer was also modular.

Photo 10: This Daven stepped attenuator was a ubiquitous fader found in many hollow-state mixers.

in which I performed in 1964 as a sideman. Since broadcast consoles were mission-critical, almost all hollow-state models used modular architecture (see **Photo 8**). Thus if a mic or phono preamp failed in service, the engineer could quickly unplug the defective unit and replace it with a spare from the station's stock, resulting in a minimum of downtime.

The main/program amplifier (see **Photo 9**) was also pluggable, and used a push-pull 6V6 pair to provide the capability of driving long lines, such as might be used for the studio-transmitter audio link, or to drive 600 Ω professional headphones for monitoring.

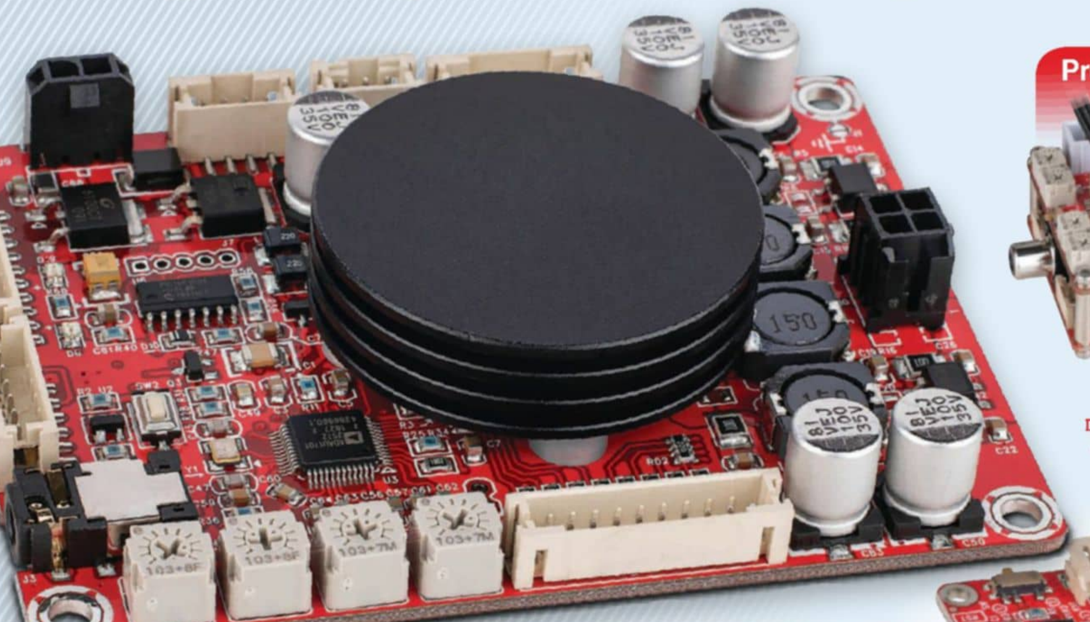
Often modular preamps and program amps were designed to drive 600 Ω loads, and the level controls used were not simple potentiometers as are used today, but were stepped "T" or "ladder" attenuators that could be configured for either 600 Ω input or high (greater than 1000 Ω) output impedances and whose attenuation could be varied, typically, in 2-dB steps. **Photo 10** shows a Daven stepped attenuator, with a typical knob used for the rotary faders common in mixers of the mid-1960s and earlier.

Modern Mixer Functionality

Despite the differences in circuit-design philosophy, the most obvious difference in mixer design between the days of the classic mixers we've discussed and modern analog mixers is the vast increase in functionality in today's mixers. A modern analog mixer may have up to 80 channels, each assignable to four or more subgroups pannable to left or right output; perhaps a summed mono output as well; half-a-dozen or more auxiliary channels into which signals from each channel can be separately mixed; an input level "trim" control; low-cut filters switchable by channel; built-in phantom power for capacitor mics; and perhaps four-band equalization on each channel with sweepable midrange frequencies. Digital mixers may add programmability and several DSP-like functions. Compared to this palette of capabilities, the old four-in, two-out hollow-state mixers may seem limited indeed.

However, mixer designers today stand on the shoulders of the giants—engineers at Western Electric, RCA, Altec Lansing, Gates, and Collins. These were the people who laid the groundwork for our modern mixers by developing relatively simple units that were used to produce trendsetting shows in the glory days of radio broadcasting, not to mention recording phonograph records by great musical performers from Al Jolson and Paul Whiteman's orchestra to Elvis Presley and the Boston Pops!

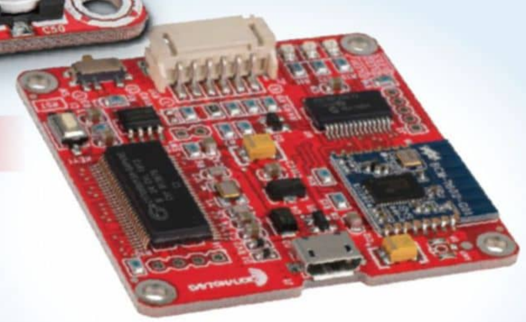
Customized Audio with Digital Signal Processing **Made Easy!**



Dayton Audio DSPB-K
DSP Kernel Board and
DSPB-KE Kernel
DSP Expansion 2-In 3-Out
for Non-DSP Amp Boards

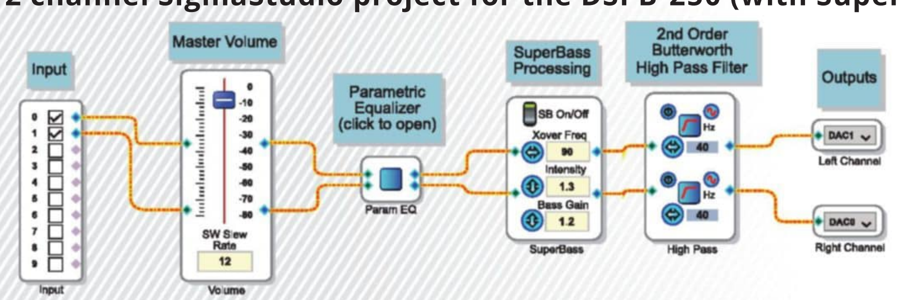
Dayton Audio DSPB Class D Audio Amplifier Board with DSP

Combined with either the DSPB-250, 2 x 50W (#325-126) or the DSPB-100, 1 x 100W (#325-135) amp board with the ADAU1701 digital signal processor, the Dayton Audio Class D Audio Amplifier Board with DSP enables you to customize sound using the SigmaStudio software and Dayton Audio DSPB-ICP1 Bluetooth programmer. Using either one of these amp boards as the heart of your next DSP audio project, or combining it with Dayton Audio KAB-250v3 Audio Amplifier board, you're sure to easily create an awesome Bluetooth DSP 2 or 4-channel experience.



Dayton Audio DSPB-ICP1 In-Circuit Programmer
for DSPB-250/100 and DSPB-K Boards

2 channel SigmaStudio project for the DSPB-250 (with Superbass processing).



DSP programming maximizes the output potential of small speakers and makes the sound of any audio project uniquely yours.

A dramatic expansion of the popular KAB amplifier line, the DSPB series of amplifiers use the same familiar form factor and connections of the KAB line, but now with an Analog Devices ADAU1701 DSP built right onto the board! Used by audio professionals and DIYers alike, the ADAU1701 processor allows for precise EQ, advanced crossover slopes, bass enhancement, time alignment, limiting and SO much more. Perfect for any audio tinkerer! All of the best portable speakers on the market use DSP to get the best performance possible out of small speakers and to create a unique sound signature, and now you can too! The thoughtful design of the DSPB amplifiers also allows them to be used in conjunction with any existing KAB amplifier for up to 4 channels of DSP and Bluetooth input. Dayton Audio didn't stop there, the DSPB-K + DSPB-KE is a DSP preamp that has the same ADAU1701 processor, but lets you use any amplifier you want.

Custom OEM inquiries welcome.

DAYTONAUDIO

SAVE $15 **Code: AXMQ1Q4** Expires 9/30/2020
off $150 order + FREE SHIPPING!
Not valid with any other offer, on prior purchases or backorders.
Not valid on Commercial Sales accounts. No cash value. Valid for one time use.

Distributed by:
PARTS EXPRESS
AUDIO COMPONENTS & SOLUTIONS
parts-express.com
1-800-338-0531

and tube noise. For the highest reliability with DC heater power, the general advice is not to make the heater positive with respect to the cathode; yet, some RCA data books recommend that you make the heater positive with respect to the cathode, ideally by 40 V or so, presumably to minimize noise. So as with everything in life, there are trade-offs. With an AC heater, the polarity between the heater and the cathode reverses with the mains frequency and that appears to ameliorate some of the effects mentioned earlier. In any case, unless you have insurmountable hum issues, it is best to run the heater on AC.

One way to avoid or limit any of the mentioned parasitic currents is to return the heater to ground, or whatever point is selected, via a resistor, to make the leakage (or in case of a short, the shortage) current loop high impedance. The old picture tube datasheets often recommended grounding (or biasing) the heater through a 1M resistor, but in audio amps it can be (much) smaller (see **Figure 3**).

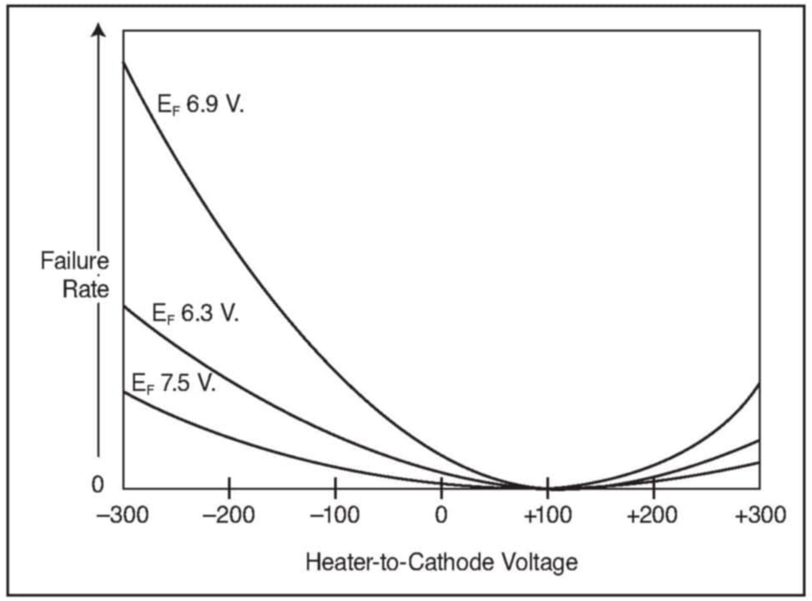


Figure 3: Increasing Vkf above approximately 90 V decreases tube lifetime.

Coming back to the maximum allowed voltage between cathode and heater, obviously you want to keep it below the datasheet value to avoid flashovers or shorts. But even with voltage differences as low as 30 V and heaters on AC, there can be ionization of remaining gas molecules inside the tube, deteriorating the isolation between cathode and heater—leading to shorts in the long run. If there's enough current capacity in the circuit, you can get a heater that is "stuck" to a cathode, resulting in a defective tube. As mentioned earlier, there will be an increase in leakage between the cathode and the heater and increased tube noise long before there is an actual short. It seems a long-term effect though, so only of importance if you want to eke maximum lifetime out of your tube. Although stories abound that a mild mechanical shock to such a tube can separate the heater from the cathode, I have not been able to confirm that.

Grid Current

Power tubes are often operated with positive Vgk, which cause grid current to flow. As long as the tube is designed for that and the drive circuit is low impedance, that'll be fine. (If the drive circuit is not low impedance, the grid current will cause distortion of the drive signal).

Sometimes, you might want to use small signal tubes (e.g., the ubiquitous dual triodes), with very low supply voltages, 50 V or less, and consequently very low Vak. In those cases the Vgk gets very low too, and even when it doesn't actually become positive, grid current can start to flow at Vgk more positive than -1 V. You can do that, if you keep the grid current low, but you need to condition the tube for that. The grid surface has all kinds of foreign material, and when grid current starts to flow, this material will be "burned off" so to speak. That means that the characteristics will change during the initial period with grid current. This is one case where burn-in is useful!

Fixed vs. Autobias

Seemingly having no relationship per se to tube life, the choice for biasing method nevertheless has an impact on the probability of tube failure. If a circuit fault develops in a fixed bias circuit, tube current may run away unchecked, leading to tube destruction. With auto-bias, there is always the cathode resistor, providing some amount of negative feedback to limit tube current.

A similar issue can arise with fixed screen bias. If in such a circuit the anode voltage collapses due to some failure, the screen will probably start to run red hot, causing destructive gassing, warping of elements, and finally tube failure.